New Methods of Amplifier Evaluation: 3DA & I/O Distortion Analysis

It is sometimes surprising to discover that two amplifiers sound much more different than their specs would lead one to believe. And after working hard to improve one aspect of an amplifier's performance, it is disappointing to hear no significant improvement in sound quality. Obviously better analysis techniques are needed; we must find a more accurate way of revealing the sources of amplifier performance problems so that we can come up with amps that come closer to the ideal.

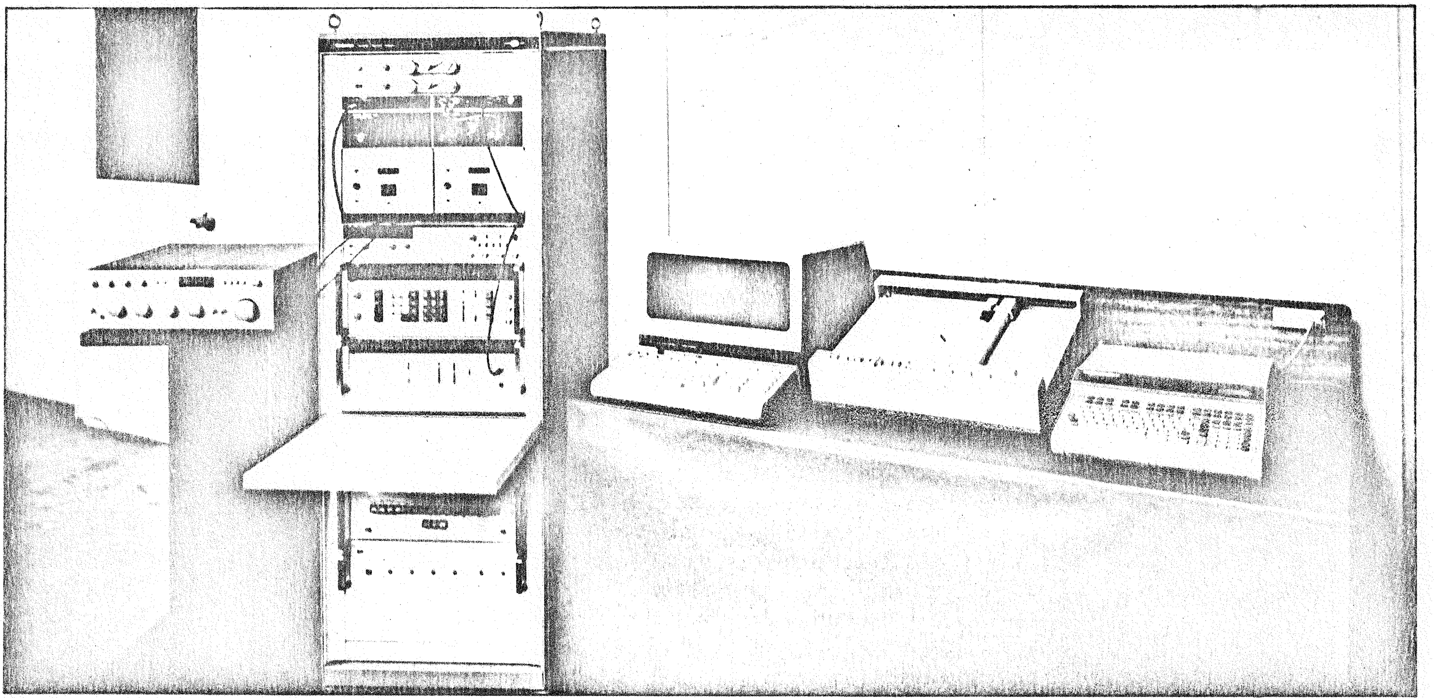
Sound quality is not determined by the flatness of frequency response alone. Nor will merely minimizing distortion result in optimum performance. Rather, it is the combination of all performance factors that finally determine sound quality. If we are going to make any significant progress in amplifier design, it is important to seek out each of these factors and grasp their overall relationships in a coherent way. Amplitude vs. frequency, output vs. distortion, noise characteristics, power bandwidth, phase characteristics, and the like are all conventional ways of looking at data concerned with amp performance. The problem is that while any one of these is certainly useful, it cannot amount to more than a section of the total picture.

It is very hard to grasp the sum total of an amp's performance characteristics even if one investigates a great amount of such data. The only alternative for a coherent picture is human hearing, which is why listening tests are an extremely important part of building amplifiers. Unfortunately, it is very hard to convert the results of such tests into hard scientific data. The human brain does not have the necessary analytical or memory capability. Such factors as the passage of time, the number of people in the sample, and the listening room all affect the reliability of the test results. Until now there has been little luck among those who have tried to come up with test methods that would consistently agree with and support the results of human listening tests. At Technics, we are happy to announce some significant developments in this area. In the following explanation we will introduce two new methods of analyzing amplifier performance.

3DA (Three Dimensional Analysis) System

A coherent and broad way of expressing overall amplifier characteristics is necessary if we want an objective, scientific grasp of total performance. This method of expression should have a significant relationship with subjective sound quality. 3DA is a method of analyzing sound quality and performance characteristics based on a three dimensional representation of the relationships between the three most important measurable factors significantly related to perceived sound quality in amplifiers. As rhythm, melody, and harmony are the three elements of music, we can say that the three most important factors affecting amplifier sound quality are distortion, frequency response, and dynamic characteristics (in broad terms including phase, noise, and output in the definition of each). The aim is for flat frequency response, low distortion, and wide dynamic range (meaning high S/N ratio with high power capability).

A major problem in amplifier design has been that decisions have been based on a 2-dimensional analysis of problems. The amplifier designer must work with graphs having a vertical and horizontal axis on which are plotted the results of tests concerned with the relationship between only two factors at a time. A coherent, integrated view of the situation is not easy.



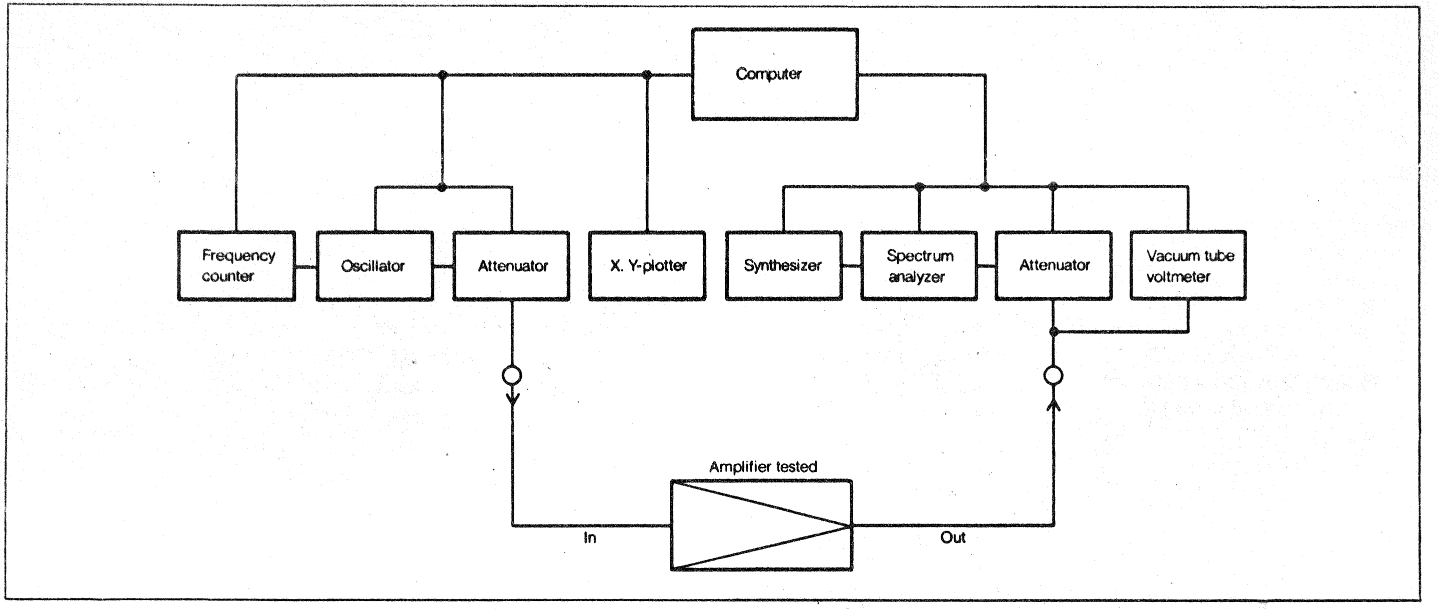
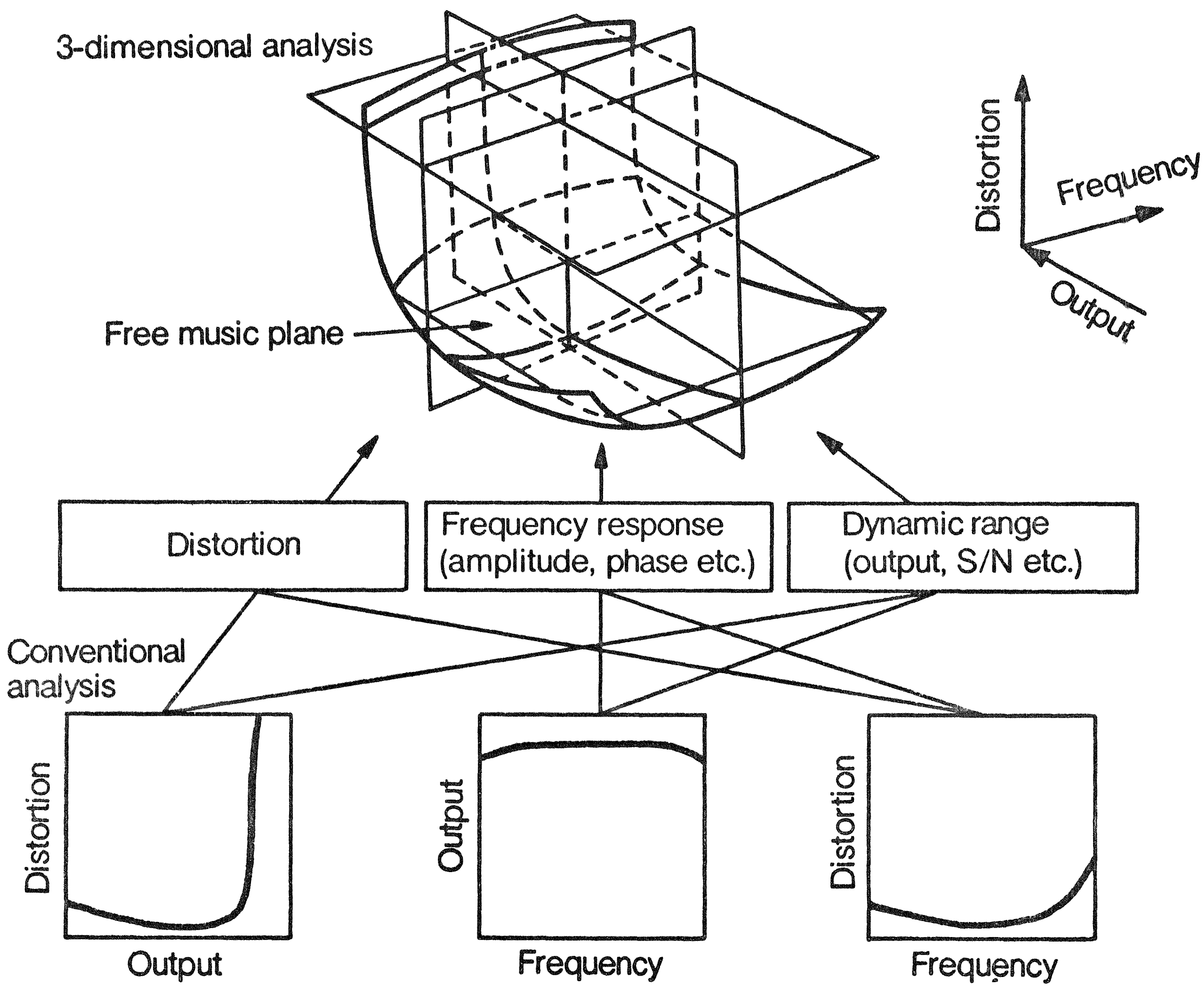


