Slewing Induced Distortion in Audio Amplifiers: Part I

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THIS STUDY BEGAN as a general investigation into forms of transient IM distortion in op amps in general, and in IC op amps in particular. Since so very much has been written in the past few years on transient intermodulation distortion (TIM) in audio amplifiers[1-11, 15, 17]--most particularly the power amplifier--my original object was to seek some answers to which I could attach numbers *and correlate these with what I heard.*

Many of these writings damn IC op amps as the bane of the quality sound we all seek and treasure. On the face of some of it, one should dismiss the use of op amps in an audio signal path as something abhorrent, to be avoided at all costs. And indeed, some of the correspondence I receive doesn't just suggest this, it virtually demands it. I'm sure we have all read more than one equipment review which has mentioned "IC" or "transistor" sound, the harsh, hard, gritty stuff that grates the sensitivities.

But, if we sit back and reflect on the overall recording-to-reproduction system (of which our own end is only a part), we can see some obvious inconsistencies. We all know a recorded signal goes through many, many amplifiers before it reaches our ears and most of them are beyond our control. Consider an obviously well recorded example of today's releases and I think we can all agree the sound *can* be very good. And solid state amplifiers are used almost exclusively in the recording process.

Many console manufacturers use design concepts based largely on op amps, of either IC or modular variety. Some have excellent track records, while others do not. So great a number of companies use IC op amps in their products that the sheer numbers as well as the design differences (even if all details were available) would prevent us pinpointing which of these sound good and which sound bad.

Some attempts have been made to identify the "solid state" sound, to use an overworked and undefined term, most notably the Hamm paper[5] which appeared in the AES *Journal*. Hamm condemns solid state amplifiers by alleging that they sound hard when overloaded, due to their generation of an almost purely odd harmonic distortion product structure. While it is not my aim in this particular installment to get into the issues of the sound-during-clipping phenomenon, I hope to deal with them in the near future.

One point which Hamm's studies stress, however (and one which is probably familiar to us all), is that odd harmonic distortion in audio amplifiers is painfully obvious to the ear. This point ties in quite well with the subject of this article, the control of the slewing distortion mechanism, which produces odd harmonic distortion, *inherently.*

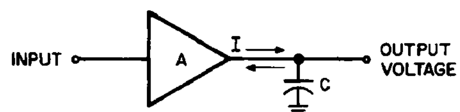
I hope the results of this study will clarify the use of IC op amps in audio, to a point that the reader will believe they don't automatically generate problems, can be used with confidence, and are highly predictable in both measured results and *listening quality.* Since the published correlation of measured results with sound quality is so woefully lacking in the audio field, I suspect any progress at all in this area will be more than welcome.

A General Look at Slew Rate

Slew rate limiting can occur at virtually any point in the audio chain, but is most likely to occur at points of maximum voltage swing, where the required rate of change is greatest. The limitation comes about due to a fundamental voltage/current relationship in capacitors as illustrated in block form in Fig. A.

Here an audio amplifier is represented by the symbol A. Capacitor C, which in practice could be either an integral part of the amplifier or an external load capacitance, is electrically connected across the output of the amplifier. Thus it sees the full output volt-

FIG. A



SR = slew rate = maximum output voltage rate of change, in V/µS (µsec.) or V/S

in terms of circuit parameters,

SR = I/C where I is capacitor charging current (amp) and C is capacitance being charged (Farads). Yields SR in terms of V/S (divide by 10^6 for V/µS).

Example: I - 1mA = 1×10^{-3},
C = 0.01µF = 1×10^{-8}
SR = 0.1V/µS

Fig.A: General representation of slew rate limiting.

FIG. B

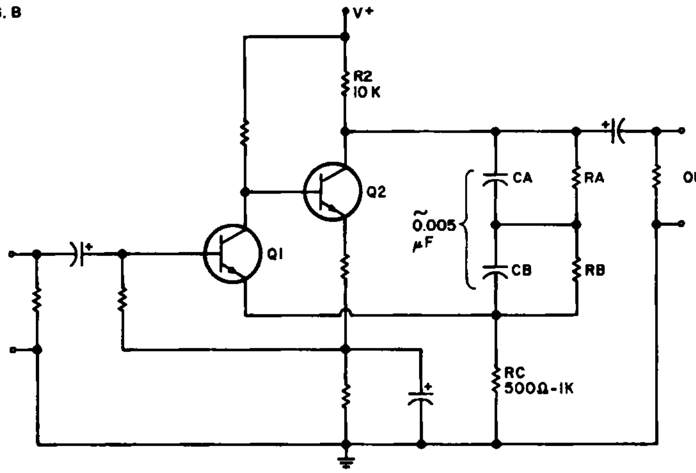
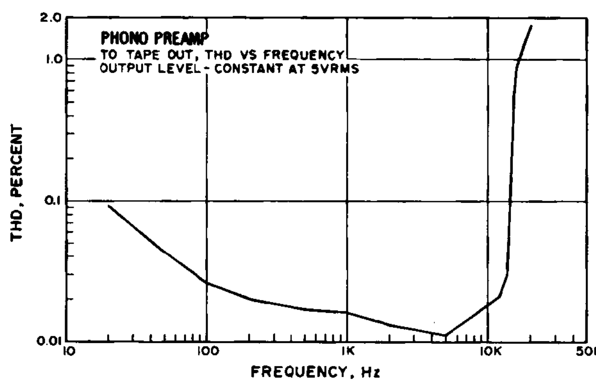


Fig.B: Classic two transistor RIAA preamp.

