Intermodulation at the amplifier-loudspeaker interface

Part 1: Analysis of one source of audible difference between amplifiers

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Intermodulation occurs between an amplfied signal and a delayed version returned from a loudspeaker through a feedback loop, when open-loop output impedance is high compared to speaker impedance. Part one of this article analyses this and a second part describes a measurement method with results of tests on different types of amplifier circuit and suggestions for avoiding the effect.

The sound quality at the low-frequency end of the audio reproduction chain has often been discussed in such subjective terms as firm, soft, dry and mellow. As far as loudspeakers are concerned, the change in sound impression may be explained as a result of different technical characteristics of the drivers, filters and cabinets. Amplifiers present a more serious problem because the level of harmonic distortion at these frequencies is usually low, the frequency response is relatively flat, and output damping is almost always adequate.

An intriguing question sometimes encountered in practice is why the sound may perceivably change at the low end of the frequency spectrum when the same listening environment and the same loudspeaker system is used and only the power amplifier is changed. It is our experience that certain power amplifier circuit topologies sound different to others, although no directly explainable difference is noted in the electrical performance of the circuits when tested with resistive load. The following analysis shows that, under certain conditions, the loudspeaker reaction to the drive signal can propagate in the feedback loop of a power amplifier and intermodulate with the drive signal itself. This may partly answer the question.

The dynamic loudspeaker provides a complex load to the amplifier. As much has been written about its behaviour (see, for instance, references 1), it is sufficient here only to present a short list of some of the most important factors affecting the interface between the loudspeaker and the amplifier.

The total compliance of the cone suspension and the loudspeaker cabinet, and the cone mass, form a damped mechanical resonance, typically in the frequency range of 30 to 80Hz for the woofer and at correspondingly higher frequencies for the squawker and tweeter. Other mechanical resonances are created by the different moving parts of the cone, excited by the voice coil, but not necessarily rigidly coupled to it. All these mechanical resonances behave like parallel tuned circuits in series with the voice coil resistance and inductance. The crossover filters also exhibit complex reactive behaviour, especially around the crossover frequencies. Figs 1 & 2 show the impedance of two popular loudspeaker systems manifesting both cone and crossover filter resonances.

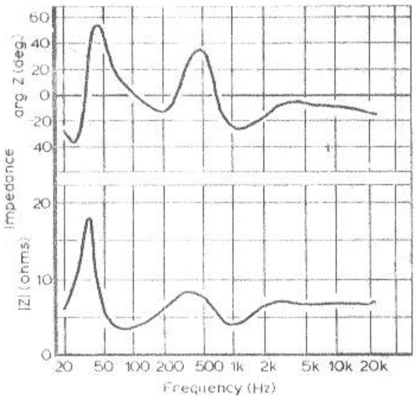

Fig. 1. Magnitude and phase of the terminal impedance of an Acoustic Research AR3a loudspeaker system, measured with the controls in midposition. Resonant frequencies are 32Hz, 330Hz and 2.5kHz.

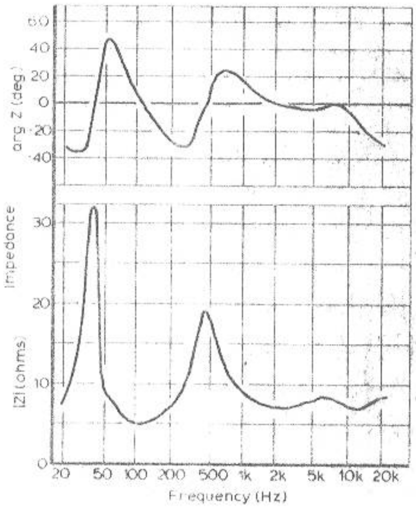

Fig 2. Magnitude and phase of the terminal impedance of a Yamaha NS-1000 Monitor loudspeaker system, measured with the controls in midposition. Resonant frequencies are 38Hz, 410Hz, and 5.5kHz.

Energy is stored in all these reactances, especially in the resonances. Because a reactance cannot dissipate energy, and the internal dissipation in the loudspeaker is low at these resonances, most of the stored energy returns to the amplifier and is dissipated in it. In addition, the loudspeaker terminal impedance is non-linear, and cone break-up, delayed responses and acoustical reflections create generator effects in the loudspeaker. Fig. 3 shows a greatly simplified equivalent circuit of a loudspeaker, taking into account only few of the effects discussed.