Photo 1. 3DA automatic measurement system with block diagram.

At Technics, we decided that things would be much easier if we could convert such flat graphs into a 3-dimensional representation covering the complete relationships between the major factors contributing to amplifier performance. Conventional measurement of distortion is an analog technique in which the fundamental is filtered out and the remaining components are compared with the original waveform. Since it is hard to analyse the distortion components, and noise and hum are included in the resulsts, figures obtained by this method have a rather shallow relationship with sound quality. We have greatly improved on this approach by employing digital techniques with a spectrum analyzer and computer to analyse the results obtained from 4000 test points. With this system we are able to separate noise and hum from the output waveform and analyse each of the upper harmonic distortion components (2nd, 3rd, 4th, ·······) with a very high degree of precision. Photo 1 shows this 3DA automatic measurement system.

As you can see in figure 1, a 3-dimensional graph is formed with output and frequency on the X and Y axes while distortion is on the Z axis. This provides an extremely precise view of the overall distortion characteristics of an amplifier.

Fig. 1: 3-dimensional analysis.



Differences between 2-dimensional and 3-dimensional representations

To better understand 3DA, it may help to examine some of the differences between 2-dimensional and 3-dimensional representations. In figure 2 you will see an example of third angle projection, a method widely used for technical drawings of solid bodies. Even for people with much experience with this method, it is still quite difficult to form an overall picture in one's mind from these front, top, and side views of the object. The same object is much easier to understand when portrayed in the 3-dimensional manner shown in figure 3. In the same way, 3DA lets you get an overall grasp of amplifier performance characteristics at a single glance. As shown in figure 4, you can use 3DA to obtain conventional two dimensional data also. For example if you take (B) as your output reference, you could check the relationship between distortion and frequency in a 2-dimensional manner. You can do likewise for frequency or distortion by using cross-section (A) or (C) as the reference.
The fact that you can obtain both an overall view and a conventional 2-dimensional partial view from 3DA is a big advantage, not to be overlooked.
By using 3DA we can see that the reason why two amps may sound different despite similar specs, is simply because the specifications or graphs are merely 2-dimensional cross sections. We would have to look at an awful lot of conventional data in order to get the kind of overall view possible with 3DA. The reason why two amps have different sound quality is because their overall performance characteristics are different; such differences can be clearly seen by a single look at a 3DA diagram.

Fig. 2. Third-angle projection of solid body.

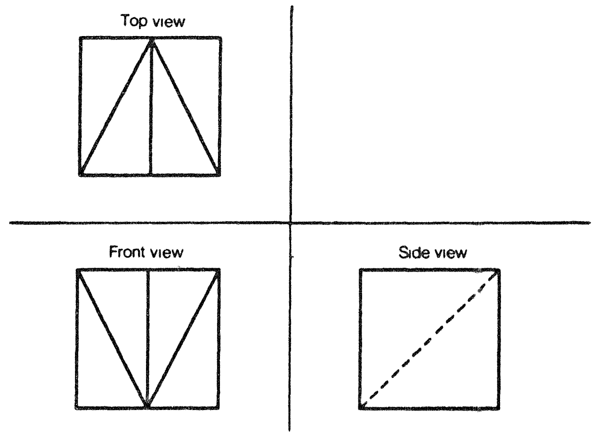


Fig. 3. 3-D diagram of body in figure 2.

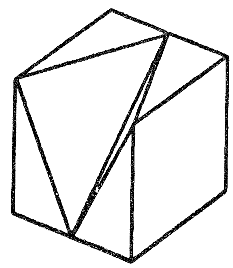
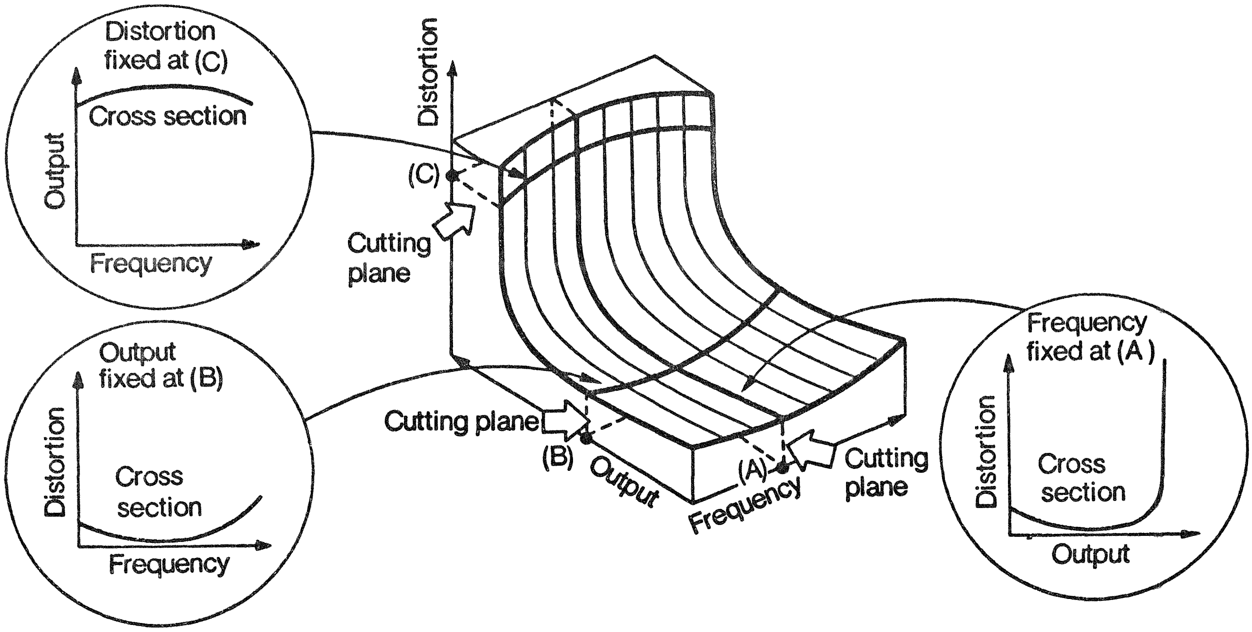


Fig. 4. Reading 3DA data.



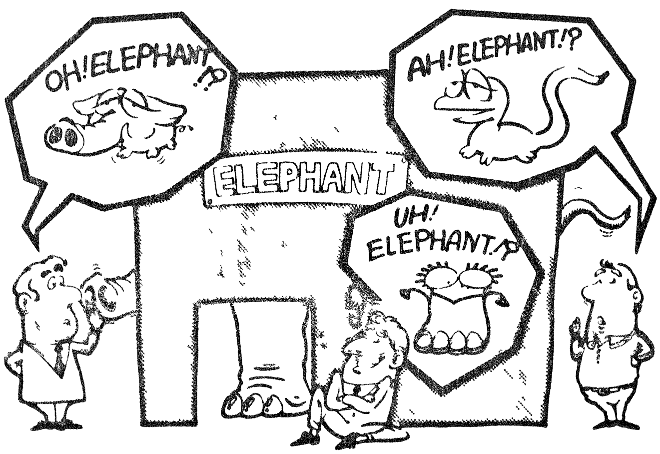
Human perception and actual characteristics

As mentioned above, one cannot do without listening tests when designing and building amplifiers. In the pursuit of sound quality, it is of great importance to understand both the nature of perception and as much as possible about actual characteristics which may affect it, since hearing is one of our senses.

Figure 5-1 shows the famous Muller-Lyer optical illusion. In spite of the fact that the two vertical lines are the same length, they look different. In this way, even vision, supposedly the most accurate of our senses, can be easily deceived due to the conditions under which something is perceived. Much the same is true of human hearing. For example, one's judgement of sound quality would be greatly affected if in the one case one performed a listening test while very tired or in the other if one were very alert and at ease.

If, however, one has a scientific overall grasp of the overall characteristics of the amp being tested, one could estimate the extent that such psychological or physical factors are influencing one's judgement. This would allow the possibility of scientifically correcting for unstable factors. A listening test improved in this way would certainly be a help in building amps with better sound quality.

As a method of analysis that strengthens the connection between listening test results and amplifier characteristics, 3DA is a first step toward a stricter new standard of amplifier judgement.



Can you understand the whole from measurement of a part?

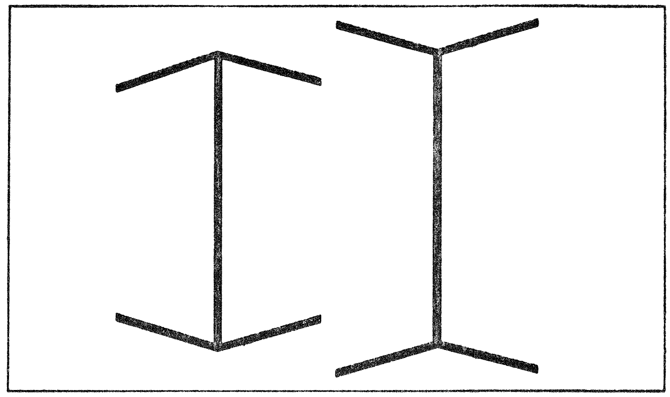


Fig. 5-1. The Müller-Lyer illusion. (Which vertical line looks longer?)