FIG. C



age swing. It is a fundamental circuit relationship (*regardless* of the type of active devices used) that the maximum output current available from this amplifier will determine the maximum rate of voltage change which can appear across this capacitor.

This can be stated mathematically quite simply. For a current I, the rate of change, or slew rate (abbreviated SR), is simply SR = I/C. With I in Amperes and C in Farads, SR is in units of Volts per second (V/S). More commonly, SR is given in V/µS, as 10V/µS. 10^7V/S would be equivalent to 10V/µS.

A constant current into a fixed value capacitor will result in a linear, or ramp-like waveform of voltage. Audio signals are not ramps, or triangular waveforms, to be sure, but for a sine wave the maximum rate of change occurs at the zero crossings. This factor is the basis of the so-called "full power bandwidth" (abbreviated fp) which relates SR and a maximum full amplitude sine wave signal. This relationship is simply

$$fp = \frac{SR}{2\pi Eop}$$

where Eop is the peak output voltage. Thus the two parameters are directly related, and slew rate can be expressed in terms of fp as

$$SR = 2\pi Eopfp$$

We will do well to remember that fp is by definition the beginning of *complete* slew rate limiting, and generally will be accompanied by 1-3% THD. The desired output sine wave under slew rate limited conditions will in actuality more nearly resemble a triangular wave, due to the rate limiting effect.

The above is about as complete a discussion as you will be able to find in many references on the subject, as if to imply that is all there is to it. Nothing could be further from the truth. In fact, slew rate, or *slewing induced distortion, is the single largest distortion mechanism in solid state audio amplifiers today.*

This is a little appreciated fact, and is evident by both the dearth of published material on it in audio literature, and the number of product specifications which neither recognize nor define it. Further, recent comments to this writer by individuals seemingly knowledgeable in audio even indicates some confusion among professionals about how to measure slew rate. Adding to the confusion, a rash of recent articles speak of transient IM distortion as though it were something entirely separate from slewing induced distortion when in actuality it is not, in many cases. I hope this article will clear up some of this confusion and provide the reader with a convincing overview of the magnitude of slew induced distortion problems in audio.

As I suggested earlier, many discussions of amplifier slewing rate imply (or flatly state) that it is a mechanism which suddenly produces distortion when the full power bandwidth point is reached. This is simply not true, unless you accept the premise that distortion is only significant when it reaches 1%. In reality, the approach of the slewing rate limit of an amplifier can produce easily measurable and significant distortion products at frequencies as low as 1/10 or 1/5 of the full power bandwidth. I have personally observed this phenomenon on many, many samples of different IC op amp types, as well as more conventional preamp circuits and power amplifiers.

Many people evidently see no reason to get excited over the slew rate flap; and simply do not believe it is a major audible defect. I can recall one design engineer (who claimed to be an audiophile) who said any distortion above 10kHz was not worth worrying over, as it was by definition inaudible. This is naive optimism.

Consider the presence of two high level, HF tones with a close frequency spacing. Their LF intermodulation product pops out down in the bass or mid frequency region, and is *highly* audible. And this is exactly the way slewing induced distortion works on two-tone IM tests, producing strong LF intermodulation components. You should also consider brief HF transients as well as tones. If the slewing rate of an amplifier is pushed (not even exceeded, necessarily) even momentarily, intermodulation will occur and will probably be audible. It can also occur (indeed, is most likely) with high level *supersonic* signals which will cross-modulate down into the audible region.

These audio circuit ills can be dealt with in many ways, which we will discuss later in this series. First, however, let's examine cases which give rise to slewing problems that I hope will give you a deeper appreciation of the severity of the problem. In other words, if you are still unconvinced, let me cite some graphic examples to make a believer of you.

One case in point is the familiar RIAA phono preamp stage. I would venture a guess that a great many of them suffer (at least potentially, if not in fact) from slewing induced distortion, simply because they cannot fully charge their own equalization capacitors. Fig. B, a simplified schematic of the classic two-transistor feedback pair as applied to RIAA phono preamp use, will demonstrate this.

At high frequencies, equalization feedback capacitors CA and CB appear as a single equivalent capacitance to ground in series with RC. A typical (lumped) value of capacitance for CA-CB will be in the range of 0.005µF, with RC in the range of 500 Ohms to 1k. Collector or load resistor RL may be about 10k.

Much is written about phono preamp overload phenomena, to the point that supply voltages are being run at 30 to 40V, to handle high cartridges outputs, with the object of yielding a 1kHz output of 10V RMS or so. *[The Technics SU 9600 reportedly uses 136V in its preamp. See* Wireless World*, Nov. '76, p.41.--Ed.]* Here we uncover the inconsistency of this thinking, however. If such a stage must handle 10V at 1kHz, is it not reasonable to expect it also to do so at 20kHz? I would think so, but it is just about impossible with typical circuit values. For instance, a 10V RMS level is 14V peak, and at 20kHz the required slew rate is

$$SR = 2\pi Eopfp$$
$$= 2\pi(14)(20000)$$
$$= 1{,}750{,}000 \text{ V/S (or 1.75 V/µS)}$$

since SR = I/C, and using a C of 0.005µF the required charging current is

$$I = (SR)(C)$$
$$= (1.75 \times 10^6)(5 \times 10^{-9})$$
$$= 8.75\text{mA}$$

Now 8.75mA by itself may not appear to be an insurmountable limit, but this stage will be generating about 1% distortion at 20kHz for this level of current. If, as a safety factor, we raise the current by a factor of 5, the Q2 stage will be running at over 50mA which will get the distortion down, but certainly creates other problems, as Q2 will be dissipating nearly a Watt. I believe it should be obvious that the circuit values of Fig. B will not allow 10V RMS @ 20kHz.