Now analyse a feedback amplifier having two different loads, as shown in Fig. 4. A pure resistance R is used when measuring the characteristics of the amplifier. A loudspeaker, represented by the grossly simplified equivalent circuit of Fig. 3, is the true load. It is assumed to have a linear resistance R and negligible voice coil inductance L_v to facilitate the analysis. The circuit is far from perfect, but this analysis is to illustrate the basic mechanism of distortion only, not to calculate it to a high degree of accuracy. Similarly, the amplifier is assumed to have an infinite input impedance, and no frequency compensation. All these approximations do not affect the result of the analysis. Note that a new parameter, the open-loop output impedance Z, has been incorporated in the circuit in contrast to prior analyses.

The input signal V_1 is taken to be a step function $V(t)$, so that its Laplace transform

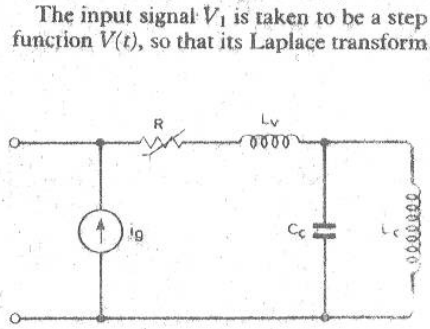

Fig. 3. Simplified loudspeaker equivalent circuit. L_C and C_C are cone dynamic mass and suspension compliance, respectively. L_V the voice coil inductance, R the voice coil resistance, including the radiation resistance, and i_g the generator effect current source.

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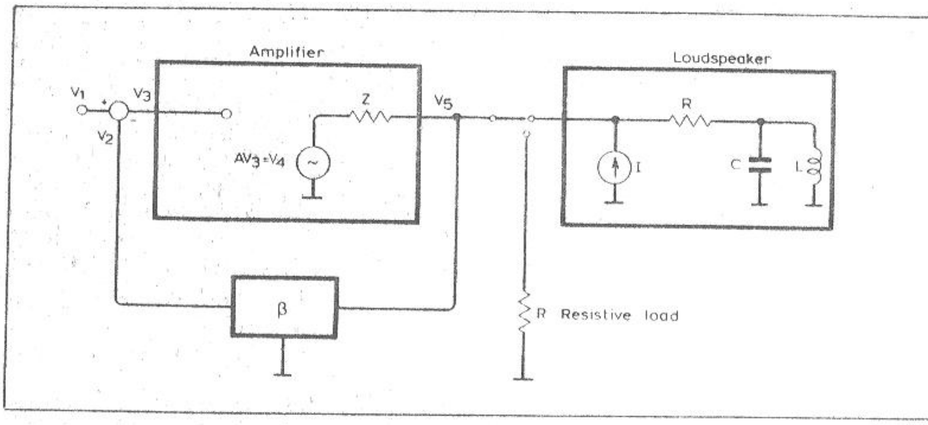

Fig. 4. Equivalent circuit of the amplifier-loudspeaker combination used in the analysis. Amplifier has resistive feedback network β, gain of A and open-loop output impedance Z. No frequency-dependent effects are incorporated in the amplifier circuit. The loudspeaker is assumed to have negligible voice coil inductance and is considered to be perfectly linear.

is $\mathcal{L}[V_1]=v_1/s$. The analysis is based on linear theory.

For the resistive load R, the transforms of voltages V_4 and V_5 are

$$V_4(s)=\frac{A(1+Z/R)}{s(1+Z/R+\beta A)}v_1$$

and

$$V_5(s)=\frac{A}{s(1+Z/R+\beta A)}V_1 \qquad 1$$

The inverse transforms are both perfect step functions and the only difference to standard feedback equations is the term Z/R. An adequate damping factor necessitates that the closed-loop output impedance of the amplifier be much smaller than the loudspeaker impedance, i.e.