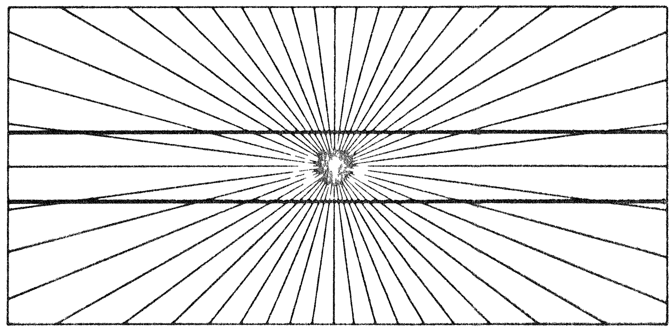


Fig. 5-2. Are the two horizontal lines straight?

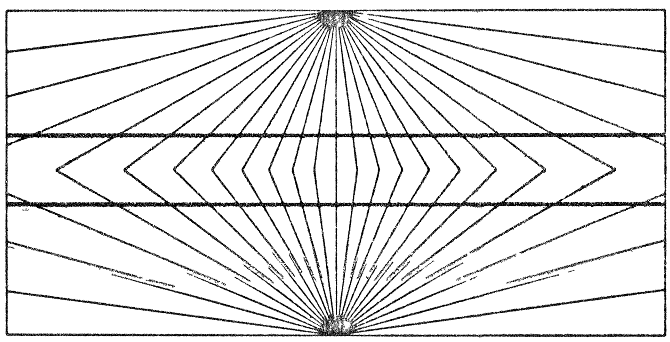


Fig. 5-3. Are the two horizontal lines straight?

Using 3DA in the pursuit of sound quality and performance

Next we will examine 3DA 3-dimensional data for some actual amplifiers. Figures 6 and 7 are 3DA system results for two integrated amplifiers having different kinds of circuitry. In both cases the tuner terminals are used for the input, and the test results are taken from the speaker terminals so that both preamp and power amp sections are being tested.
The characteristics of these two amps look a lot alike if one looks at the conventional output vs. distortion data obtained at 1kHz, but with 3DA data, the differences become quite clear. In these 3-dimensional graphs, one of the bottom axes is output (0.2W~200W), the other is frequency (10Hz~100kHz) and the vertical axis is distortion (0.0001%~0.1%).
Amp A shown in figure 6 is smooth in overall shape with low distortion over both a wide power and frequency range. Amp B in figure 7 shows a rough low range (residual power supply hum) and a rise in distortion at high output as frequency gets higher (which may be due to output transistor switching distortion or the influence of electromagnetic induction from the DC power supply output leads of the power amp section of the amplifier).

The 3DA approach has no parallel anywhere; it is an epoch making original developed by Technics. It permits one to pursue low distortion, flat frequency response, and wide dynamic range in a 3-dimensional way, and thereby obtain better overall performance for significant improvement of sound quality. It also makes possible the concept of a "free music plane" covering the entire area where audio reproduction is kept within the minimum distortion standard of 0.01%. In other words, the free music plane is made up of the parts of the 3DA graph where frequency and output are accompanied by no more than 0.01% distortion. Therefore, if we were to produce 3DA data for an ideal amplifier, it would look like figure 8.
In this way, amplifier quality can be determined by the wideness of this plane. Instead of blindly pursuing performance at single points, 3DA permits one to go after total performance and sound quality by providing an overall, coherent view of total characteristics.

Fig. 6. 3DA characteristics of amp A.

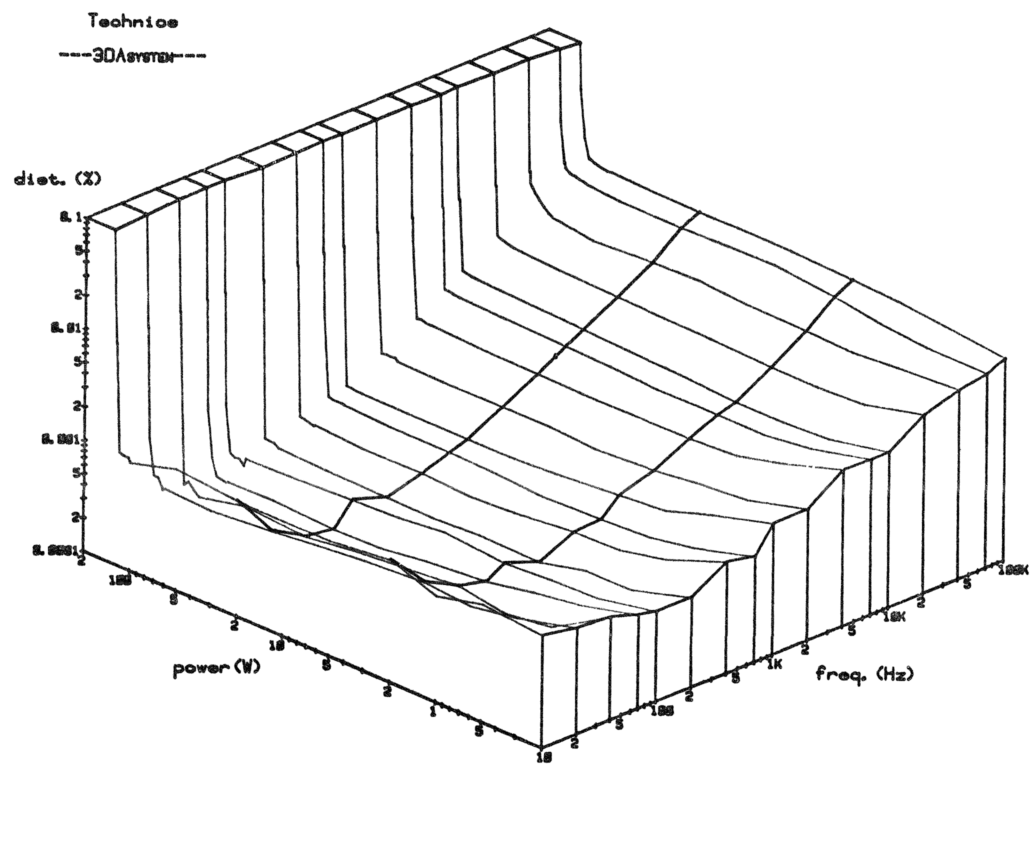


Fig. 7. 3DA characteristics of amp B.

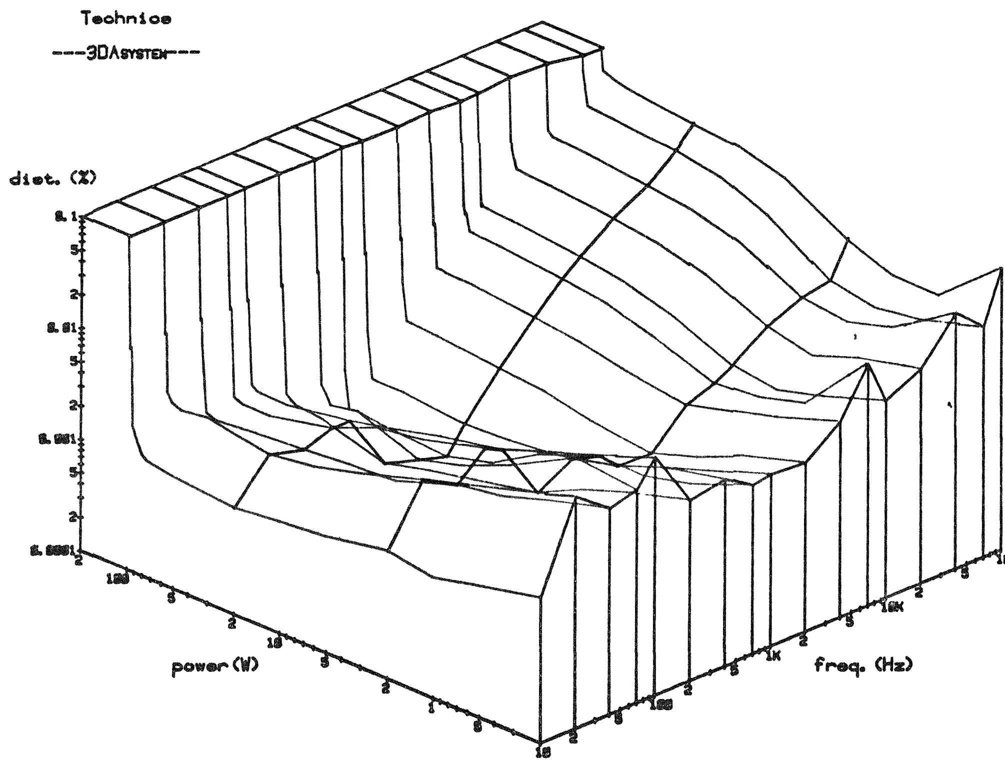
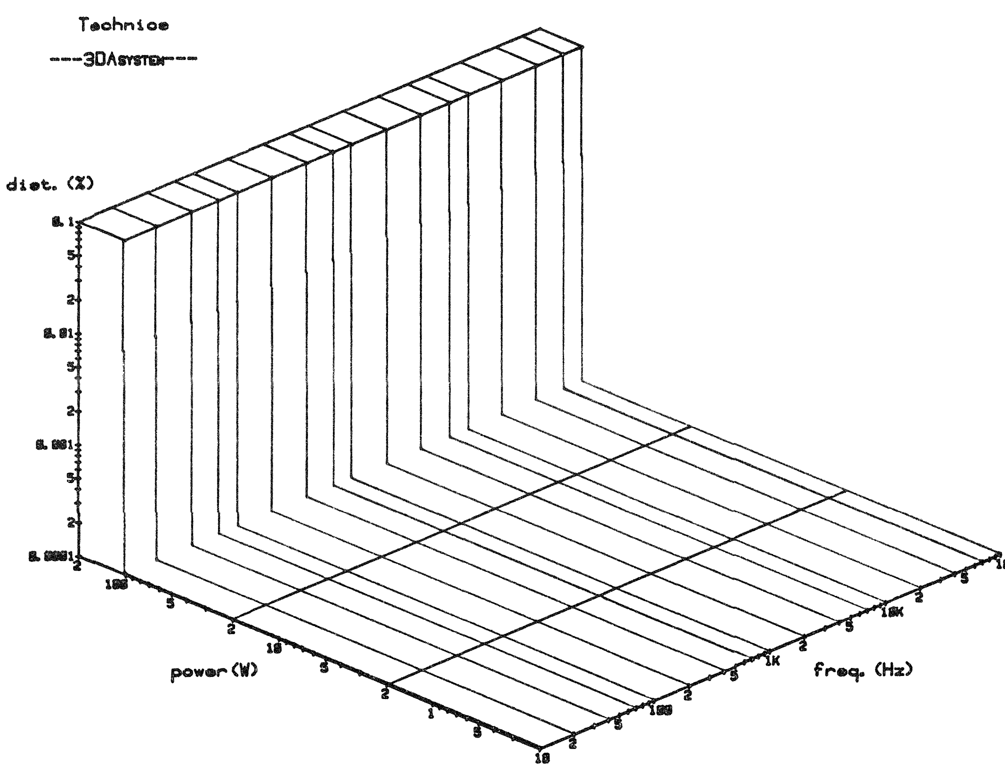


Fig. 8. 3DA characteristics of the ideal amplifier.