One might be tempted to reduce capacitor size to gain relief, but this will in turn raise resistance proportionately, creating noise problems. In a limit-

FIG. D

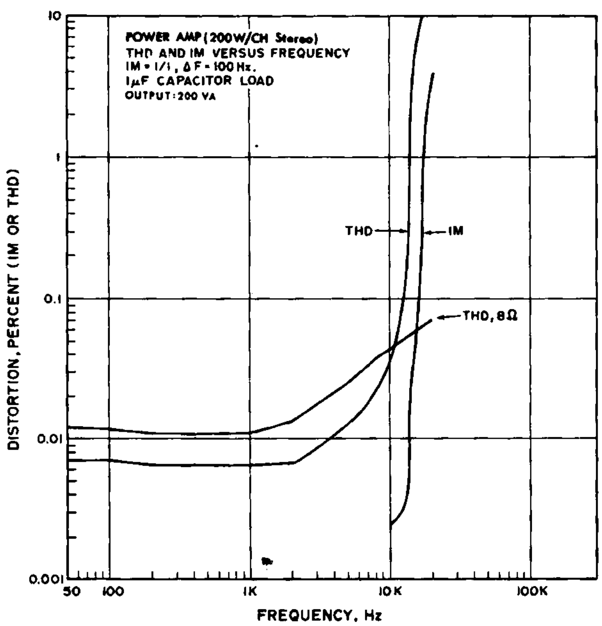


FIG. E

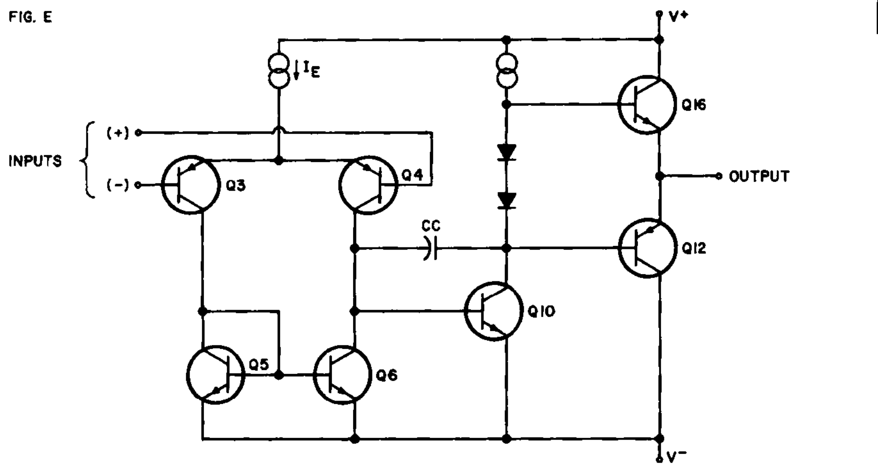


Fig.E: Simplified IC op amp (301 or 741).

ed look into this problem I have noted a great many examples of this type of circuit which suffer from the same fundamental ill. Graphically the distortion generated by such a stage is shown in Fig. C, the THD performance of a phono preamp circuit in a currently popular preamp. This circuit is severely slew limited, and although capable of 10V @ 1kHz would only produce 5V @ 20kHz, the level at which the data was taken. The measured slew rate was 0.65V/μS, which agrees reasonably well with the bias current and capacitance values.

My point here is that although the amplifier circuit *without* capacitor loading may have a high (and adequate) slew rate by itself, it simply cannot supply sufficient output current to drive the RIAA feedback network to full output at high frequencies. The result is gross distortion at these frequencies, in terms of both THD and IM.

The example in Fig. C is by no means an isolated case; a great many widely publicized "high performance" IC phono preamps suffer from similar ills. Viewed in this light, it is somewhat ludicrous to tout a preamp circuit for high overload, high undistorted output capability at a frequency of 1kHz, if it cannot also produce a similar output at 20kHz. As we will see in the course of this discussion, one of the yardsticks which can detect the presence of slew induced distortion is a high resolution measurement of THD through the audio band up to 100kHz or so.

If the distortion level at full voltage output at 20kHz is low (on the order of 0.01%), and within a factor of 2 or so of the 1kHz distortion, the amplifier is likely to be free of slew induced distortion, and thus of the accompanying intermodulation effects. On the other hand, if the amplifier cannot produce full voltage undistorted output at 20kHz, it is *highly* likely to be suffering from slew induced distortion.

Before departing the subject of phono preamps and their susceptibility to this type of distortion, I should mention that Holman's paper, "New Factors in Phonograph Preamplifier Design,"[16] *[see also* The BAS Speaker, *Nov., Dec. '75 and Jan. 76.--Ed.]* gives an excellent discussion of testing methods for phono preamps to detect this form of distortion. This paper is required reading for the audiophile.

The signal path is full of such things as line amplifiers, tone amplifiers, and equalizers. This generally very broad area of signal processing can utilize many different forms of circuit technology, but some of the most efficient realizations come about with the use of IC op amps. A major portion of this article will deal with a study of slew distortion in IC op amps, with conceptual results which are equally applicable to preamp, line amp, or power amp--indeed *any* audio amplifier. First, however, let me comment on slewing in the output stage of the power amplifier.

