Slewing Induced Distortion: Part 4

Phase IV: Listening Tests for SID

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WE COME NOW to the acid test or, in popular parlance, where the rubber meets the road: listening. If all the electrical tests we've made in this series are significant, some definite patterns should evolve in listening tests of the op amps studied for SID. If we find no pattern in listening, all the above work will probably be rejected as meaningless by the ultra purist, "subjective only" advocates.

Fortunately, as we will see, the answers from the listening tests not only correlate with the measurements, *they correlate well*. But before we get into that, let's look first at how the tests were done.

Test Setup

For this testing phase, I did a fair amount of preliminary work to prepare my system for an easy and repeatable "objective" subjective test (if there is such a thing). In the general interest of simplicity and practicality, these listening tests were all done in mono, evaluating one IC at a time.

Fig.IV-1 is the block diagram of the system. Both channels of my reference stereo power amp (Ampzilla) are fed in parallel from a single channel of a Dyna PAT-5 preamp. This guarantees a balanced mono signal from each speaker, which are Magnepan MG-II's. All sources were operated either in a mono mode (tuner) or strapped in parallel for mono, in the case of tape and phono. Thus only one signal channel of the PAT-5 is used for the tests.

The PAT-5 channel I actually used was one specially modified for these tests to eliminate all traces of SID. Details of the modification are to be described in a future article (and embodied in a kit), but in general consisted of wholesale replacement of circuitry, with the use of the highest performance ICs (in terms of SID) selected from the THD and IM tests. This modification resulted in a full output level (7V) THD of less than 0.002%, across the full audio band.

Listening tests on this modified version of the PAT-5 opened up new dimensions in musicality, and even an increased depth and spaciousness, even though operation is strictly mono. I doubt if valid listening tests could be performed for SID without this type of modification, as quite a bit of SID was originally present in both the phono and high level circuitry. Unfortunately, this factor will complicate the exact duplication of my tests by other readers, at least for the present time.

The audible separation and identification of SID in an experimental audio circuit will be very difficult or impossible, unless the test system is already low in SID. In other words, you can't easily A-B a before/after switch for SID if you have a large measure of it in your present preamp or power amp. However, if you have a first-rate system, with a smooth clean upper range, it should be possible.

An audible verification of this is a completely free and unrestrained high end which will reproduce transients with an exact and detailed naturalness. High level cymbals and/or traps should sound effortless, with crisp reproduction. Upper register violins should be smooth, sweet, and warm, with no traces of edge or hardness whatsoever.

You'll know what I mean by this if violins elicit a "goose bump" effect when reproduced on your system. It is this sort of quality which is needed for reference in comparing various ICs. This of course assumes your recordings and/or other sources to be top-notch.

Given the above, you can compare various ICs for SID effects in an appropriate listening test circuit. In my case I connected the listening test circuit between the TAPE OUT jack of the PAT-5 and the TAPE 2 "in" jack. The PAT-5 is operated with all input sources except TAPE 2, with the monitor switch on TAPE 2.

In this manner the INPUT/MONITOR push button switch can be used to select between the input reference signal (A) and the signal through the test circuit (B) which appears as the Tape 2 monitor signal. Other preamps (if used) have similar facilities. Thus the tape monitor switch is used as an A-B select between the original source and the version passed through the test circuit.

The listening test circuit is shown in Fig.IV-2, and is especially designed for maximum sensitivity to SID. The first amplifier A1 is a 6 to 20dB gain stage (gain set by R3), which scales up the signal from its normal line level to one which will drive the U.U.T. to near full voltage output in normal operation. The device used for A1 is a 318, selected for its virtually zero SID and wide bandwidth.

I operated the U.U.T. in the unity gain inverting mode, for the reasons previously cited. Its output, which is greater than the input by the gain factor of the A1 stage, is then scaled down to the original level by R9/R10-R36. The output signal across R10 is equal to the original input level, within the allowance of resistor tolerances. For A-B testing, levels should be matched, either by the use of matched pairs for the like resistors noted, or by trimming one resistor (such as R9).

In my circuit I matched levels within ±0.1dB by trimming R9. C3 removes any DC offset generated by the test circuit. Exclusive of the U.U.T., distortion through this circuit is less than 0.004% and consists mostly of noise. Thus there is fair assurance that what you will be listening to is actually distortion in the U.U.T., not in the remainder of the circuit. This was also audibly verified by using a 318 in the U.U.T. position.

In use the circuit is fairly simple, but an oscilloscope is a handy operational aid, and can also visually indicate SID. With the circuit connected and operating, check first for correct 1:1 signal reproduction across R10-R3B. Input levels and/or the use of gain control R3 should be adjusted for an amplitude @ A1-6 of close to 20Vpp, but below clipping. Avoid clipping, which confuses the distortion issue. Now, depending on the U.U.T. in service, you may or may not be able to hear distortion.