Dynamic characteristics and the importance of static characteristics

If one looks at 3DA data in another way, one can see it as the formation of a 3-dimensional graph by means of the addition of a frequency axis to a large number of conventional static characteristic data compiled for output vs. distortion at various frequencies. This raises the question of the relevance of this approach to sound quality which is thought to be a matter of dynamic or transient characteristics.

These days, one hears much talk of slewing rate, rise time, TIM, and other factors involved in transient handling ability of amplifiers.

To build a high-speed amp with a fast rise time, one aims to extend high frequency response. The problem with this kind of approach is that while it may very well contribute to an improvement in sound quality, when one goes after just a single item, one can too easily lose sight of other factors which may result in decreased fidelity in some other area. With this in mind, it seems relevent to examine the connection between such transient characteristics and their corresponding static characteristics.

Slewing Rate

If one considers this factor to be a numerical indication (V/μs) of how high frequency a signal can be faithfully reproduced at high output and low distortion, one can see that slewing rate corresponds, as is, to conventional so-called "static" data for maximum output vs. frequency response at a fixed level of distortion (1% for example). You can see this at a glance in the 3DA data for the amplifier shown in figure 9. For an amplifier with a high slewing rate, maximum output would extend as shown by the dotted line.

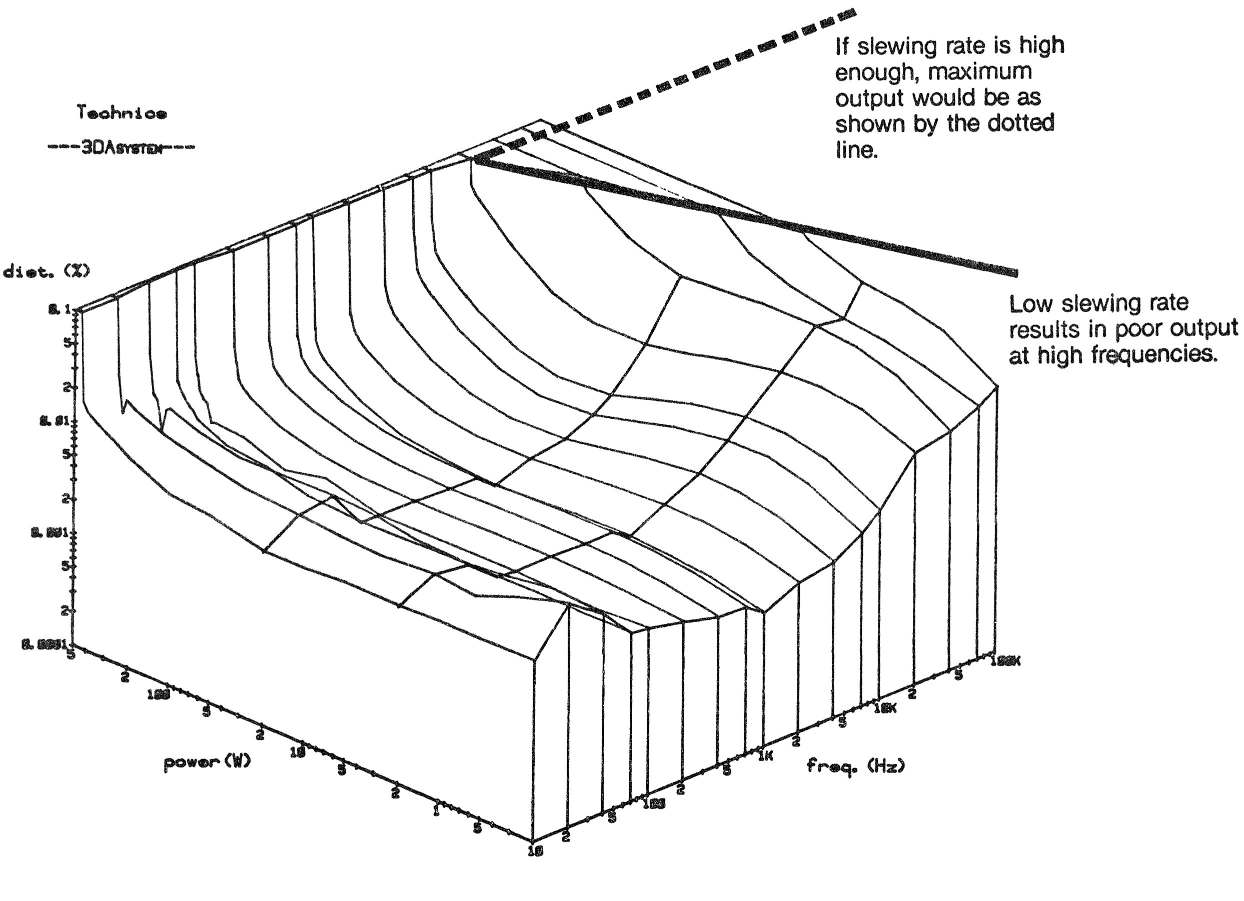
Rise time

This is a matter of how fast the amp can respond to an input signal with a fast attack time. Such a signal has many high frequency components; the faster the rise time, the more high frequency signals make up the input signal in question. Therefore, an amplifier needs good high frequency response in order to exhibit a fast rise time. It follows, theoretically, that by extending frequency response, which represents a static characteristic, one can obtain a faster rise time.

TIM (Transient Intermodulation)

When a square wave or similar waveform having many high frequency components is used as the input, if the signal to be fed back has too high a high frequency signal level, part of the signal will be clipped and a feedback defeating phenomenon will occur. This means that it is possible to find out whether an amp will generate TIM by investigating the relationship between what is conventionally considered the amp's frequen-

Fig. 9. 3DA characteristics of a low slewing rate amp.



cy response and what changes occur to which frequencies at maximum output (the clipping point).
As described above, we can improve performance by studying and controlling the interrelationships between the so-called static characteristics of distortion, dynamic range, and frequency response.
In short, we cannot do without thorough improvement of static characteristics if we want better sound quality. (However, one prerequisite for this is that the power supply circuitry must be stabilized so that when there is a large transient input, the operational points of each of the amplification stages will not be upset.) Once again, the important point is that we must have an accurate way of looking at many static characteristics at the same time, rather than paying attention to only a single item. Since 3DA allows us to analyse the three important factors affecting sound quality in a more comprehensive and more accurate manner, it can be of great help in the improvement of not only static characteristics but dynamic transient characteristics as well. Furthermore, it can aid us by preventing us from losing sight of overall performance even if we must work on improving single aspects of performance at a time.
Figure 10, 11 and 12 show 3DA results for several amplifiers. These are presented for your reference.

Fig. 10. 3DA results for circuit with high overall distortion level.

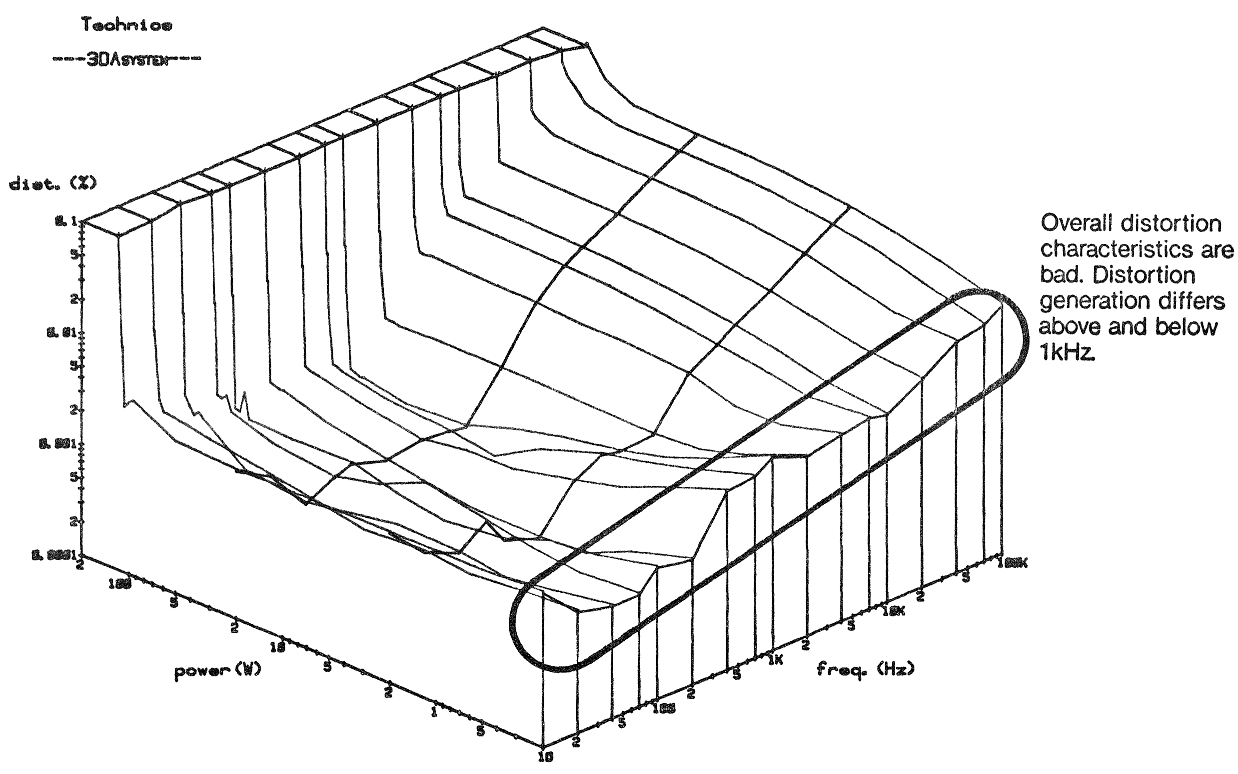


Fig. 11. 3DA results for circuit with high output idling current.