The final power output stage of amplifiers is acutely sensitive to slewing induced distortion, for several reasons. First, the voltage swing is at its maximum level, because of the high powers involved. A 200 Watt into 8Ω amplifier, for instance, must swing about 56 Volts peak for full output. In terms of the full power relationship, this would require a slew rate of 7V/μS for 20kHz reproduction. Since this rate would produce a 1% distortion at this frequency, a slew rate for this power level should obviously be several times 7V/μS.

Ideally, a slew rate which will result in minimum slew induced distortion is on the order of 0.5 to 1V/μS *per peak output Volt*. In the case of the 200 Watt amplifier this would imply a 56V/μS slew rate, which gives you an idea of the severity of the problem. (The rationale behind the 1V/μS per peak Volt is explained in detail below.)

Given a power amplifier which could slew at a rate on the order of 50V/μS (which is no mean feat, by the way), it should ideally be capable of this slew rate *into varying loads*. In practice it is one thing to build an amplifier which can slew at 50V/μS into a resistor, and quite another thing to do so into a reactive load.

Considering a 1μF capacitor as an example, a 50V/μS slew rate would require an output current of 50 Amperes. It is quite easy to see how difficult this becomes when the accompanying high voltages which must simultaneously be handled are considered. Practical amplifiers under these conditions will slew at rates closer to 10V/μS.

Thus, we should clearly understand that a power amplifier can be slew limited from one of two sources: either its own internal compensation capacitance(s) or from reactive loading which causes the protection circuitry to activate, so limiting the output slew rate. *Either* of these two conditions will severely distort high frequency waveforms causing serious intermodulation products, which will result in poor reproduction. I do not ask you to accept these statements as factual on faith; they may be readily demonstrated by measured data.

Fig. D is a graph of measured distortion, in the form of both THD and two tone 1:1 HF IM. These are plotted in percentages as a function of driving frequency. This is also justified in the IM case, as the difference frequency of the two tones is less than 100Hz (1% or less). The conditions are a 1μF pure reactive load and a constant 200VA output voltage level. The amplifier is a popular 200 Watt per channel unit. Slew rate into this capacitance was measured at 5V/μS which corresponds to an fp of 14.2kHz.

In the case of THD, we may note that it is 2% at the fp point but has begun to rise far below this frequency. I include also a reference curve of an 8Ω resistive load THD at a 200 Watt level which by contrast is less than 0.1% even out to 20kHz. Some of the gentle upward slope in this curve is probably slew limiting due to the amplifier's internal compensation, although some output stage crossover conduction spikes are also present. Note that both curves break away from the LF plateau at a fairly low frequency, about 2kHz.

The IM curve gives convincing evidence for the slew induced distortion argument, demonstrating that low frequency products at levels of several per cent can be produced simply due to slew limiting. Here the two-tone pair is swept in frequency from 10 to 20kHz, and the general shape of the IM distortion rise is remarkably similar to the THD rise, although less sensitive. This example indicates a correlation between the two methods of testing for this particular distortion mechanism. Further evidence of this correlation is indicated in the op amp data to follow.

The two-tone 1:1 HF IM test is a particularly appropriate one for power amplifier tests, for several reasons. Although not quite as straightforward as a THD test it is still relatively simple to implement compared to the sine/square method or noise transfer test, and gives reasonable sensitivity to this distortion mechanism.

Extended range THD tests to 100kHz or more at full output voltage level with high resolution equipment can also reliably indicate SID (and consequently TIM) but are not desirable in power amps for two reasons. First, they will cause a very high stress on the output stage due to storage time effects, with even possible destruction in some cases.

Second, as pointed out by others,[12 17] the harmonic distortion figures measured will most likely be erroneous, due to the natural rolloff with frequency of the amplifier. Two-tone IM tests up to 20 (or 30) kHz do not stress the amplifier output transistors nearly as much, and since the products being measured are reflected downward in the audio spectrum there is no loss of accuracy due to rolloff.

The two-tone technique is not new, but has not been used to any substantial degree in the U.S., particularly in amplifier testing. I advocate the adoption of the 1:1 two-tone swept HF IM test as a standard technique for audio amplifiers, at both power output and signal processing levels. Future equipment tests in this publication will utilize the technique, and work will also be initiated on an instrument suitable for home construction.

Some comments are appropriate here on the overall problem insofar as power amps are concerned. The more insight we gain about the power amplifier slewing problem, the more staggering the situation appears to be. Some of these problems, for example the high fidelity drive into reactive loads, seem almost insoluble with presently available technology. This article cannot really hope to completely address the power amplifier problem, and will not attempt to. What we hope for is an overview of the mechanism of slewing induced distortion in its general form, and a fairly definitive picture of how it can be measured (and controlled) in low level amplifiers, particularly IC op amps.

This article is the first of an ongoing series on audio amplifier distortion; future installments may perhaps more completely treat the power amplifier case. Many of the points and measurement techniques we will discuss are equally applicable to low level and power amplifiers, and as the narrative progresses this will be underscored.

The Op Amp Slew Limiting Mechanism

Like so many other things, IC op amps are used in audio in ways that are good and bad. The outcome depends upon both the user's viewpoint and his/her level of understanding. Incomplete understanding of the problems which arise in effectively applying IC op amps to audio is in itself understandable: the range of available devices is staggering. Yet, certain general principles govern effective use of these devices, and designers should at least understand these fundamentals. I attempted to examine some of these problems in my AES paper,[23] which I later expanded into one chapter of my first book[24] and the audio volume derived from it.[25] However, this material is no longer adequate, for two major reasons.