You can gain some familiarity with the sound of SID by purposely using a very low slew rate device, and playing a high level, high frequency selection (with suitable quality). One method of ensuring this is to use first a 301A overcompensated with 330pF. By using the scope to monitor the summing point of the U.U.T. on a sensitivity of about 50mV/division, you should see increasingly large voltage levels coinciding with the HF passages which trigger slew limiting.

This can be tested audibly, by switching the output of the test circuit to DISTORTION PRODUCTS, which allows you to listen to these error components. When SID is generated you will hear a hard, gritty, and grating distorted sound that

FIG. IV-1

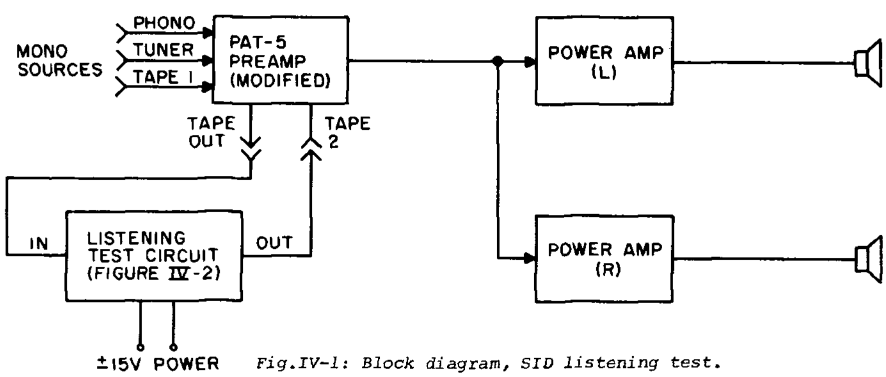
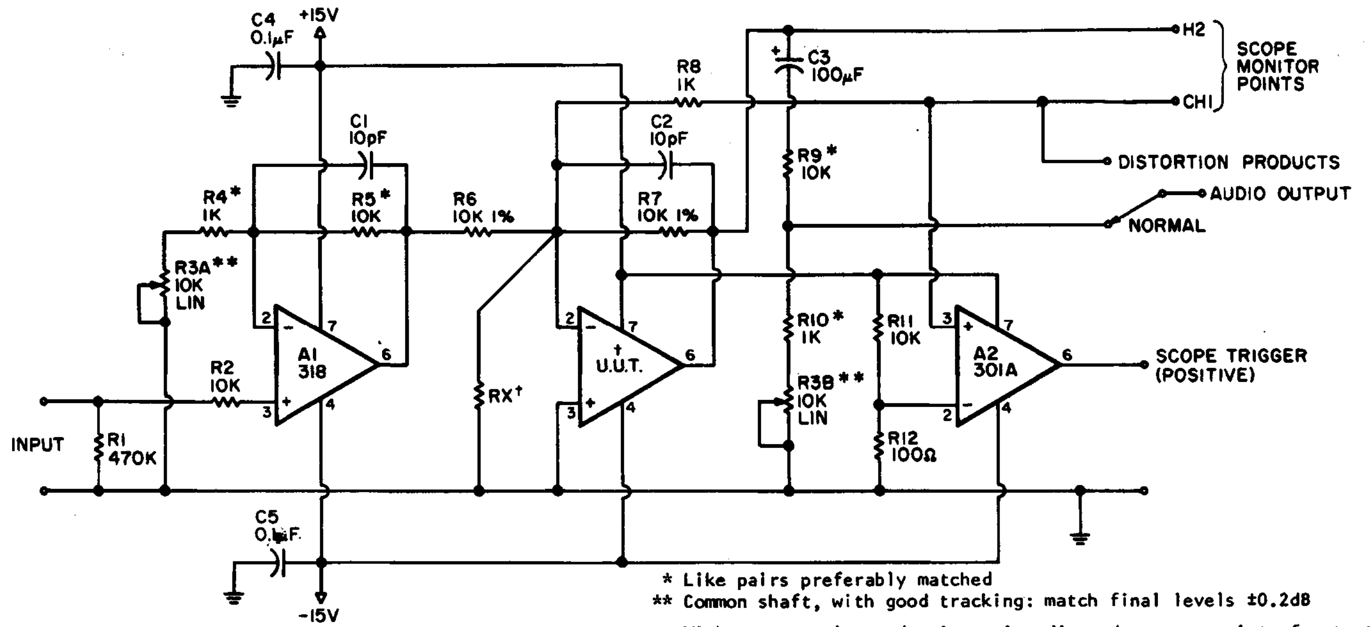


Fig.IV-1: Block diagram, SID listening test.

FIG. IV-2



* Like pairs preferably matched
** Common shaft, with good tracking: match final levels ±0.2dB
† With compensation and noise gain adjusted as appropriate for testing

Fig.IV-2: Circuit for SID listening tests.

is unmistakable. You now have a reference for what to listen for within the overall sound picture. Listening to the summing point in effect scales up the distortion, and isolates it for scrutiny. It will never be this bad in a normal frame of things, but this test will give you an audible perspective of its nature.