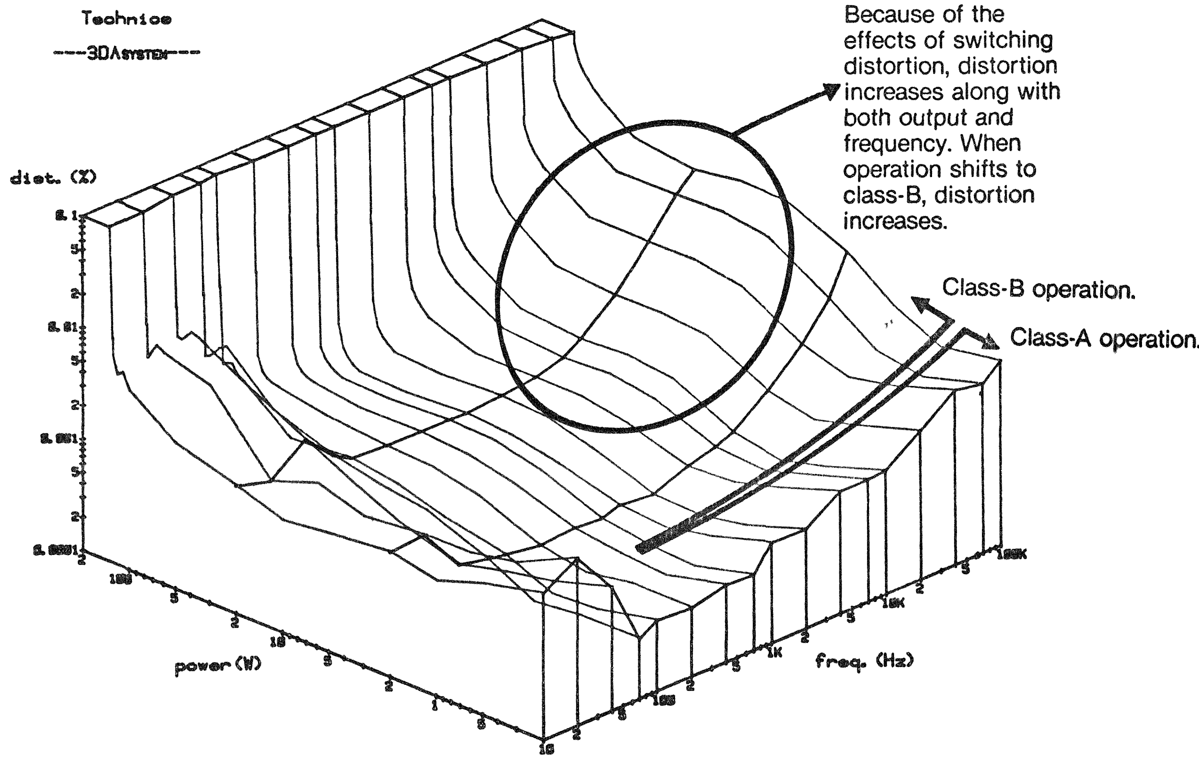
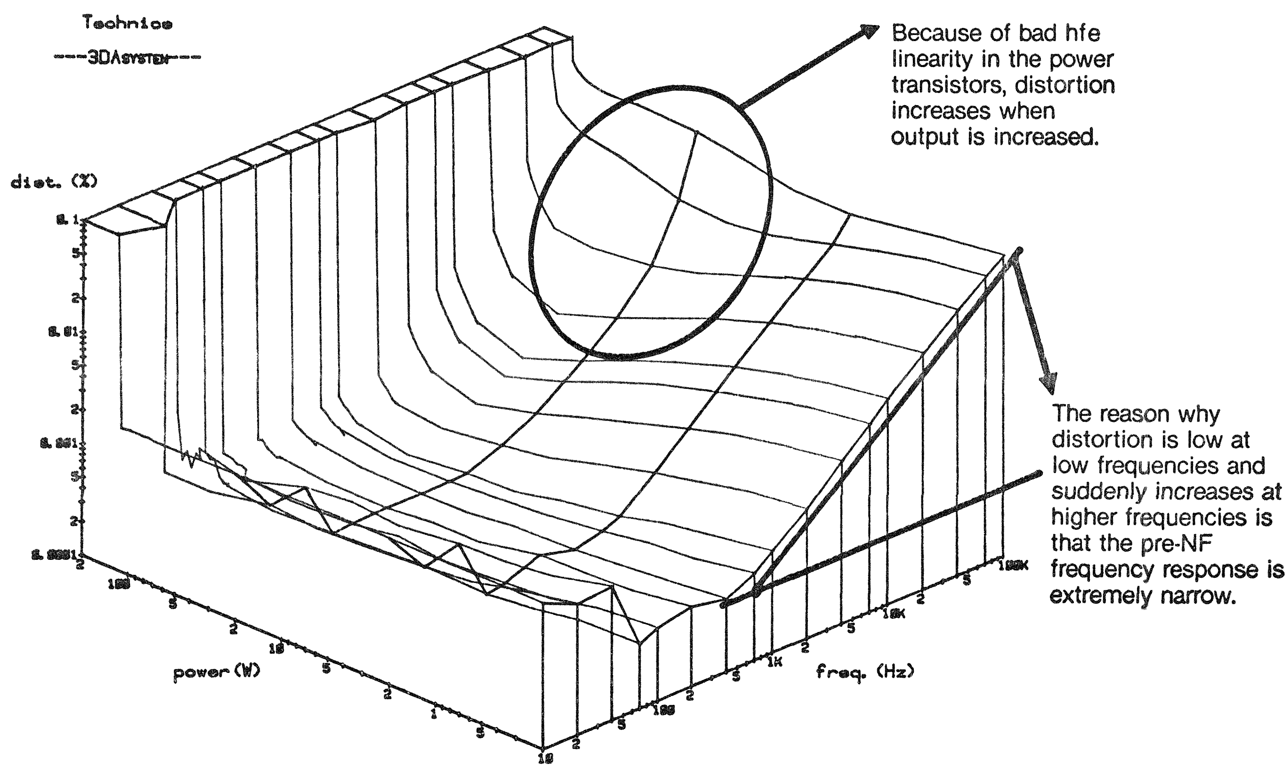


Fig. 12. 3DA results for amp with high midrange NF.



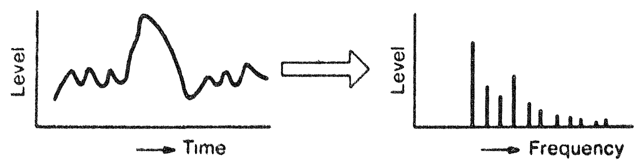
3DA: The New Comprehensive Method for Accurate Analysis of Both Static and Transient Characteristics.

Technics' 3-dimensional analysis is an advanced method of comprehensively correlating output power, distortion, and frequency, the three basic factors that determine an amplifier's basic performance. A computer is used to draw a 3-dimensional graphic profile of the relationships between these factors to enable quick and easy judgement of overall performance characteristics.

Although output, distortion, and frequency interrelationships are commonly categorized as static characteristics, 3DA is so accurate and comprehensive that upon close examination it also reveals an amplifier's transient characteristics, including TIM, slewing rate, and rise time. The following material should make clear, in the light of 3DA, how the complex questions of transient performance can be answered by means of a more thorough understanding of static characteristics.

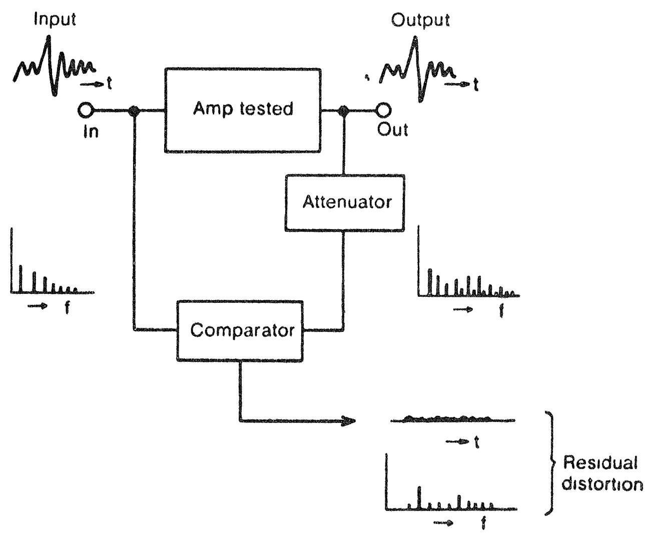
The validity of using 3DA to understand transient performance is supported by Fourier's theorem which states that any complex waveform can be broken down into a infinite number of sine waves.

The Fourier transform clearly shows the connection between information charted on a time axis and on a frequency axis. By means of this approach, one can see that what at first seems a very complicated music signal is in fact entirely composed of a number of pure sine waves.



Time axis → Frequency axis

In this context, consider the cliche about "a straight wire with gain". Such an ideal amp would only amplify the level of each of the original sine wave components; it would neither remove, nor add new frequencies. If we were to attenuate the output of our ideal amp by an amount equal to its gain, and then compare it to the input signal, there would be no difference in amplitude at any frequency. This I/O (input/output) comparison method is shown in the diagram below. Unfortunately, I/O comparison will reveal a difference in a real amp. The output components that make up this difference are called distortion.