First, many new, improved devices have appeared since they were published, and second, slew induced distortion warrants a much more extensive discussion. SID in audio circuits (particularly in op amps) is probably the *only* major distortion mechanism, if the design is a reasonable one. This may sound startling, but the cases to which this statement applies are probably more numerous than many people suspect. I am quite sure this type of distortion causes many audio circuits to sound bad, due to the nature of their distortion products.

However (at least in IC op amps) SID can be dealt with, using appropriate design techniques and fairly simple test procedures, even with a minimum of equipment. Op amp slewing rate problems can indeed give rise to TIM, but I suspect that an amplifier truly free of SID will never have TIM. However, I am not sure that a so-called "TIM-free" amplifier cannot have slewing problems. If you consider the slew rate into reactive loads, the amplifier will, of course, generate IM as I have shown above.

In the testing part of this study, I examined IC op amp slewing by closely measuring the behavior of a large number of devices. Out of this I developed a predictive analysis technique showing whether a given device would be free of SID for a given application, as well as several general criteria for slewing specifications, for devices as well as for circuits.

Slew induced distortion comes about because of the nature of an op amp's design. It is a phenomenon that can probably never be completely eliminated, but it can certainly be minimized to manageable proportions. To understand the basic mechanism, refer to Fig. E, a much simplified diagram of a 301 or 741 IC op amp (this general circuit is equally applicable to many power amplifier designs as well).

This amplifier consists of an input differential pair Q3-Q4 fed by a constant current source, I_E. The Q3-Q4 outputs are fed into a current mirror comprised of Q5-Q6, which converts the differential output to single-ended form at the base of Q10. Q10 is the second voltage gain stage of the amplifier, and its output collector voltage swing is buffered by transistors Q12-Q16 before appearing at the output terminal.

This amplifier has an overall low frequency voltage gain of 100dB or more, and is compensated for unity gain stability by Cc, a Miller integrating capacitor connected around voltage gain stage Q10. This capacitor causes the voltage gain of Q10 to decrease at high frequencies, yielding the necessary gain/phase characteristics for stability.

Because of the very high open loop gain and the necessity for a stable closed loop under feedback conditions, the presence of Cc is a necessary requirement, at least for a general purpose op amp.

However, the presence of that frequency compensation capacitor has a very serious effect on the amplifier's speed, most notably its slewing rate, or output voltage rate of change ability. Examining the diagram you can observe that the right terminal of Cc sees essentially the full amplifier output voltage (Q12-Q16 being unity gain buffers). The voltage rate of change across this capacitor (which is, in fact, the amplifier's slew rate) is determined by the current into it, which can only come from Q4 or Q6. The maximum current Q4 can deliver is I_E, under an input condition where Q3 is fully off; the maximum current Q6 can sink is again I_E, when Q4 is fully off (by virtue of the current mirror Q5, Q6). Thus the peak current into the capacitor is I_E, for either charge or discharge.

With simple capacitor current/voltage relationships, we can express the voltage across Cc, which is the circuit's slew rate.

$$SR = \frac{\Delta Eo}{\Delta t} = \frac{I_E}{Cc} \text{ Volts/second}$$

In a 301 or 741 amplifier, I_E is about 15µA, and Cc is 30pF. Therefore the slew rate is 0.5 Volts per microsecond. This means the amplifier can execute a full scale output swing from +10V to -10V (20V) in 40µS.

In terms of sine wave output signals, there is the aforementioned equation which relates SR to the full power (maximum p-p voltage) sine wave output frequency, fp. In the case of typical op amps, the specified peak output swing is 10V peak, for devices we will be discussing. For the 0.5V/µS slew rate mentioned above, the corresponding fp is then

$$fp = \frac{SR}{2\pi Eop}$$
$$\approx 8\text{kHz}$$

Doesn't sound too encouraging, does it? Actually in terms of a 20kHz full power bandwidth, the SR would be 1.25V/µS.

Now we may look more deeply into this slew rate limiting mechanism involving the amplifier's input stage. In practice op amps are intended to be used in conditions of an ideal 50-50 current balance in the input stage, a state where I_E splits equally between Q3 and Q4. Under such a condition the differential input voltage is *near zero*, recalling one of our fundamental axioms (see reference 24, chapter 1).

Under changing conditions of common mode input voltage, frequency, output level, loading, etc., this condition *must constantly be maintained, if the device is to function as an op amp.* If the input stage is *not* balanced, the input voltage is by definition not zero, *therefore it is not operating as an op amp.* This condition can (and will) occur when an input signal is applied which exceeds the amplifier's slewing ability. However, it can also happen to a lesser degree, for signals which require rates of change just below the slew rate. This would also be an example of an unbalanced condition (but less than 100%).

In the case of a square wave input with a rise time faster than the slew rate, the input stage will toggle back and forth between conduction states of Q3 and Q4, as the loop attempts to follow the fast square wave. Fig. Fa illustrates this effect. Note that during the slewing rate interval where the output is ramping to a new level, very large input differential voltages occur. Again, by definition, the amplifier is not functioning as an op amp during this slewing interval.