At this slew rate setting, SID is relatively easy to generate and detect. By switching the monitor point back to NORMAL, distortion should be readily apparent with most program material, particularly high level passages with high frequencies. To verify that you are hearing true SID, rotate R3 to a lower gain setting, which will result in less output voltage from the U.U.T. The harsh distortion will disappear, although the overall level remains the same (assuming the R3 a-b sections track well). This demonstrates how SID is level dependent, being worst at the highest voltages.

With your ears thus "calibrated," you are ready for analytical listening tests. This is best done in steps, beginning with a low slew rate device such as the 741 or a 709 with x1 compensation. Careful listening to HF components of program material should reveal edginess on crescendos and peaks. The scope should also serve as a verification of this, by displaying bursts of fuzz at the summing point which coincide with the SID.

By using the scope in a dual trace mode, with the second channel monitoring the U.U.T. output, watch for the maximum levels which can show SID. With repeated listening, you should be able to A-B the source and the U.U.T. to verify whether the SID is actually coming from the U.U.T. or the source.

I find many terms describe the sound of SID. Interestingly, the subjective reviewers have used a great many of them for years to describe audible defects. Yet no one has directly linked these colorations to measurable circuit parameters. Part of this difficulty is in finding the right relationship; an even greater one is to quantify the audible degradation as well as the electrical measurements, once a relation has been established. At present, I'll admit I am a lot worse at quantifying the subjective experience of SID than the electrically measured form.

Yet audible gradations do seem to evolve from the listening experience. The audible degradations detected in these listening tests did not appear at all to be characteristic of the particular device type; rather they were related to the device's slew rate capability.

Several factors are important to the overall success of the listening tests. One of these is the correct choice of program material. It should emphasize high frequencies, the range of 2-3kHz and up. The levels should preferably be as high as possible and feature extended solo passages: violin concertos fill this bill nicely, if the recording is first-rate. One of my most useful test records is the Bruch concerto (see listing), which features a final movement which can demonstrate (expose) SID better than a lot of others.

Bluegrass material is useful but requires more care. Selections which feature well recorded solo fiddles can be almost as useful as a violin concerto. Plucked string instruments such as guitar, banjo, and mandolin are also useful indicators of transient quality, but they must be *acoustical* pickups only. A good test record for this type of material is the Mike Auldridge album listed; it is well recorded and outstanding in its dynamics.

Rock and typical pop music is the least useful, and in general should not be considered for evaluating SID. So much distortion and compression is built into these recordings that you'll never be able to separate things properly.

The Sheffield direct cut discs are outstanding in unrestricted dynamics, the best I know of to demonstrate what transients should sound like. In commercial recordings, this is about as close as you can get to studio sound. The Thelma Houston/Pressure Cooker release listed is useful for its spectacular drum transients and muted brass, which will pinpoint SID.

In the listening tests our procedure was to plug a device into the U.U.T. socket and listen carefully to the HF program content. Generally, I used only the HF range for comparison, giving little attention to lows. A first level, or most sensitive comparison could be made on one of the more controlled and "steady state" musical selections such as the Bruch (best) or Mike Auldridge cuts (next best).

I carefully monitored the solo violin in the extended high level passages for differences between A and B states. This would show up first as a barely discernible dulling of the string tone, or slight loss of warmth and sweetness, a vanishing of the "air."

These differences were so subtle they would probably not be detectable at all without an A-B comparison. I also no-

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SELECTED TEST RECORDINGS

*1 *Bruch and Sibelius violin concertos--Zino Francescatti, Schippers/Bernstein, New York Philharmonic*, COL MS 6731
2 *Orff: Carmina Burana--M.T. Thomas, Cleveland Orchestra*, COL MX 33172
3 *Mahler: 8th Symphony--Solti, Chicago Symphony*, London OSA-1295
4 *Lincoln Mayorga & Distinguished Colleagues, Vol.III*, Sheffield LAB-2
*5 *Thelma Houston/Pressure Cooker, "I've Got the Music for Me,"* Sheffield LAB-2
*6 *Mike Auldridge, "Blues and Bluegrass"* Takoma D-041
7 *Linda Ronstadt, "Hasten Down the Wind,"* Asylum TE-1072
8 *Linda Ronstadt, "Prisoner in Disguise,"* Asylum TE-1045

** Particularly useful*

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ticed that this first slight amount of degradation (level B) would not be detected at lower U.U.T. output voltage levels; only at near maximum output could it be observed.

The next detectable level of audible deterioration (level C) was a more apparent dulling of strings, and loss of warmth. The sound of strings is now tending toward dryness. This is a difference of degree, beyond the just detectable. Overall HF response is still generally satisfactory, and the audible coloration is slight. This level may possibly be audible in a straight listening test, if the listener is familiar with the musical selection.