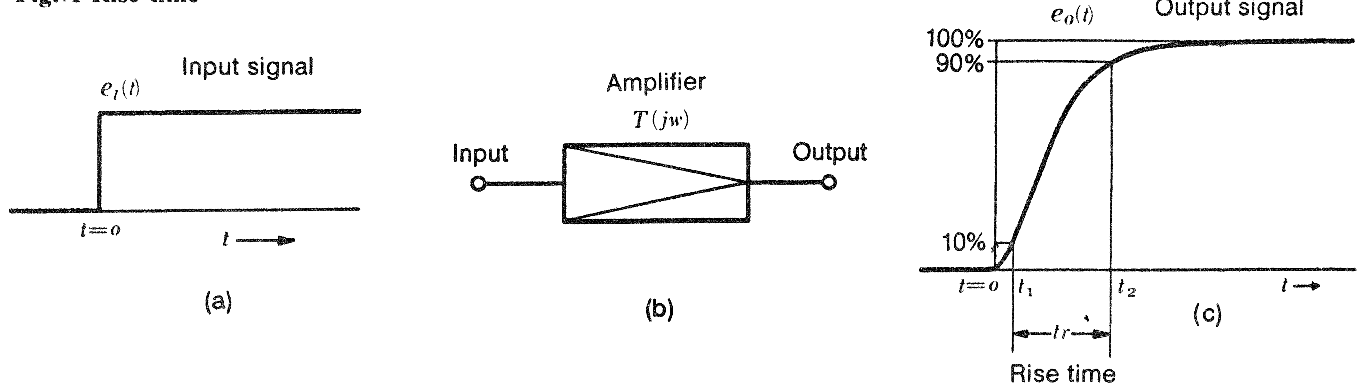
Residual distortion (Distortion revealed by I/O comparison)

- * Caused by non-linear amp performace
  - Non-clipping distortion— Caused by circuit components or circuit non-linearity.
  - Clipping distortion—
    - Power supply voltage clipping in the low frequency range.
    - **High range clipping which arises within the circuitry.**

      This is what is conventionally explained in terms of transient characteristics, including slewing rate and TIM.
- * Caused by differences in the amp's frequency response.
- * Noise produced within the amp.

1. Rise time

Fig. 1 Rise time



A definition:

When a step input (Fig. 1a) is applied to an amp's signal path circuitry, the output waveform can be viewed in the context of Fig. 1c. "Rise time" is the time it takes for the output signal level to increase in amplitude from 10% to 90% of its peak value.

If the amp's transmission characteristics and the input signal content are known, one can use complex mathematical computations (involving integral-differential equations) to obtain the waveform by solving for t, from which we can derive the rise time.

To avoid this difficult procedure, it is much more common to just use visual observation of the waveform. A much simpler algebraic alternative for finding time response is to use the Laplace transform, which bypasses step-by-step calculation of differential equations. By using the Laplace transform on the input waveform and transmission characteristics, this mathematical substitution method converts a time domain phenomenon into an algebraic equation (as multiplication becomes addition in logarithmic computation) in the complex frequency domain (a function F(s) in the complex domain); after solving this equation, the inverse Laplace transform returns the result to the time (real) domain, thus allowing one to find the rise time.

1. Rise time in an amp having −6dB attenuation in the high range.

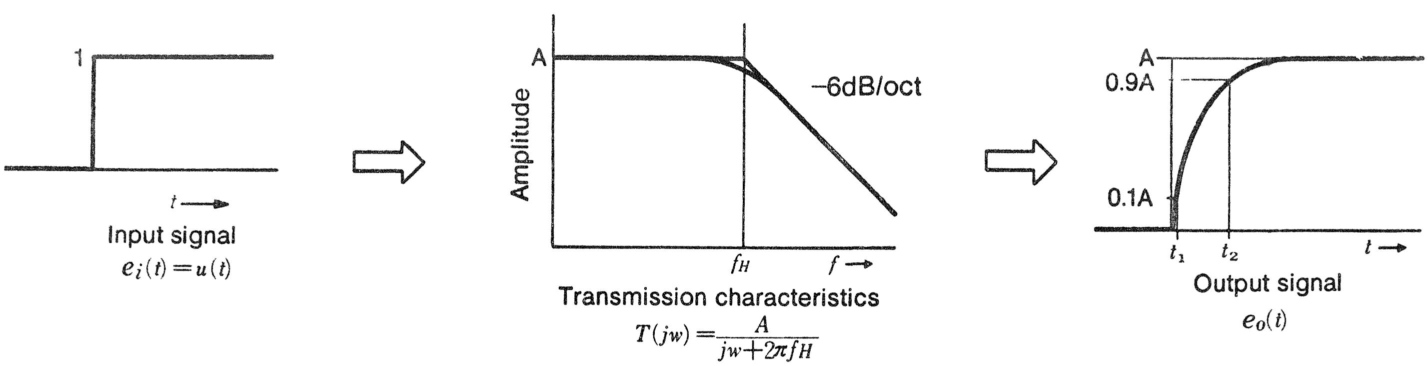
Given a −6dB/oct, attenuation from the cutoff frequency (f_H), since the Fig. 1 input signal, $e_i(t)$ is a step function, we get $\mathscr{L}[e_i(t)] = E_i(s) = \frac{1}{s}$; and since the transmission characteristics are described as −6dB/oct, at frequencies above f_H, $T(j\omega) = \frac{A}{j\omega + 2\pi f_H}$,

we get $\mathscr{L}[T(j\omega)] = T(s) = \frac{A}{s + 2\pi f_H}$

Consequently, the output signal $e_o(t)$ becomes

$\mathscr{L}[e_o(t)] = E_o(s) = E_i(s) \times T(s)$

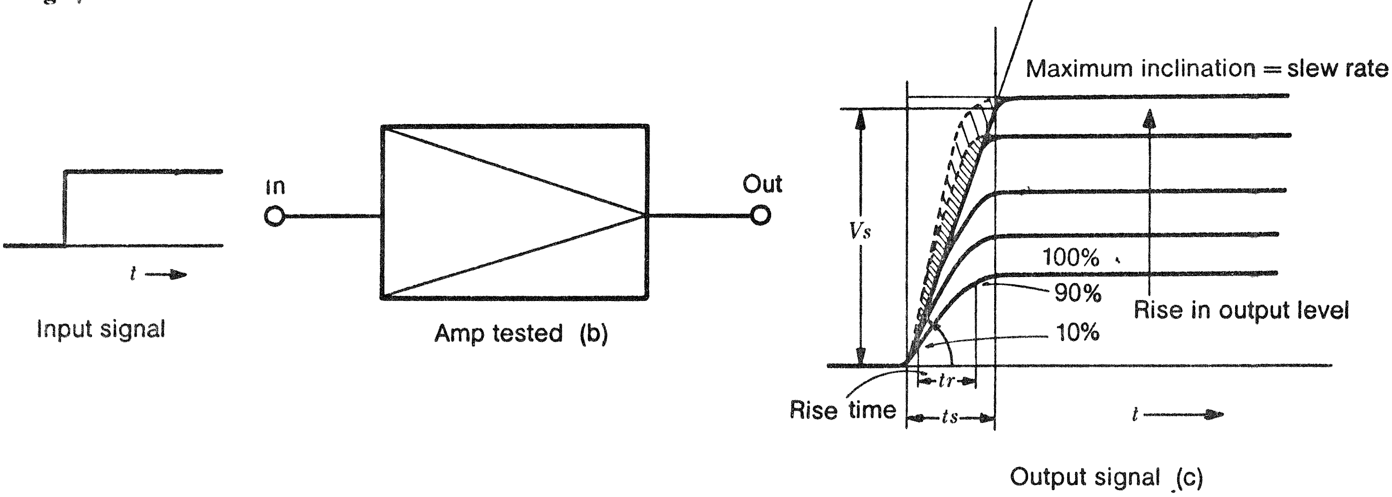
and the result is obtained as the product of the Laplace transform of the input signal and transmission characteristics.



$$E_i(s) = \frac{1}{s} \qquad \times \qquad T(s) = \frac{A}{s + 2\pi f_H} \qquad = \qquad E_o(s) = \frac{A}{s(s + 2\pi f_H)}$$

2. Slew rate: amplifier performance under large-signal conditions.

Fig. 2 Slew rate

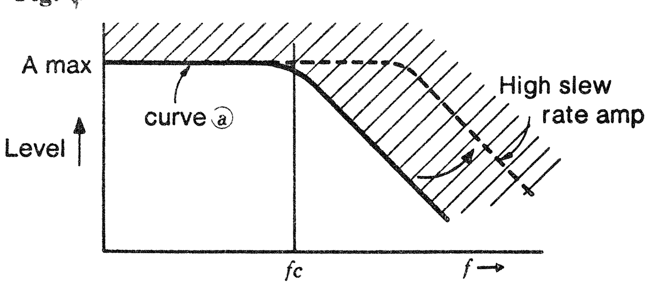


1) Like rise time, slew rate is a factor that describes amplifier response for step input. While rise time expresses response for a step input when the amplifier is operated within its linear region, slew rate is measured by increasing the signal level (Fig. 2c) until the leading edge of the output signal reaches the maximum inclination obtainable; this is expressed in V/μs. This means that a slew rate evaluation of response for a step input includes performance in the amplifier's nonlinear region of operation. Therefore, slew rate is related to an amp's clipping level.

2) This brings us to the matter of frequency response at the clipping level (or at a certain, set distortion level), taking as our example an amp in which the maximum output voltage decreases with frequency (Fig. 3), the shaded area in the diagram on the left covers the region of nonlinear operation, in which clipping prevents the amp from being able to reproduce the signal properly. This response can be expressed as:

$$T_{max}(j\omega)=\frac{A_{max}}{j\omega+2\pi f_c}$$

Fig. 3

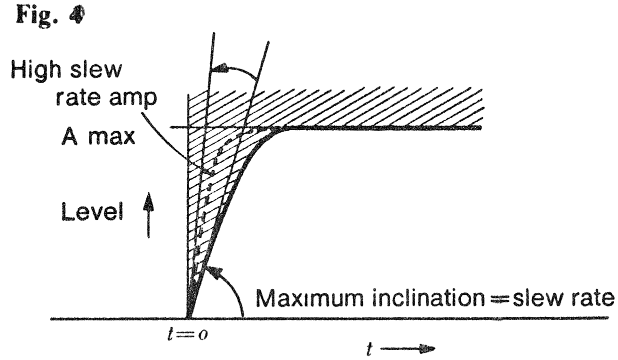


Therefore, an undistorted output waveform for the step input can be expressed with the same formula as (1) above:

$$e_o(t)=A_{max}(1-e^{-2\pi f_c t})$$

Viewed in the time domain (Fig. 4), the shaded portion

Fig. 4



of the graph indicates amplifier operation in the nonlinear region, which is to say, the region in which the signal distorts. Since slew rate expresses the maximum inclination at this point, it can be found by differentiating $e_o(t)=A_{max}(1-e^{-2\pi f_c t})$ and seeking the maximum value.

$$e_o(t)=A_{max}(1-e^{-2\pi f_c t}),\quad \frac{de_o(t)}{dt}=A_{max}\cdot 2\pi f_c\, e^{-2\pi f_c t}$$

$$\left[\frac{d\,e_o(t)}{dt}\right]_{max}=\left[\frac{de_o(t)}{dt}\right]_{t=0}=\boxed{2\pi f_c\cdot A_{max}=\text{slew rate}}\quad\ldots.(4)$$

Using the inverse Laplace transform on $E_0(s)$, we get

$$e_o(t)=A(1-e^{-2\pi f_H t}) \qquad \ldots\ldots\ldots\ldots\ldots(1)$$

From this, if we pursue t_1 and t_2 for the times at 10% and 90% amplitude, we get:

$$0.1A=A(1-e^{-2\pi f_H t_1}) \rightarrow \quad 2\pi f_H t_1=-\ln 0.9$$

$$0.9A=A(1-e^{-2\pi f_H t_2}) \rightarrow \quad 2\pi f_H t_2=-\ln 0.1$$

Consequently, rise time t_r is given by the formula (2):

$$t_r=t_2-t_1=\frac{\ln 0.9-\ln 0.1}{2\pi f_H}\doteqdot\frac{0.35}{f_H} \qquad \ldots\ldots(2)$$

This shows that the higher the amp's cutoff frequency (refering to high-range frequency response, f_H is the frequency point having −3dB amplitude compared with the midrange), the shorter the rise time.