$$R \gg Z/(1+\beta A)$$

which yields a further simplification. Taking the inverse Laplace transform, the voltages are found in time domain

$$V_4(t)=\frac{A(1+Z/R)}{(1+\beta A)}v_1\,U(t) \qquad 3$$

and

$$V_5(t)=\frac{A}{(1+\beta A)}v_1\,U(t).$$

If now the loudspeaker is substituted for the load, the situation changes markedly. Assuming the damping to be adequate, as by equation 2, equations 1 take the form

$$V_4(s)=\frac{A}{1+\beta A}\Big[-\beta ZI(s)+\frac{v_1}{s}\left(1+\frac{Z}{R}\right)\frac{s^2LC+sL/(R+Z)+1}{s^2LC+sL/R+1}\Big] \qquad 4$$

and

$$V_5(s)=\frac{A}{s(1+\beta A)}v_1.$$

No change has occurred in the transformed output voltage V_5 of the amplifier. This is to be expected, as the feedback effectively controls the output voltage. However, the internal drive voltage of equation 4 now contains complex terms consisting of the parameters in the loudspeaker equivalent circuit. To study the behaviour of this voltage in time domain, the inverse Laplace transform of equation 4 yields

$$V_4(t)=\frac{A}{1+\beta A}\left[-\beta ZI(t)+v_1\left(1+\frac{Z}{R}\right)\left\{1-\frac{Z}{(R+Z)}\frac{1}{Q}\exp\left(-\frac{\omega_1 t}{2Q}\right)\sin\omega t\right\}\right] \qquad 5$$

where $\omega_1=(1/LC-1/4R^2C^2)^{1/2}$, the resonant frequency of the loudspeaker cone, terminals short-circuited, and $Q=\omega_1 RC$, the approximate quality factor at resonance.

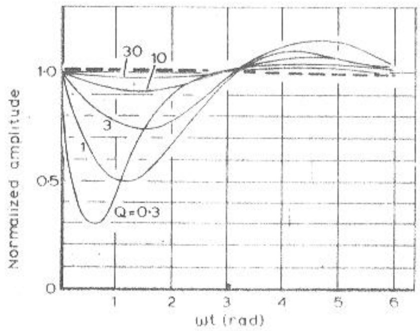

Fig. 5. Typical waveforms from equation 7 as functions of normalized time, with the resonant quality factor Q as parameter. The loudspeaker-generated oscillation is very large, especially for low values of Q. Corresponding waveform for resistive load is shown with a dashed line.

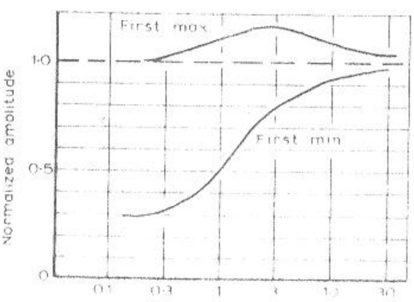

Fig. 6. Values of the first minimum and the first maximum of equation 7 as a function of quality factor Q.

The first term corresponds to the effect of any current generated in the voice coil of the loudspeaker by the vibration of the cone. Assuming that the feedback is large, $1+BA \gg 1$, say greater than 30dB, the first term becomes

$$V_4(t)=-ZI(t)$$

showing that the amplifier internal drive voltage necessary to serve as a sink for the loudspeaker generator current is directly proportional to the open-loop output impedance Z. Dividing this equation by the nominal signal level of equation 3 the ratio of the loudspeaker-generated signal to the driver signal can be found

$$\frac{V_4(t)_{generator}}{V_4(t)_{signal}}=\frac{Z}{R+Z}\frac{I(t)_{generator}}{I(t)_{signal}}$$

Similarly, the last term of equation 5 can be divided by the signal level, equation 3 which yields the ratio of the resonant oscillation in V_4 to the signal in V_4

$$\frac{V_4(t)_{oscillation}}{V_4(t)_{signal}}=$$

$$1-\frac{Z}{R+Z}\frac{1}{Q}\exp\left(-\frac{\omega_1 t}{2Q}\right)\sin\omega t. \qquad 7$$

This represents a damped oscillation at the cone resonance frequency. There are negative minima and positive maxima at

$$T=\frac{1}{\omega_1}(\arctan 2Q+n\pi)$$

where n is an integer, with values

$$V_4(T)=1-\frac{Z}{R+Z}\frac{2}{(1+4Q^2)^{1/2}}\exp\left(-\frac{\arctan 2Q+n\pi}{2Q}\right).$$

Assuming $Z \gg R$, some typical waveforms of equation 7 are plotted in Fig. 5, and the values of the first minima and maxima are plotted in Fig.6 as functions of Q. The amplitude of oscillation increases with decreasing Q. The reason for this apparently strange behaviour is that, when the Q of the resonant circuit is lowered, the circuit absorbs more energy from a broadband signal spectrum.