Level D is one of marginal quality, with noticeable losses of string tone, warmth, and dimensioning. This is the first really serious level of audible defects, and can be noted by instruments which seem to "collapse" in their sonic image in level B, as opposed to a three-dimensional image which projects in level A.

I noted this in various instruments such as violins, banjo, mandolin, and traps. Instruments tend to sound "covered up" or recessed into the overall sound. Massed strings begin to take on an edge with high levels, and instruments generally blend as a homogenous source rather than concerted individual ones. Live voice reproduction sounds constricted or restrained, with a lack of natural quality.

The final categorized defect (level E) is recognizable on any sort of program material, and need not be sought after--it is apparent. Coloration and distortion of highs is obvious, and may also be accompanied by grit, fuzz, etc. In this severe case, the defect seems to affect voices to a greater degree, being noticeable on live speakers, particularly on sibilants. A speaking voice under these conditions sounds quite constricted or restrained, as if in a form of HF crossover distortion. The loss of naturalness is immediately apparent when A/B compared, but also obvious even without. HF musical transients may be almost completely subdued to the point of loss, with their residual manifestations being smeared. It is this quality level which was purposely induced initially.

Naturalness is the one general term which comes to my mind as most apt for judging the perspective of audible SID. If an amplifier sounds completely free and natural for any form of source material, regardless of difficulty, it is probably free of audible SID. The various levels of deterioration just described (B-D) all remove degrees of natural quality, to level E where the lack of naturalness is grossly apparent. For levels B and C, the degradations seem to be subtractions or losses from the original sonic image. With level D, these losses are increased, but with added distortions such as "edginess," "grit," or "constriction."

I suspect (but don't know of a simple way of proving it as yet) that levels B and C can be equated to the approach of slew limiting, as was evident in THD tests as the initial rapid climb in distortion. Level D seems to be this degree, plus perhaps occasional spillovers into complete slew limiting where the rasping of edginess and grit takes place. The transients sound as though they are covered up.

Level E is slew limiting for a great percentage of the time. This may be generally confirmed by the observation of ample bursts of summing point voltage, which indicates the open loop condition of slew limiting. I have made no effort at all to classify defect levels worse than level E, since this is already an intolerable degree of distortion.

To confirm or deny my thesis about the differentiation of quality levels B-C and D-E, and whether or not actual slew rate limiting was being triggered, I added the A2 comparator stage to the listening circuit (Fig.IV-2). I did so after I had already categorized the majority of the ICs for sound quality. This circuit simply triggers the scope very positively when a slewing condition is present in the U.U.T.

In operation, the (-) input of A2 is biased at +150mV by R11-R12. When a U.U.T. summing point voltage deviation occurs greater than this level, A2's output goes positive. This level change can be used to trigger the scope externally, so as to coincide the start of the sweep with a slewing interval when present. This allows positive identification and verification of slewing.

The A2 comparator's only weakness is that it responds just as well to *clipping* of the U.U.T. However, clipping is easily recognized, as the U.U.T. output (CH2) will be at a negative saturation level as the sweep starts. Conversely, a true slewing interval will start at some more positive level near zero, and ramp negative. You can (with careful observation) actually *measure* a device's slew rate under program conditions using this technique.

Using the comparator to detect slewing conditions and carefully setting the drive level to the U.U.T., I made a number of listening/electrical tests to

Table IV-1
Listening test results (referred to full output of ±10V)

SID category	I Deterioration			II Gross distortion	
Quality level	A	B	C	D	E
Audible character	No differences detected for any program material	Just discernible softening, loss of sweetness	Further softening, somewhat dry, generally satisfactory with slight loss of dimension	Colorations apparent, loss of dimension, "covered" sound, dulled transients, constricted, edge begins	Coloration and distortion obvious, more constricted covered sound, transients smeared, grit, edginess, fuzz
Associated slew rates	>4V/μS	2-4V/μS	1-2V/μS	0.5-1V/μS	<0.5V/μS
Samples tested	318, 518 TDA1034	1456 4136 (2V/μS)	4136 (1V/μS)	741	
	2625 2525				
	8007 NE536 AD540 3140		4741 356*		
	NE541 (x100 comp) NE540 (x100 comp) TL084 OP-01	NE541* (x10 comp) NE540* (x10 comp)			
	530A	530			
			535 538* 1741S		
	531 (x10 comp)		531 (x1 comp)		
	2720 (5V/μS)		2720 (1.6V/μS)	2720 (0.5V/μS)	2720 (0.16V/μS)
	301A (x10 comp) 301A (x100 comp) 301A (FF)			301A (x1 comp)	
		709 (x10 comp)			709 (x1 comp)

* *Audible ranking possibly due to factors other than slew rate*

verify the presence of slew limiting, on what material, and how often. The results are indeed interesting and I am sure will stimulate lively reader discussions.