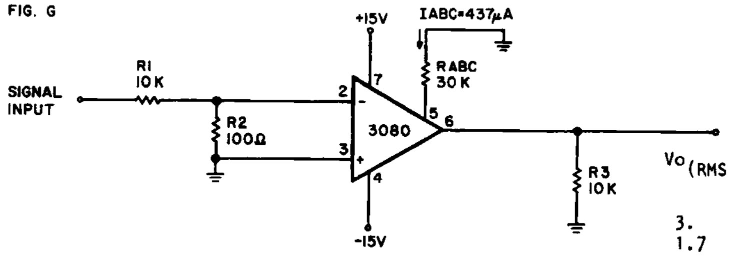
The effect of a sine wave input is shown in Fig. Fb. In the normal input/output example, the input voltage is reproduced as a reasonable replica (here on a 1-1 basis). But, when a frequency and amplitude combination is applied which exceeds the slewing ability of the circuit, the output becomes triangular rather than sinusoidal in shape, as you can see by the superimposed (dotted) proper waveform shape. If we raise the frequency, the amplitude would fall even further, and become more triangular in shape.

One highly interesting aspect of slew rate limitation in op amps is not always obvious, particularly if you just eyeball an output waveform on a scope. At the full power frequency, fp, THD will be on the order of about 1% (the exact number varying, due to different factors). You can see this 1% distortion rather easily by watching for the ramp-like slopes on an output sine wave.

But what happens below fp? The distortion does not just go away suddenly as you lower frequency. By contrast it can be significant at frequencies as low as 1/10 of fp. This is the real reason behind this article. SID can be subtle, and present in almost every op amp circuit.

If we examine Fig. E again and recall the statements concerning the balance of Q3-Q4, we can see that under slewing conditions Q3 and Q4 are operating far from ideally. Even at a frequency just below fp Q3 and Q4 will by necessity be swinging far beyond the 50/50 balance point. In fact they will alternate between near 100% and 0% conduction of I_E at the slew limiting frequency. With this situation, Q3 and Q4 will be generating *gross* amounts of odd order distortion, predominantly third harmonic. This is the best way to identify SID in THD measurements: watch for the appearance of third harmonic components of distortion.

FIG. G



$V_{O(RMS)}$	ΔI_O (% I_{ABC})	THD (%)
3.	97	10
1.7	55	2.7
0.95	31	0.83
0.53	17	0.26
0.3	9.7	0.09
0.17	5.5	0.07

Fig.G: Differential input stage model to illustrate distortion generated.

To illustrate the seriousness of the distortion generated in a differential pair as a function of current swing, I devised the test circuit of Fig. G. This simple circuit uses a 3080 OTA as a differential pair model which delivers an output current into R3, a fixed load resistor. The current set up in R_{ABC} is I_{ABC}, which for our purposes is analogous to I_E of Fig. E. The output current of a 3080 is $\pm I_{ABC}$, which is similar to the $\pm I_E$ output current of Q3-Q4 in Fig. E.

A signal input applied to the circuit at various levels allows the output to be measured from near full scale output current swings, downward. The results demonstrate the problem impressively. This should clearly illustrate the potential non-linearities of an op amp as it approaches its slewing rate limit. Even relatively small imbalances in the differential pair produce appreciable distortion.

The slew rate of a given amplifier *is in no way directly altered by feedback.* Slew rate is a parameter *independent of feedback,* and will measure the same whether tested open or closed loop. Actually to be completely correct, slew rate can only be measured open loop, since by definition causing an amplifier to slew by excitation with a fast step will open the loop momentarily, during the slewing interval. But feedback, regardless of whether it is 100% or 1%, will not change the basic slew rate, since slew rate is determined by I/C relations.

While Fig. G has illustrated the non-linearity of the differential stage open loop, the matter can also be demonstrated in a different manner, quite simply, and with little equipment.

Fig. H is a plot of the actual p-p input error voltage of a 741 op amp, operated in a unity gain inverter. This particular connnection is a convenient one, as the error voltage appears between the summing point and ground, thus is easily observed.

Although this error voltage is ideally zero, in a practical amplifier with a 6dB per octave open loop rolloff it will rise 6dB/octave with frequency. This rise will be quite predictable at frequencies below the slew limiting point.

In Fig. H the rise may be observed from 500Hz to about 7kHz, where the voltage begins to rise much faster. This rise signifies the onset of slew limiting, as the loop is forced to create much larger error signals to swing the input stage to greater percentages of output, in charging the compensation capacitor.

This test is not very sensitive, nor does it yield quantitative data. It does, however, quite simply demonstrate the non-linearity and abrupt deviation from predicted behavior associated with slew limiting.

Although specific designs of op amp input stages vary widely and take on many forms different from the simple one shown, a great many of them use bipolar input stages without emitter degeneration. It is this type of input stage which is most susceptible to the non-linear, voltage in/current out problem which causes high distortion under slewing conditions. The input transistors can be either NPN or PNP, but if undegenerated they will be highly non-linear away from balance, or during slewing.

To raise slew rate in an op amp, either I_E must be raised or Cc lowered. If the op amp is an externally compensated type, and is to be used at a high gain, Cc can be reduced, which raises slew rate in direct proportion. However, such a solution is not always desirable, or possible. For instance, if the op amp is used at unity gain, slew rate must be increased by other means.

FIG. Fa

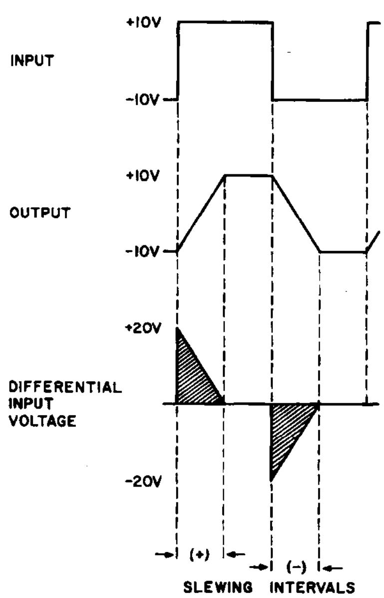


Fig.Fa: Response of op amp to a square wave input.