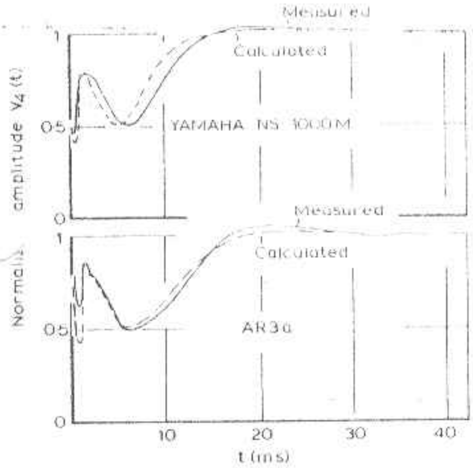

Fig. 7. Measured responses V4(t) and calculated responses for the AR3a and NS-1000M loudspeaker systems. Only the two first resonances around 35Hz and 400Hz were taken into account in the calculated values. The good match of the responses show that the theoretical model used is satisfactory.

To check the validity of the approximations made, the calculated measured responses $V_4(t)$ are shown in Fig.7 for the two loudspeaker systems of Figs 1 & 2. The calculated results are very close to the measured ones, which is surprising considering the complexity of the real three-way loudspeaker systems. This proves that the simple equivalent circuit of Fig. 3 is satisfactory for this analysis.

To be continued

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Intermodulation at the amplifier-loudspeaker interface

Part 2: Causes/how to avoid it/measurements on four types of amplifier circuit

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The effect described is but one of the numerous phenomena affecting the quality of low-frequency sound reproduction. It does not seem probable that its distortion could be dramatically higher than the measured SMPTE – intermodulation distortion of the amplifier, unless protection circuitry malfunctions. However, the theory presented may explain some of the subtle differences in the sound quality between different circuit topologies having otherwise equal standard measurement data. Noting that most valve amplifiers have basically a high open-loop output impedance and employ moderate amounts of feedback (the situation is the inverse for many solid state amplifiers), the theory may also explain some of the audible differences of these amplifiers.

The analysis of part 1 shows that the loudspeaker reflects back to the amplifier signal which may be of the same order of magnitude as the original drive signal. The situation is worse when the open-loop output impedance of the amplifier is comparable to, or greater than, the specified load impedance.

Inside the feedback loop, the amplifier must now handle two simultaneous large signals – the original drive signal and the loudspeaker reaction signal. If the amplifier has any internal non-linearities, these two signals may interfere and produce intermodulation components with each other. As the input signal is normally composed of a full frequency spectrum, but the loudspeaker-generated reaction consists predominantly of frequency components near the cone resonances and crossover filter resonances, the nature of this distortion is to add coloration to the sound. In addition, the positive maxima shown may cause unwanted clipping near amplifier maximum output power.

The basic reasons for the distortion are that (a) the loudspeaker does not simply consume energy; it also stores and returns it. (b), Although the closed-loop output impedance of the amplifier is apparently very low, it is not a true physical impedance as it has been generated by feedback. The feedback, in turn, forces the loudspeaker reactive current to cause a corrective signal which circulates around the feedback loop. (c) In the internal non-linearities of the amplifier this signal will intermodulate with the forward signal to produce a change in the spectral composition of the distortion products.

The two basic characteristics affecting the magnitude of this distortion are the open-loop output impedance and the amount of feedback. The dependence is fundamental, i.e. if one or both of these characteristics is brought to zero, interface intermodulation will not occur. The effect increases with feedback if the feedback is small or moderate say, below 20 dB. Above that, increasing feedback will no longer increase distortion. Also, it is generated in the internal non-linearity of the amplifier. As it is basically a low-frequency effect, the stage where the non-linearity is situated in the forward path is immaterial.

The above analysis requires sufficient linearity from the amplifier for the transforms to be valid. In high-quality audio amplifiers this condition is usually met in the normal operating range of the unit. However, a large reaction signal can cause the amplifier to enter a region of severe non-linearity when operated in the vicinity of its maximum output power. The need of a non-linear analysis is indicated in this case.