Generally, once the device possessed a slew rate setting of 1V/µS (or more), there was never any gross limiting observed. Clipping of the U.U.T. occasionally triggered the scope, but since this could be easily identified, it was dismissed.

This general picture did not change appreciably with a device slew rate near 0.5V/µS, but some slew limiting did take place on the most difficult material on hand, the Bruch concerto. However, to reach a U.U.T. output level which would occasionally slew the U.U.T. on true HF signals, the overall level was such that frequent clipping was taking place on peaks.

This indicates one of two things. Either we need a more demanding test work, or it is unrealistic to expect 8-10kHz and up levels to be at or near peak program levels. Since the latter is more probable, this indicates that actual slew limiting would be relatively infrequent for 0.5V/µS to 1V/µS devices (although certainly still possible).

Below 0.5V/µS and at about 0.05 to 0.1V/µS frequent slew limiting takes place. On a rich HF range recording such as the Bruch, this is overwhelmingly evident, and the output waveform can readily be observed to contain the triangular slew limited components. Even on more ordinary program material slew limiting is not at all infrequent. We should expect this, since the fp of a 0.1V/µS device is only 1.6kHz which is near the center of the audio spectrum energy distribution, and thus wide open for slew overloading.

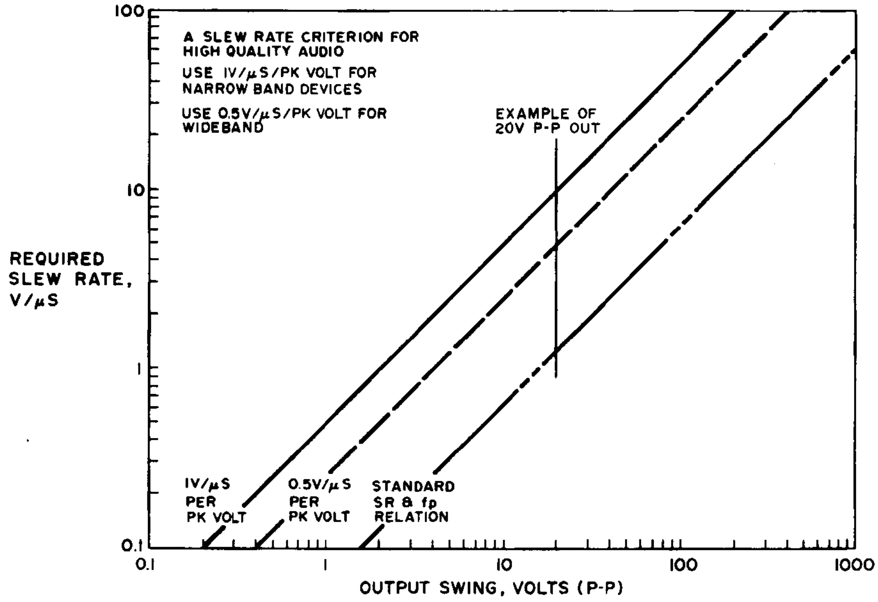
To bring these tests into a comprehensive focus, two broad but distinct categories of audible distortion seem to be associated with SID. At low levels of SID, the audible effects are a general loss of naturalness and dulling of detail. I will call this category I SID, a general deterioration, and it encompasses quality levels B and C and overlaps D.

A more serious form of audible distortion due to SID is associated with complete slew limiting, the condition when an amplifier is called upon to deliver a rate of rise in excess of its slewing ability. Gross distortion is evident in this case, apparent by fuzz, grit, and harsh reproduction on signal peaks. This I will call category II SID, and it encompasses quality levels D and E.

Level D brackets SID categories I and II, and depending upon the specific slew rate and program material may produce results of either. A low slew rate device such as a 0.5V/µS unit may occasionally slew limit and produce category II SID on certain program material, but not consistently. It will generally be producing category I SID (to its worst degree) accompanied by the associated dulled HF range.

Table IV-1 summarizes my listening test findings very nicely, and assigns an associated slew rate range for each quality level. I made these quality level and slew rate range judgments during the listening tests, using the devices shown. I will not say my results are immutable, but I do feel they illustrate a very definite pattern. This pattern is simply that higher slew rate devices

FIG. IV-3



generally sound better, and if sufficient slew rate is not present, a device can sound disastrously bad (category II).

A less concrete result is the exact differentiation of quality levels B-D and I submit these results as one subjective listener to the situation. Others may hear things a little differently, but I do believe my point has been made that slew rate can be linked directly to audible quality, and in different gradations.

This thesis is experimentally supported by several devices whose quality level could be changed by an adjustment in slew rate: i.e., the 301A, 709, 4136 (different samples) and the 2720 which could be spotted at various levels.