FIG. Fb

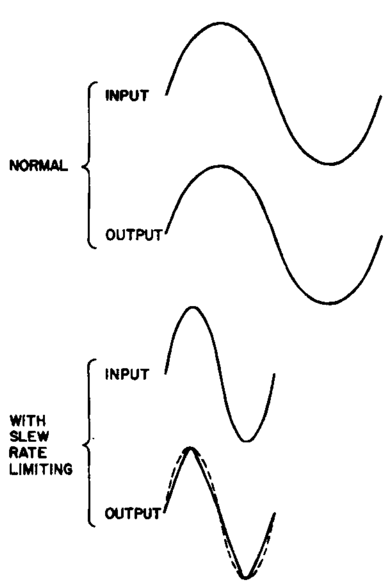


Fig.Fb: Response of op amp to a sine wave input below and above rate limiting.

FIG. H

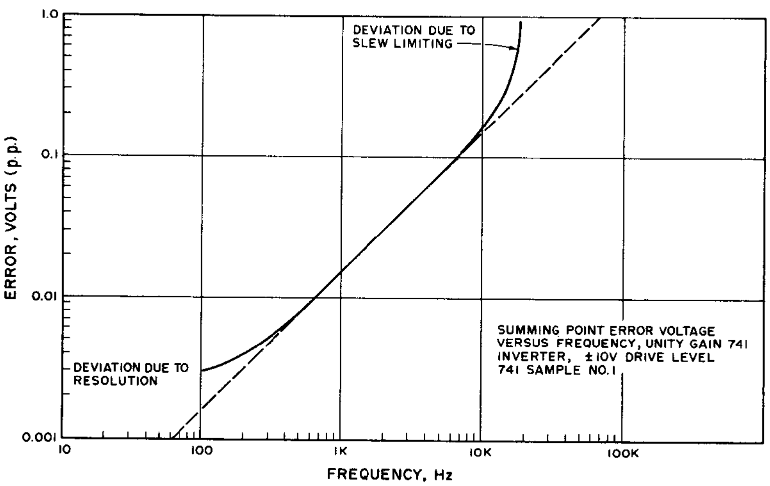


FIG. Ia1: NPN INPUTS

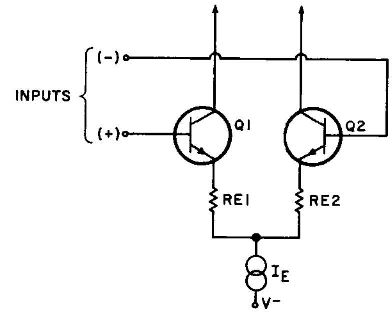


FIG. Ia2: PNP INPUTS

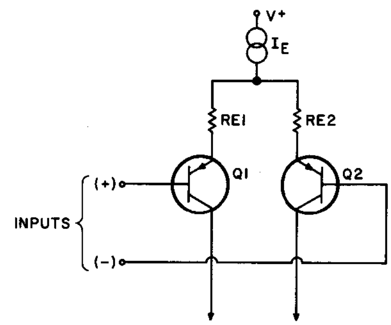


Fig.Ia: Emitter degeneration.

Some op amp designs attack the problem by various means of increasing I_E, which allows more current to charge Cc. This is usually accompanied by the use of emitter degeneration in the differential pair, to lower transconductance. This scheme is illustrated (in much simplified form) in Figs. Ia1 and Ia2. RE1 and RE2 are the emitter degeneration resistors, and Q1-Q2 the input differential pair. Since the use of RE1 and RE2 lowers the stage gain, I_E can be raised, and thus slew rate is raised.

Another method uses input stage devices with lower basic transconductance: FET differential pairs, for instance, illustrated in Fig. Ib.

In Fig. Ib, a P channel JFET pair is the input setup, while in Ib2 PMOS devices are shown. Both techniques lower input stage gain directly, because FETs have lower basic transconductance than do bipolars. Thus I_E can be raised, increasing slew rate.

Another technique uses "slew enhancement" which dynamically increases I_E during the slew interval only, as illustrated by Fig. Ic.

Here Q1-Q2 are the differential amplifier pair which operate more or less conventionally, *for small signals.* At high slew rates additional current is forced by the cross-coupled arrangement, which "enhances" or raises slew rate. The performance of all of these means of slew rate improvement is discussed in detail in the testing phase of this study, coming next issue.

PREPRINTS ANYONE?

The editors of TAA are seriously considering offering preprints of the four articles in Contributing Editor Walter Jung's series, "Slewing Induced Distortion in Audio Amplifiers". Part two: Phase I Testing, Part 3: Phase II Testing and Part 4: Listening Tests along with this first part will, we estimate, run to approximately 40 pages of text and charts. For those who want to peruse this important series as a whole ahead of 1977 publication dates, the cost will be $16.50 postpaid.