We propose the following general definition

Interface intermodulation is a form of distortion in a feedback two-port network, caused by non-linear interaction between the input signal of the two-port and a signal externally injected to the output port propagating into the input via the feedback network.

This general definition is specifically used in sound reproduction equipment to denote the distortion caused by the energy stored or generated in the loudspeaker system re-entering the output of the power amplifier.

Measurement

It is possible to measure interface intermodulation by using normal distortion measurement methods. In this case the standard output loading resistor is replaced with a simulated reactive load or with a real loudspeaker. In many cases the measured distortion is increased and the spectral composition of the distortion products changes. However, in the real-world situation, a set of standardized loudspeaker loads would be needed and, because of the frequency dependencies of these loads, it would be necessary to resort to swept CCIF-type difference tone measurements. This tedious procedure can be replaced by a simpler universal method described below. The loudspeaker reaction can be simulated by letting the amplifier operate on a forward signal, while injecting a backward signal to its output. If interface intermodulation is generated, it will manifest itself through intermodulation products between the two signals appearing at the

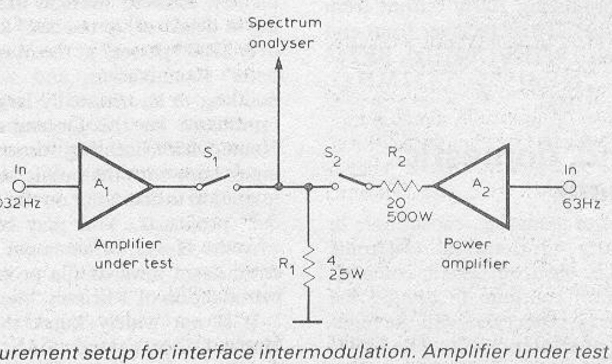

Fig. 8. Measurement setup for interface intermodulation. Amplifier under test A_1 is fed by audio frequency signal while high-quality high-power auxiliary amplifier A_2 delivers a low-frequency signal. By alternately closing switches S_1 and S_2 both signals are adjusted to have same power level in load resistance R_1. After closing both switches, intermodulation products are measured with a spectrum analyzer and referenced to the audio frequency signal. Numerical values shown are for the tests detailed in text.

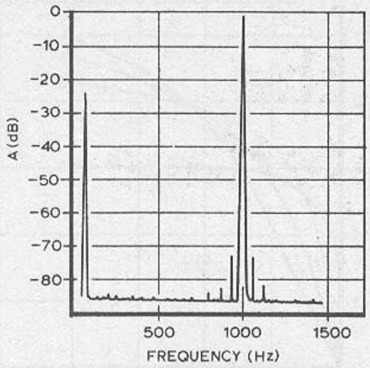

Fig. 9. Typical measurement result from high-quality commercial power amplifier using the method described in text. Note how the 63 Hz signal has been attenuated 24 dB by the feedback. Interface intermodulation in this case was 0.038%.

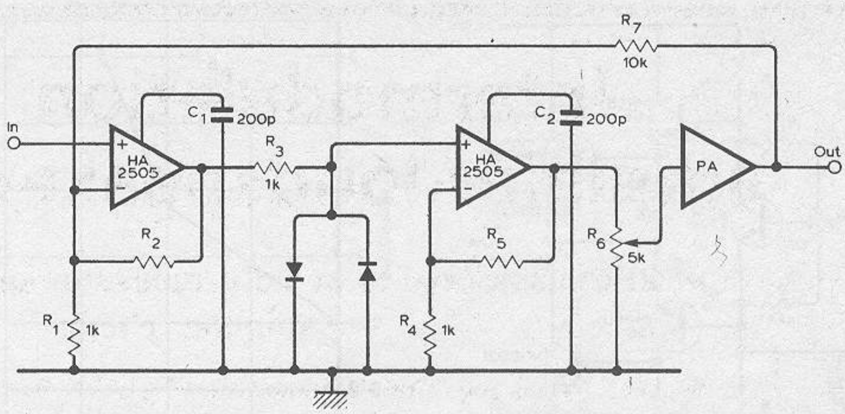

Fig. 10. Circuit used in the distortion measurements. Operational amplifiers HA2505 form the driver stages, and diodes constitute the dominant non-linearity. Various output sections PA are shown in Figs 11 to 14.