For example, if the −3dB cutoff frequency is 100kHz, t_r = 3.5 μs. For 1MHz, t_r = 0.35 μs.

Since this makes clear the direct correspondence between rise time and frequency response, the question arises as to the need for an extremely short rise time, *per se,* for faithful reproduction of the audio frequency range.

2. Rise time in an amp with a frequency response having −12dB/oct, attenuation in the high range.

In this case a quadratic equation for frequency response is used for the transmission characteristic $T(jw)$,

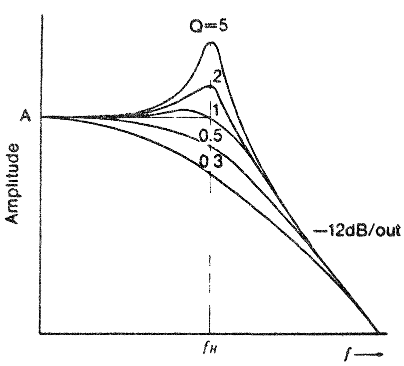
$$T(j\omega)=\frac{A}{(j\omega)^2+\frac{2\pi f_H}{Q}j\omega+(2\pi f_H)^2}$$

Therefore

$$T(s)=\frac{A}{s^2+\frac{2\pi f_H}{Q}s+(2\pi f_H)^2}$$

Therefore

$$E_o(s)=\frac{1}{s}\times\frac{A}{s^2+\frac{2\pi f_H}{Q}s+(2\pi f_H)^2}$$



Note: The −12dB/oct, transmission characteristic can be found if f_H and Q are known. If we then use the Laplace transform to compute the response to the step function in the time domain, we obtain:

$$e_o(t)=\begin{cases}\text{If } Q<0.5 \\ A\left\{1-\dfrac{e^{-(\pi f_H/Q)t}}{\sqrt{\frac{1}{4Q^2}-1}}\sinh\left(2\pi f_H\sqrt{\frac{1}{4Q^2}-1}\,t+\tanh^{-1}2Q\sqrt{\frac{1}{4Q^2}-1}\right)\right\} \\ \text{If } Q=0.5 \\ A\,\{1-(1+2\pi f_H t)\,e^{-2\pi f_H t}\} \\ \text{If } Q>0.5 \\ A\left\{1-\dfrac{e^{-(\pi f_H/Q)t}}{\sqrt{1-\frac{1}{4Q^2}}}\sin\left(2\pi f_H\sqrt{1-\frac{1}{4Q^2}}\,t+\tan^{-1}2Q\sqrt{1-\frac{1}{4Q^2}}\right)\right\}\end{cases} \qquad \ldots\ldots\ldots\ldots(3)$$

In this way, the output waveform is principally determined by the transmission characteristics (the cutoff frequency f_H and the value of Q at that point). For example, if we seek the rise time for Q = 0.5 in the equation (3) above, we can compute the solution by seeking t_1 and t_2 which are $e_o(t_1)$ = 0.1A and $e_o(t_2)$ = 0.9A. Thus we obtain:

$$t_r=t_2-t_1\doteqdot\frac{0.534}{f_H} \qquad (Q=0.5)$$

Likewise, for Q = 0.7

$$t_r=t_2-t_1\doteqdot\frac{0.342}{f_H} \qquad (Q=0.7)$$

Note: Ordinarily, there is just a change in the constant when Q changes. When Q becomes larger, the constant becomes smaller.

3. Some conclusions about rise time.

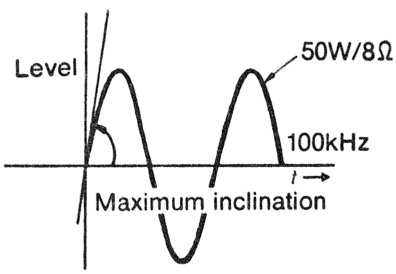
We come to the following conclusions after considering rise time in the light of the above calculations:

1) Rise time indicates how far an amp's high range frequency response extends. The higher the response, the shorter the rise time.
2) Rise time is not a factor which indicates an amp's transient characteristics; rather, it exactly corresponds with conventional amplitude frequency response, which is to say that rise time expresses static characteristics.
3) This means that the advertising appeals based on fast rise time specifications (of 0.5 μs, 0.75 μs, etc.) for "high speed amps" have no significance and in fact mean nothing more than extended frequency response (up to around 700kHz, 470kHz, etc.).
4) An amp's reproduction capability within the 20 ~ 20,000 Hz audio band is important for sound quality. There remains some doubt about the viability of merely extending high range frequency response as a means of pursueing superior amplifier performance.
5) While a fast rise time may be one sign of a good amplifier as such, when it comes to the demands made on an *audio* amp, reckless pursuit of faster rise times may be nonsensical. (Extended high range frequency response also makes the amplifier more sensitive to RF interference and spurious noise interference from electrical appliance switches, which could cause shock noise.)

In this way, it is possible to compute the slew rate from the frequency response at the clipping level if one knows A_{max} and f_c. This means that slew rate is just another way of talking about the data in a graph of an amp's frequency response at the clipping point; it just changes the terminology into V/μs to refer to the maximum inclination that can be reproduced for a stepped input. Sine waves can be used to measure clipping level frequency response and, since clipping can be measured as harmonic distortion, one can easily understand the phenomenon from specifications expressing (max.) power vs. frequency response at a set distortion level (power bandwidth).

Slew rate has become a major audio topic because it supposedly indicates an amp's transient characteristics, but under this kind of careful scrutiny, slew rate turns out to be the same thing as frequency response at the clipping level, a matter quite sufficiently expressed by conventional static measurement data using sine waves. Both the broken line waveform in Fig. 4 and the formula (4) clearly show that the amp having extended high range frequency response at the clipping level (broken line in Fig. 3) will have a higher slew rate.

3) In this conneciton, let us seek the slew rate necessary for undistorted reproduction of a 50W/8Ω, 100kHz waveform.



Since a 50W/8Ω, 100kHz signal is a $20V_{RMS}/8\Omega$, 100kHz signal, it can be expressed as

$$e(t)=\sqrt{2}\times 20 \times \sin(2\pi\times 100\times 10^3 t)\,[V].$$

The maximum inclination of this signal can be found by seeking the maximum value for $\frac{de(t)}{dt}$

$$\therefore \left[\frac{de(t)}{dt}\right]_{max}=\left[\frac{de(t)}{dt}\right]_{t=0}=\sqrt{2}\times 20\times 2\pi\times 100\times 10^3\,[V/s] \fallingdotseq 1.78\times 10^7\,[V/s]=17.8\,[V/\mu s]=S.R.$$

Which is to say that if the amp has a slew rate of 17.8V/μs or more, it will be able to reproduce the above input signal. In the same way, a slew rate of at least 25.2V/μs is sufficient to reproduce a 100W/8Ω, 100kHz input signal.

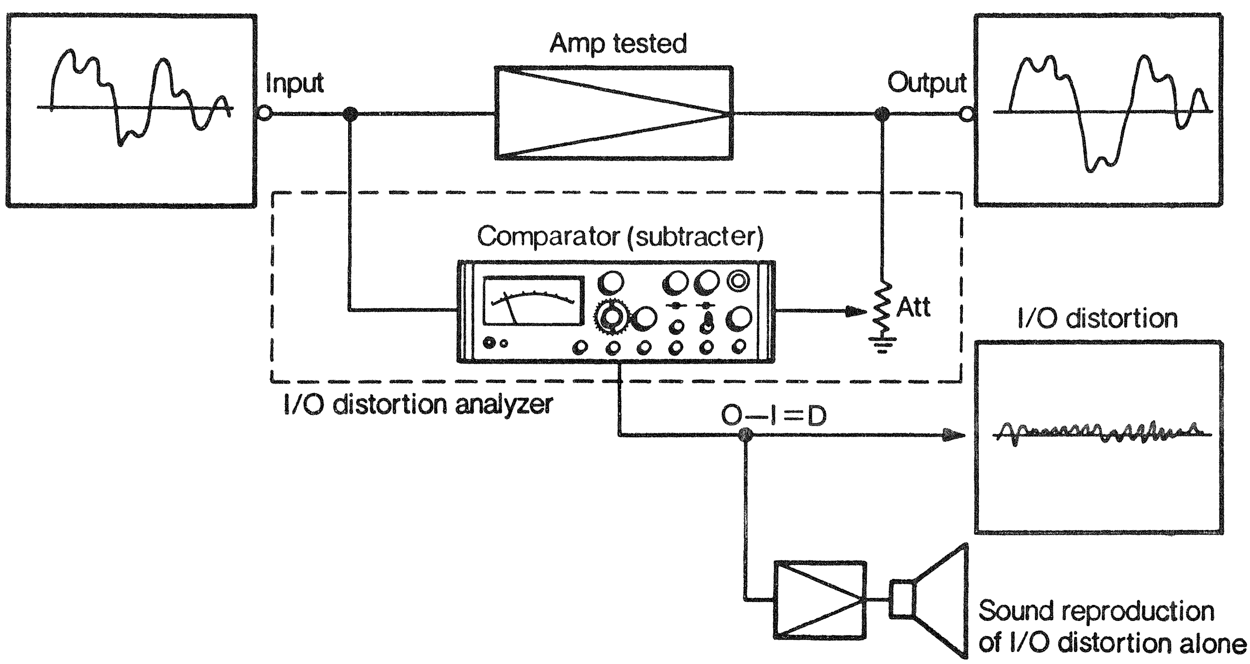
Some conclusions about slew rate.