Those who are interested please make checks payable to The Audio Amateur. If sufficient orders are on hand by April 1, 1977 we will proceed with the project. If not, checks will be returned. The preprints, if produced, will be mailed first class about April 20. Send orders to: The Audio Amateur - Preprints
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Peterborough, N.H. 03458

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FIG. Ib1: P CHANNEL J FETS

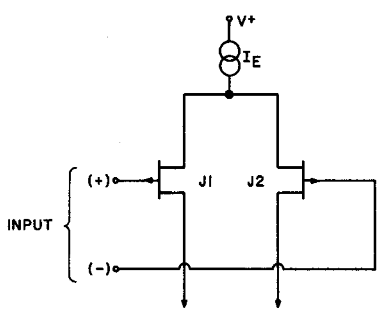

FIG. Ib2: PMOS

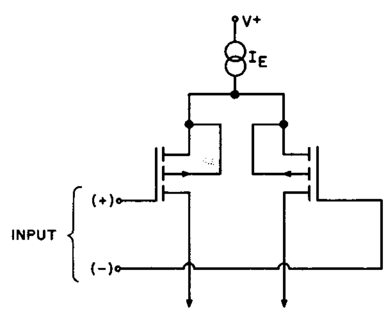

Fig.Ib: Lower transconductance input devices.

FIG. Ic

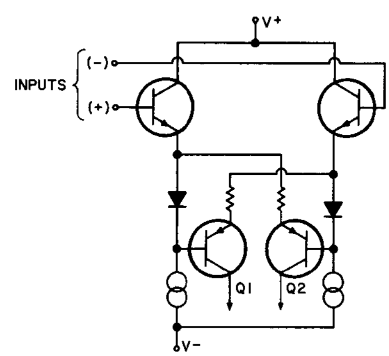

Fig.Ic: Slew enhancement.

THE NOT-QUITE-PASSIVE RADIATOR

spectrum. He designed a simple circuit (which varies daily, so far, and is apparently a long way from optimum) which applies maximum damping to the secondary radiator (shunts the voice coil) until the amplifier power applied to the primary radiator exceeds a pre-set point. He then raises the resistance across the voice coil to permit greater cone movement, thus providing a transient signal which is the sum of the output of the *two* cones.

A similar approach assumes that the first element of any transient is positive: that is, the wavefront is positive. If a diode/capacitor circuit is applied to the voice coil of the secondary radiator, it is possible to permit free cone excursion on the "downstroke," with limited rate of travel on the return, thus utilizing the secondary radiator only a small percentage of the time. Audible radiation would take place only when frequency was low enough to permit coupling between the two radiators, and then only when the signal was in the positive (ascending wavefront) portion of the curve. Strong percussives, then, might be reproduced with considerably more realism than otherwise possible.

It has been pointed out that the susceptibility of any speaker to this application can be easily tested; an important point, as some very low efficiency drivers are not suitable. Simply rap the magnet assembly while observing the cone, first with an open voice coil and then with a short (a 25¢ coin reaches most terminals). The difference should be quite significant.

To reduce duplicating experimentation, keep a log and work on only one of the several variables at a time. Remember a nearly infinite number exists of combinations of components which can be added to the circuit governing (or "reporting on") the operation of the not-quite-passive radiator.

If you decide to try this technique, good luck, be persistent, and please share your information.

THE FOLDED, STAPLED BASS HORN

Continued from page 13

the Calrad and Radio Shack 5" speakers, as they are inefficient and yield poor bass response. The speaker chamber can be modified to contain a pair of 8" speakers (cut off an inch of the cardboard at the throat for extra depth if necessary) or six 4" speakers, if desired.

This enclosure is designed to be both a quality low frequency speaker and a high quality midrange unit as well, with a useful range up to 3,000 Hz without the bother and expense of an additional crossover. The horn itself does not radiate much energy above 200 to 250 Hz: it's sole purpose is to couple the low-frequency mechanical energy of the small drivers to the surrounding air. The horn part of this enclosure is driven by the back side of the speakers and works to restore the low frequency energy which is normally lost through poor coupling.

The midrange frequencies are directly radiated from the front of the four speakers, which constitute a small array. Several developed mathematical models help explain why an array can be superior, in the midrange frequencies at least, to just one of the speakers by itself. Whatever the ultimate reason for this may be, the fact remains that speaker arrays are quite effective.

The 5" drivers are lightweight and rigid enough to respond accurately at midrange frequencies; however, this is certainly not the case at higher frequencies. The cones are simply neither lightweight enough or inertia-free to follow the amplifier signal faithfully at the higher frequencies, and should be crossed over to a high quality tweeter at a maximum upper limit of 3,000 Hz. Even casual listening can detect offensive intermodulation distortion in the higher frequencies by the severe ringing when playing a very complex source such as choral music. A single low-pass crossover coil however, will limit the speakers' response to the midrange frequencies where they perform quite well. (See next issue for a crossover circuit.--Ed.) The grouping of the four speakers becomes directional at frequencies above 1500 Hz, so that the ideal crossover would be below that frequency.

Any good tweeter or tweeter-midrange will help out in the high end, although the best complement by far for this horn is a version of the Heil driver which you can construct for around $25. The details will appear in the next issue.

The speakers will be ideally suited to a lower power amplifier than what is commonly in use today, as the enclosure increases the efficiency of the speakers tremendously. A twenty watt amplifier should be sufficient to drive the speakers to very high volume levels while still having enough reserve to do permanent damage. The enclosure works best in a corner, with the mouth about nine inches from the wall, or if no corner is available, the mouth should be facing a wall 6½ inches away. This placement will best utilize the wall or corner as an extension of the horn, further improving low frequency coupling.

SLEWING INDUCED DISTORTION IN AUDIO AMPLIFIERS

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