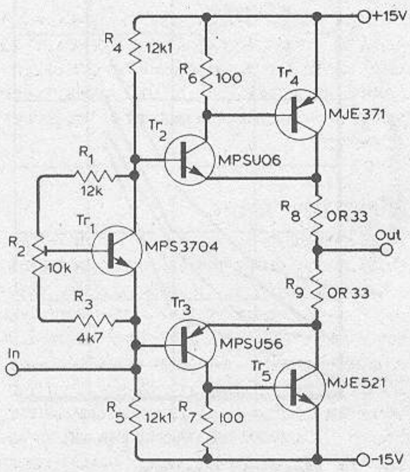

Fig. 11. Compound-stage circuit (A). Quiescent current 100 mA, open-loop output impedance 0.9 ohm.

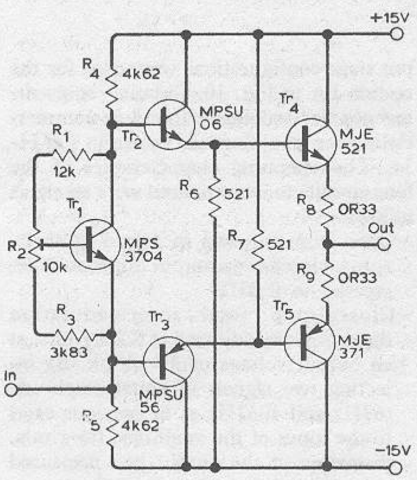

Fig. 12. Complementary double emitter-follower circuit configuration (B). Quiescent current 500 mA, open-loop output impedance 1.2 ohm.

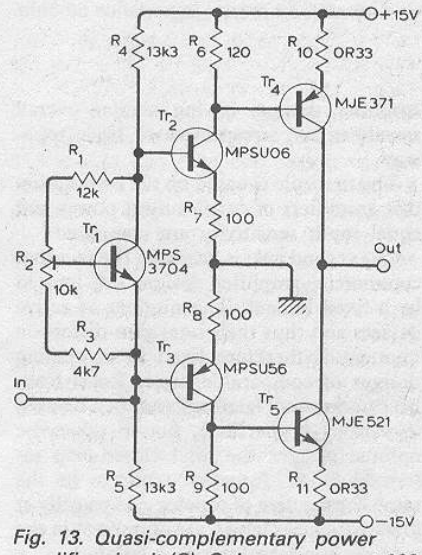

Fig. 13. Quasi-complementary power amplifier circuit (C). Quiescent current 100 mA, open-loop output impedance 2.7 ohm.

output. The measuring procedure is thus a variant of the two-tone difference-frequency method. In real life there is a dependence between the forward and backward signals. In this method, these signals are independent, to facilitate measurement. However, as far as the generation of intermodulation in the amplifier is concerned, this does not change the physical phenomenon considered.

A proposal for a measurement method is depicted in Fig. 8. The procedure is

1. Switch S_1 is closed and S_2 opened. An audio-frequency sinusoidal signal is connected to the input of the amplifier under test A_1 and is adjusted to yield a desired output level to a specified load resistance R_1.
2. Amplifier A_1 output is disconnected from the load R_1 by opening switch $S_1$1. A low-frequency sinusoidal power source A_2 is connected to the load by closing switch S_2 and is adjusted to yield the same output level across load R than in step 1. Note: power source A_2 has to have sufficient internal resistance R_2 so as not appreciably change the apparent load of A_1 when switch S_1 is closed. This power source must also have sufficient power output. A safe rule is that the rating of the power source is five to ten times greater than that of the amplifier under test.
3. Both switches S_1 and S_2 are closed, with both output signals being fed simultaneously to the load. The intermodulation products between the two signals are measured across the load by using a spectrum analyser or an intermodulation distortion analyser.
4. The r.m.s. sum of all intermodulation products (i.e. neglecting all harmonic components of the primary signals) is calculated and the distortion indicated as a percentage, referenced to the audio-frequency signal at the output of A_1.

The test frequencies used are in most cases not critical and can be selected to minimize the effect of such external disturbances as mains frequency hum. Their frequency ratio may be optimized so that the harmonic frequencies of the low-frequency signal do not coincide with the frequencies of the intermodulation products. Various frequencies and load resistances may be used in different countries, depending on mains frequency and standard loudspeaker impedances. The results reported were obtained using a load resistance of four ohms and frequencies of 63 Hz and 1032 Hz. A typical measurement result is given in Fig. 9, which shows the intermodulation spectrum generated.