Several devices on the IV-1 chart do not fit the general pattern of audible quality proportional to slew rate. These are starred, indicating that their ranking is probably due to other factors. For example, the NE540 and 541, with x10 compensation, slewed at 4V/µS or more. However, their sound for this condition was not of A level quality (indistinguishable from source), it was more of a level B type. This may be due to the rise in THD for these devices which approaches 0.1% @ 20kHz. When compensated for a x100 condition, they both became A level quality, and their distortion is lower for this condition.

The slew enhanced devices present some unique listening experiences. One of my initial curiosities concerning these units was whether or not the effects of the class AB input stage mechanism can be heard; the results make me believe they can. The 535, 1741S and x1 comp 531 all sound about the same, of a general C quality although perhaps bordering on D level. One can hear the beginnings of an edge and a definite loss of imaging.

The 538 units presented problems in evaluation, as two of the three samples exhibited a curious parasitic instability. The one unit which behaved more or less normally seemed to be better than the 535, 531 (x1) and 1741S, but it still was not altogether what I thought it should be. It seemed to be of B quality, but because the results were inconsistent in two of three cases, it is listed in IV-1 as a C.

The 530 was the best performer of the slew enhanced units, and sounded only slightly less than A quality. This unit showed lower THD than any of the others, which indicates that the ear is sensitive to really quite low levels of distortion in the 10-20kHz range.

The 356, on the basis of its slew rate and THD, one would expect to do fairly well. However, it was noticeably less than transparent, losing dimension and yielding a dry sort of sound; hence its C rating. Some of this may be the device's basic asymmetry, as noted above. This underscores the necessity of more research into this aspect of IC performance.

All these anomalies indicate that the ear can indeed perceive *very* small levels of distortion, down to well below 0.1%, perhaps as low as 0.01 or 0.02%. It may be that what we are hearing is not exclusively distortion, but other interrelated results which cannot as yet be completely pinpointed.

A New Slew Rate Criterion

This information, in conjunction with the results of the electrical tests, can be used as the foundation of a *new slew rate criterion*. It should be one based on the requirements necessary for high quality audio in a 20kHz bandwidth, and may be used as a predictive tool in circuit design or evaluation. I present it in graph form in Fig. IV-3.

To use this nomograph, only the required amplifier p-p output voltage need be known; the graph will then tell you the required slew rate in V/µS. As an example, an op amp intended to deliver its rated 20V swing is plotted (vertical line). This line intersects the standard fp relation (shown dotted, for reference only) at a level of 1.25V/µS. As has been shown, this is inadequate for high quality use; it is included here only for perspective.

The remaining two lines correspond to the new slew rate criterion, and represent slew rates of 0.5 and 1V/µS per peak output volt. The 0.5V/μS curve may be viewed as a minimum objective, the 1V/μS as a conservative one. The example of the op amp would thus require a 5V/μS (minimum) or 10V/μS device slew rate, as indicated by the intersection of the vertical line at these points.

Since two range extremes of the criterion are presented, the question arises: which should a designer use? The 1V/μS per peak output volt is the most conservative, and provides allowance for other error factors. It should preferably be used for instances where an excess of bandwidth is *not* present. The 0.5/μS curve can be used for wide band devices, which will have higher feedback factors and are therefore " more forgiving."

This criterion is justified on the basis of both the electrical and the listening tests. In fact, the latter would indicate a 0.5V/μS rating per peak output volt to be adequate.

As a final perspective on the listening tests, the reader should appreciate that since they were all done at maximum amplifier output level, they are as pessimistic as can be. In practice, this means any given device will perform proportionally better as the output levels are lowered. This should be an exact linear relationship, that is, a 2/1 performance increase, if you halve the output swing. In terms of the quality level brackets, this means a device will move to the left, or improve, in quality level. Since the brackets are intended to be roughly binary weightings (at least from B-D), a device could possibly move more than a single quality level.

Exactly how much improvement will result should be taken with a grain of salt, however, since what we are dealing with here are peak program levels, not nicely defined sine wave amplitudes. Regardless of level, however, the relative rankings of slew rate (and thus the associated devices) will still hold, which is simply to say "faster is better" except for the special cases as noted.

You can also take the new slew rate criterion, of course, and extend it *upward* in level to include power amplifiers. The Table IV-1 data, which are based on ±10V output levels, would in this case be optimistic. Of course the ICs tested here do not generally operate at higher voltage levels (with the exception of the 541, a power driver), but the slew rates associated with different quality levels should be almost directly proportional. Hence, a ±50V output swing should require five times the slew rates shown for the respective quality levels to attain comparable qualities.

At such an operating level, the crossover from level D to E would be 2.5V/μS, and A level quality would occur at around 20VμS. I can't say definitely that all this will prove out, as I haven't run as many tests on power amps as on ICs. I do know, however, that a faster, symmetrically slewing power amp sounds better, just as ICs do (see my Dyna 400 review in Issue #2, 1977,p. 48).

A System Slew Rate Perspective

With all this information we can now assemble a slew rate/signal flow diagram for the entire audio system. The Fig.IV-4 diagram applies the new criterion to an audio system on a logical, stage-by-stage basis, and shows the required slew rate of each stage. Although this drawing is somewhat hypothetical, and the exact numbers may vary in an individual case, the general principles of the relationship will hold for any set of values.

Defining the slew rate requirements for the components of a system begins with the power amplifier, which is first specified in terms of slew rate from its rated output voltage. Note the correct key terms here is *voltage*, not power. It is peak voltage swing which determines the required slew rate. To use a popular example, assume a 200 Watt into 8 Ohm amplifier. This results in a 112 Volt p-p output voltage for rated power. From Fig.IV-3, this requires a slew rate of 50V/μS (actually 56V/μS, but rounded off for purposes of illustration).

If the power amp is to be the weakest or limiting link of the system, all preceding stage slew rates should be in excess of the figure predicted by this step, when related to their individual levels.

To determine the slew rate required of the previous stage, the power amplifier slew rate must be *referred to its own input*, by dividing by the stage voltage gain to give equal basis comparison. In Fig.IV-4, assuming a 50V/μS power amp slew rate and a voltage gain of 25 (typical), this power amplifier's "input referred" slew rate would be 2V/μS. This means that even in the case of an "infinitely" fast power amp, a 2V/μS input slew rate from the preceding source would result in no more than a 50V/μS final output level slew rate.

Since an "infinitely fast" power amp does not exist, the driving source should have a slew rate in excess of the power amp's input referred slew rate, to avoid deteriorating the latter's final slew rate to less than 50V/μS (or whatever figure is appropriate).

My rule of thumb used here is a x5 ratio, but even greater multipliers will yield more conservative results. Therefore, in this case the preamp's final stage should possess a 10V/μS or more slew rate, so as not to deteriorate the power amp's 2V/μS input referred slew rate.

You may extend this rationale to all other fixed voltage gain stages, such as the output stage of the preamp (line amp block), which will typically run a gain of 10 (20dB). For an output slew rate of 10V/μS this stage will then have an input referred slew rate of 1V/μS. Again, using the multiplier of 5, all input sources should have a slew rate of 5V/μS so as not to deteriorate this slew rate and thus the final system slew rate.

Working toward the system input we find the master volume control which can pass preamp (or other source) voltage levels near maximum output, regardless of the final playback level. We may therefore justifiably regard the preamp in terms of its rated output voltage for slew rate, rather than referring it to the input referred slew rate of the line output stage. We can do this by using the new criteria of Fig.IV-3. For a preamp with ±10V output, the minimum slew rate would be 5V/μS, as noted in the example case here.

Further points should be dealt with on a system basis, such as out-of-band rolloffs to prevent possible supersonic slew limiting or IM generation. Typically such measures appear at the power amp input in the form of a passive RC network, although in some cases they may be appropriate at other points. Also, tone control and/or equalization HF boosts should be carefully considered on a system basis, as they can require slew rates in excess of that implied from the above, if the boosts affect frequencies above audibility.

Conclusion

After three series of electrical tests and one listening test on some 100 ICs, many things have been learned about audio performance feedback amplifiers as related to slew rate. However, as in many such instances of research, the important thing is to separate the wheat from the chaff. In plain words, the wheat follows.

The slew rate, or large signal voltage rate of change ability of an audio amplifier to be used in feedback or "op amp" type circuits is of *major* importance. This fact is demonstrated by both electrically measured and audible results on a large number of amplifiers of widely differing designs. In all cases where slewing rate limitations cause deviation from an ideal balanced input stage state, easily measurable distortions result.

Contrary to many previously published comments, this form of distortion can be identified and quantified (most effectively) by simple THD tests. Except for some specialized cases, the results of THD performance tests can be directly related to the slew rate of a given amplifier. This may also be stated for other forms of test, such as two-tone HF IM, and the sine/square technique, but those tests generally seem to be less sensitive to detection of SID.

Listening tests of these same amplifiers yield results which correlate surprisingly well, to the degree that audible sound quality may also be ranked by slew rate in a large percentage of cases. Listening test results reveal two major categories of distortion due to slew

FIG. IV-4

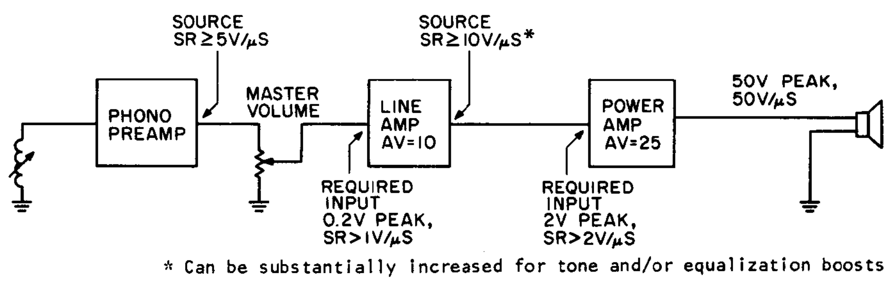


Fig.IV-4: Audio system slew rate/signal level flow rate: category I, a general deterioriation, and category II, a gross distortion. In electrical terms these categories are associated with (I) the approach of the device's slew rate, and (II) the exceeding of this slew rate.