Comparison of amplifier circuit topologies

The theory developed predicts that the amount of interface intermodulation distortion depends primarily on three basic power amplifier characteristics: Open-loop output impedance, amount of feedback, and closed-loop non-linearity of the circuit. The first two properties especially vary considerably among amplifier circuit topologies. To make a valid overall comparison of different topologies, all the circuits should have

- the same closed-loop gain
- equal closed-loop distortion, and
- same output damping factor, i.e. closed-loop output impedance.

These rules represent the market place reality of various commercially competing

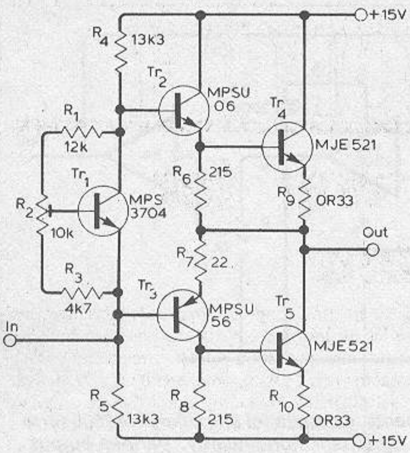

Fig. 14. Grounded-emitter complementary output circuit (D). Quiescent current 100 mA, open-loop output impedance 60 ohm.

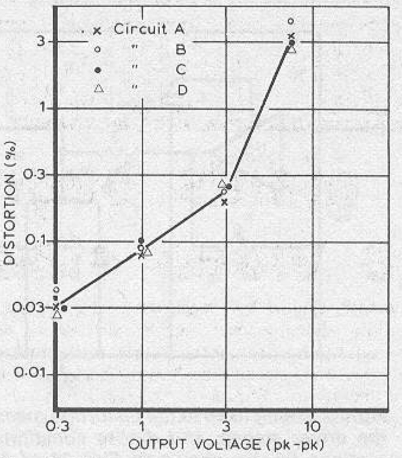

Fig. 15. Measured closed-loop intermodulation distortion for the various amplifier topologies after adjustment detailed in text.

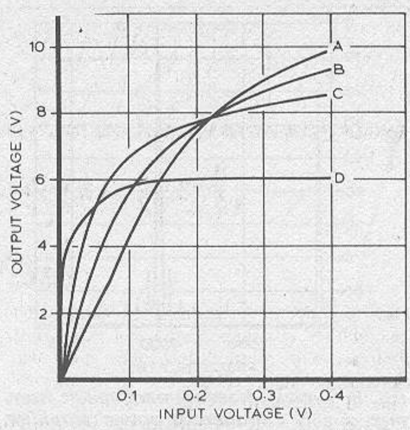

Fig. 16. Measured open-loop transfer characteristics of the various circuits after adjustment discussed in text.

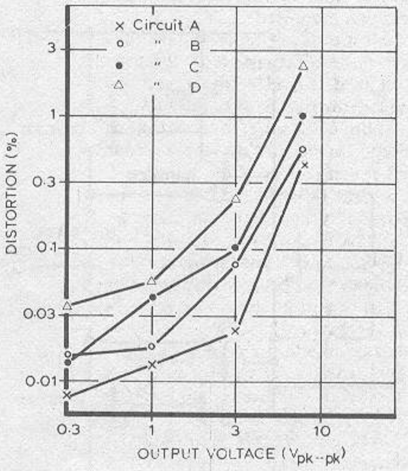

Fig. 17. Measured interface intermodulation distortion for the various amplifier circuit topologies. Results indicate clearly the roles of open-loop output impedance and feedback.

amplifier designs having similar overall specifications, irrespective of basic topology.

The first rule is based on the assumption that amplifiers of equal output power and equal input sensitivity are compared.

The second rule is based on the fact that commercial amplifier designs are limited by a fixed budget. The number of active devices and thus their total gain-distortion quotient is therefore fixed in competing designs of comparable price. Local feedback and overall feedback can then be used in various proportions, but in otherwise optimal designs the total closed-loop intermodulation distortion tends to be the same irrespective of topology, especially at low frequencies which are of interest in the case of interface intermodulation.