Interestingly, the subjective impressions of the sounds of slew limiting have been used in print in many instances for some years, yet no one has defined this link in clear and unambiguous terms, let alone quantitative ones.

Slew limiting effects in both electrical and audible performance degradations can be avoided by applying a new slew rate criterion, to wit: "The circuit, including all possible loading conditions, should possess a slew rate of 0.5V/µS (minimum) to 1V/µS (conservative) *per peak output volt.*" If this criterion is satisfied, electrically measurable distortions due to slew limiting will generally border on the unmeasurable, or THD below 0.01% of full scale (below 20kHz), as well as correspondingly low IM and TIM.

Audibly, the device performance will be such that slew limiting effects will be undetectable in a full level A-B test on the most difficult program material. The only qualifications for applying this slew rate criterion are that the device slew rate be symmetrical and that the input stage be of a class A design.

This criterion can be misapplied, if care is not taken to limit device input bandwidth to a normal 20kHz. Such a misapplication could result, for instance, when out-of-band components cause slew limiting. A more general rule for guaranteed satisfaction is to maintain the signal/device slew rate ratio at 0.25 (or less), a re-phrasing of the new slew rate criterion. This may be accomplished by appropriate passive band limiting to define signal upper slew rate limits.

A major implication of this study is the demonstrated necessity for both product specifications which include slew rate and standard test methods which recognize it. Although the solid state audio age has been with us for more than a decade, there is no present general recognition or appreciation of this distortion phenomenon. This is regrettable, as it is much more significant than most of the performance parameters typically specified and tested in audio gear. We hope this study will have a beneficial effect on this situation.

The above is a nutshell summary of the outcome of the SID study. We are aware that many of the general conclusions and specific points of this study are in conflict with some of the previously published works on TIM. No doubt this will lead to controversy, but then perhaps a clearing of the air is in order.

I have endeavored in every phase of the testing to be as objective as possible, while presenting an ample amount of reliable data which establish certain points. I feel confident that the electrical tests will generally be accepted as valid, but the listening tests (since they were done through my ears) may justify further test cases. I would welcome these in the interest of overall validity, and encourage readers to duplicate the tests on their own.

I welcome reader comments. However, please write directly to the appropriate manufacturer to find sources for a specific IC. Send along your comments and/or criticism, especially as to future research into the general subject of amplifier distortions. *[If you expect a reply, do please leave space on letters for comment, and always enclose a stamped, addressed envelope.--Ed.]* *[For a full listing of manufacturers and their addresses, see pp. 28 and 57 of TAA, Issue 3, 1977.]*

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AUDIO RESEARCH RE-WORKS DYNA ST-70

Continued from page 11

filament supplies were added, the builder could doubtless use EL34s in the finished unit.

The enterprising will note that the constant current source of the D76A is a single 6FQ7 tube and two resistors per channel and that DC balance would not be too difficult to add to the bias system. A delayed regulated high voltage power supply of higher current rating with separate filament supplies (possibly DC) would be the natural perfectionist's version of this unit.

THE SOUND

While we have not compared the ST-70-C3 to the D76A, we have had the chance to compare it to several solid state units. We preferred it to all save the Williamson Twin 20 Mark II. The sound of the two is remarkably similar--both very pleasing, concise, and without any grit or edginess.

We lived with the unit for many months and it has that hallmark of all good power amps, it really seems to have no "character" of its own. FM from the music stations runs along satisfyingly by the hour and suddenly we are dazzled by a live broadcast of the Boston Symphony when the sound takes on an attention arresting, nourishing and satisfying character. Side by side channel comparisons with amps of similar power make it evident that the ST-70-C3 is a very elegant device for reproducing music. Modest by today's power standards no doubt, but a quite welcome and valued addition to our system.

We trust that those who elect custom options in building versions of this unit will share results with other readers through the letters column. Those who know of good sources for more difficult to find parts are encouraged to share that data.

All in all, the ST-70-C3 project ought to provide a lot of new experience for tube buffs and perhaps a few will become converts to the cult. We found it an exceptionally satisfying experience.

SPECIAL NOTICE

OLD COLONY SOUND LAB will act as sole supplier and warranty agent for the Audio Research ST-70-C3. Audio Research will not answer any mail or phone queries about the product, its construction or maintenance.