The third rule is dictated by the commercial necessity of having a reasonable or comparable damping factor specification, irrespective of topology.

The circuit shown in Fig. 10 was used for the comparative measurements. Diodes 1 and 2 create an artificial non-linearity, the magnitude of which can be adjusted by changing values of R_2, R_5 and R_6. The same resistors also set the open-loop gain and thereby the amount of overall feedback and damping factor.

In the measurements four different output stage configurations were used for the section PA in Fig. 10. Circuits representing popular topologies found in commercial power amplifiers are shown in Fig 11-14. The operating characteristics of the four circuits to be compared were set up as follows.

- Open-loop gain was increased until the r.m.s. closed-loop output impedance decreased to 0.20Ω.
- Closed-loop total intermodulation distortion was adjusted to 0.2% r.m.s. at an output voltage of 3V pk-pk. By injecting two signals of equal amplitude (63Hz and 1032Hz as in previous case) to the input of the amplifier, the r.m.s. distortion at the output was measured using a resistive 4Ω load and referencing the distortion to the 1032 Hz signal.
- These two were repeated several times in iterative fashion, as a change in the open-loop non-linearity affected the effective amount of feedback and thereby the output impedance.

In all the measurements, it was made certain that the intrinsic non-linearities of the various output circuits were negligible, as compared to the logarithmic non-linearity of D_1, D_2 in Fig. 10.

Figure 15 shows the measured closed-loop intermodulation distortion of the various circuits, while Fig. 16 shows the measured open-loop transfer characteristics of the circuits. After adjustment of the circuits, intermodulation measurements were carried out following the procedure outlined earlier. The main results are summarized in the table. Fig. 17 shows the measured values of distortion as functions of the output level. The results are in agreement with the theory presented. They also coincide accurately with earlier results measured for the same circuits using a constant value of feedback in the comparisons[2].

The results demonstrate clearly the role of the open-loop output impedance of a power amplifier in the generation of interface intermodulation distortion, the various amplifier topologies differing with each other by almost two decades. However, you must not draw far-reaching conclusions of the general usefulness of the various output circuits tested. There may exist ingenious ways to modify any of the topologies so that they will satisfy criteria for low interface distortion. Furthermore, the circuits seem to differ considerably in

continued on page 55

Summary of measurement results and conditions

	A compound	B grounded collector	C quasi-complementary	D grounded emitter
Interface distortion at 3V [%]	0.005	0.01	0.1	0.2
Open-loop output impedance [Ω]	0.9	1.2	2.7	60
Open-loop gain [dB]	33	36	43	70
Feedback [dB]	13	16	23	50
R_2 [kΩ]	1	1.5	3.2	1000
R_5 [kΩ]	42	36	32	13

General conditions for circuits: closed-loop gain 20dB; closed-loop output impedance 0.2Ω; closed-loop intermodulation distortion (CCIF) 0.2%; interface intermodulation distortion shown at output level of 3V pk-pk.

continued from page 44
their distortion behaviour close to clipping. Although these questions would be of great interest, they are not discussed as the purpose of this article is only to illustrate the basic theory.

The analysis and measurements show

- a loudspeaker, being reactive by nature, is capable of storing much of the energy it receives from the amplifier.
- this stored energy will be reflected back to the amplifier output terminals.
- the closed-loop output impedance of an amplifier is normally very low, but the open-loop impedance may be several ohms. To damp the reflected signal, feedback will generate a correction signal within the amplifier.
- the signal in the forward path of the amplifier thus consists of two components; the original input signal and the loudspeaker reaction signal, both of the same order of magnitude.
- these two signals may interact in the non-linearities of the amplifier, generating intermodulation products between the two.
- this distortion, termed interface intermodulation, will be most prominent at low frequencies where the loudspeaker reactive load is largest.

Amplifier design rules to avoid interface intermodulation

The output should provide a low open-loop output impedance to adequately attenuate the loudspeaker reaction signal so that the need for a feedback-generated damping is minimized.

Heavy overall feedback should be applied with caution.

- the susceptibility of the amplifier to interface intermodulation can be measured by using a modified difference-tone method, where one of the signals is injected to the input and one to the output of the amplifier. To create conservative worst-case test for this effect, the latter signal may be increased to equal in power the rated output power of the amplifier.

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