1. Slew rate is equivalent to frequency response at clipping level.
2. The further the frequency response extends into the high range at clipping level, and the greater the maximum output, the higher the slew rate will be.
3. Although slew rate is often considered as a special factor describing transient characteristics, there is no basis for this claim. Slew rate is exactly the same thing as viewing output level vs. frequency within the amp's nonlinear operating region or clipping level, which may also be considered as a set level of harmonic distortion.
4. While some amp manufacturers are claiming superior performance on the basis of such large slew rates as 170V/μs or 200V/μs, if we assume the amp is 100W/8Ω, this would necessarily mean that it could produce a 100W output from 670kHz to 800kHz, but if we try to test for this performance, smoke will pour from the amp and it will not operate normally. The fact is that it cannot deliver 100W in the 670kHz ~ 800kHz range. All these slew rates really indicate is that the manufacturer's amp's output waveform for a step input has a maximum inclination with which the listed slew rate is made to correspond. The idea that this approach has anything to do with the pursuit of high fidelity reproduction within the 20 ~ 20,000Hz audio frequency range seems highly dubious.

I/O Distortion Analysis
(Input-output transfer distortion)

A great deal of useful information can be obtained by using 3DA data to examine amplifier performance via it's overall static characteristics. However, it goes without saying that one can object to this approach on the grounds that it does not actually go beyond static characteristics, (even if it is possible to analyse transient phenomena in static terms). Therefore we need some way of proving that an amp with good overall static characteristics will exhibit good dynamic transient characteristics as well.
This means that we need a different measurement method so that we can verify actual amplifier operation when using a music signal as the input. It certainly would not hurt if we were to examine actual operation with a musical input in order to confirm an amp's sound quality and performance characteristics. For this purpose Technics has developed an I/O distortion analyzer that allows real-time analysis of distortion components, and which uses a musical signal as the input. Such analysis obviously includes the true dynamic characteristics of an amplifier.
In this method, the degree to which the input and output waveforms resemble each other is investigated, thus allowing us to directly view distortion or frequency response problems occuring within the amplifier. No longer is it necessary to use a sine wave input to be able to compare the input and output waveforms. With this method one can examine the degree of similarity even with a complex musical signal. Figure 13 shows the operational principle of this system.
If, for example, an amp generates TIM distortion, you will be able to see this by looking at the I/O distortion meter. Furthermore, you can listen to just the sound of the TIM distortion.

Fig. 13. I/O (input/output) distortion measurement system.



I/O Distortion Analyzer

Figure 14 shows a block diagram of the I/O distortion analyzer. At the amp's output side, an attenuator reduces the level to that of the input level; at the input side is a correction circuit for the amp's frequency response; both these signals are fed to the subtracter circuit. In this respect, while it is true that the accuracy of I/O distortion measurement is determined by the performance of the subtracter, this in turn depends on the subtracter's CMRR (common mode rejection ratio) characteristics which are as shown in figure 15. This level of performance corresponds with that of conventional distortion meters with T.H.D. at about 0.001%.

Fig. 14. I/O distortion analyzer with block diagram.

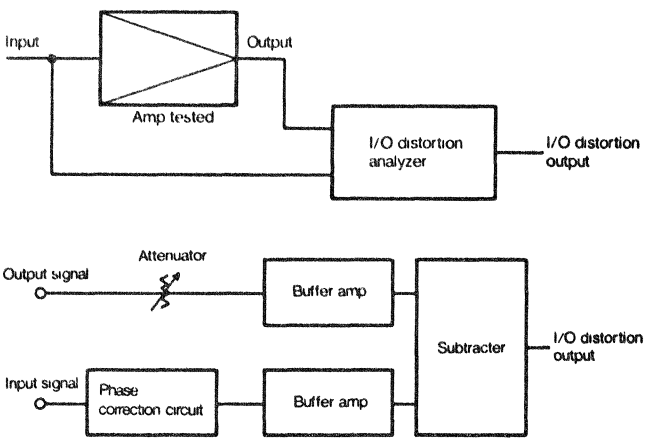
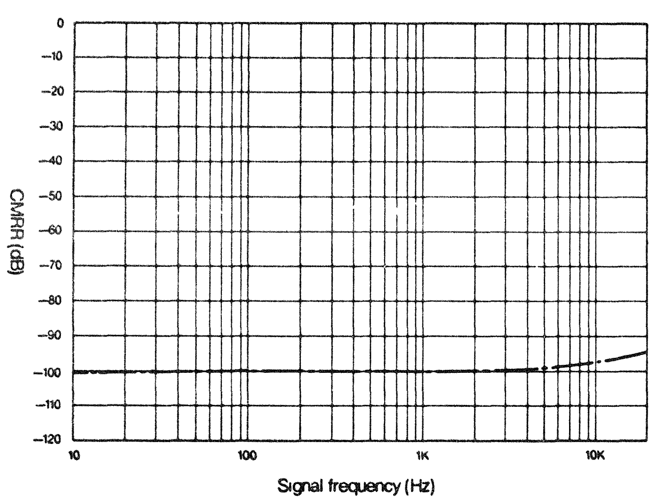
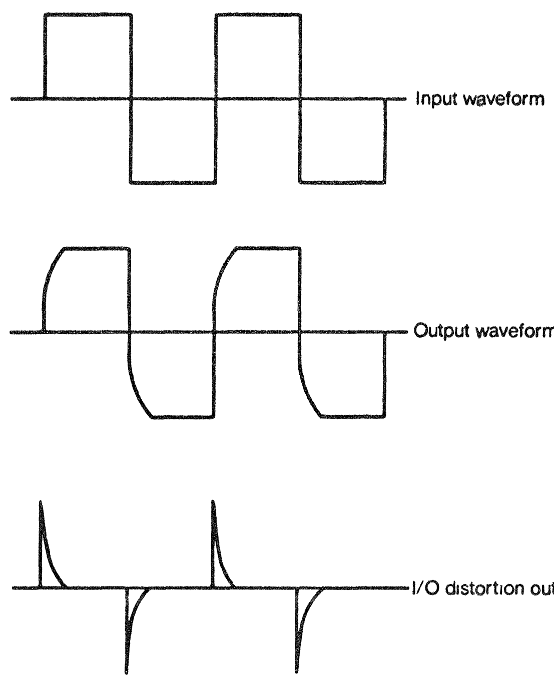


Fig. 15. Measurement limits of I/O distortion analyzer.



The frequency response correction circuit in the input is provided to prevent errors due to change in waveform shape caused by the amplifier's frequency response. For example, if the amp's frequency response shows a dip in the high range, square wave reproduction will exhibit a rounded leading edge. If the output waveform and input waveform were reduced to the same level and subtracted, this difference in the leading edge would show up as I/O distortion. This is represented in figure 16. Such I/O distortion is merely a result of frequency response, it is not really dynamic distortion such as TIM, or even T.H.D. This is why it is necessary to correct phase and frequency response. A low-pass filter with variable C and R components forms the phase correction circuit used to make the subtracter input match the frequency-response-caused phase characteristics of the amplifier. Of course, phase correction is unnecessary if frequency response differences are to be included in I/O distortion measurements. On the other hand, phase correction circuitry is quite important for the measurement of TIM or T.H.D.

Fig. 16. I/O distortion caused by the amp's frequency response.



Examples of I/O distortion measurement

Photo 2 shows I/O distortion results for a good amp using a music signal as the input. From the top down are shown the input waveform, the output waveform, and the I/O distortion waveform. The I/O distortion meter scale is graduated in 0.1% (peak to peak) increments. Therefore, the distortion generated by this amp is extremely low, especially when one considers the fact that these results were obtained with a true transient input consisting of a music signal having a very random waveform.

Photo 3 shows the results obtained from an amp known to have much third harmonic distortion as revealed by 3DA data. Looking at the I/O results, one can see the generation of distortion components not exhibited in Photo 2. The most pronounced distortion is evident during the rising edge of the signal. When one listens to the distortion signal alone, it sounds very dirty and muddy.

One can see in the photo that the input and output waveforms resemble each other quite closely, but thanks to the I/O distortion meter one can discover just how much distortion is really being generated.

This unique method allows us to check how close amp performance comes to our waveform fidelity ideal of amplification without either adding to or taking away from the input waveform, even when that input is a complex musical signal. The closer the I/O meter distortion output comes to "0", the greater the waveform fidelity of the amplifier. Perhaps it would not be going too far to say that you could express everything about an amplifier by listing specifications for only I/O distortion and maximum output.

Photo 2. Example of I/O distortion measurement (good).

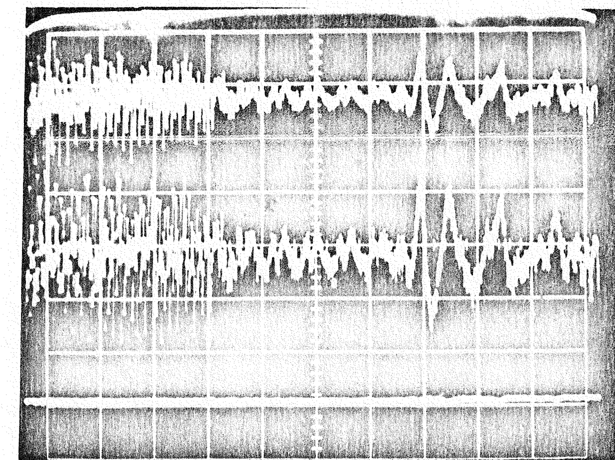
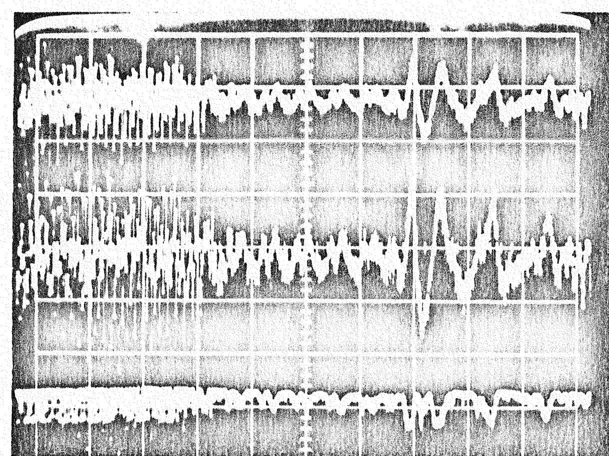


Photo 3. Example of I/O distortion measurement (bad).



Conclusion

Using the I/O distortion analyzer, a unique new measurement method, Technics has proved that an amp having good overall static characteristics, as represented by a wide "free music plane" through 3DA analysis, will also exhibit outstanding performance during actual operation when a music signal is used as the input.

In the future this approach should find wide application not merely with amplifiers, but at every step of the recording and reproduction chain. It would help even more if we could apply such test methods to each and every component and part used in audio equipment. In the mean time, using these new scientific "eyes and ears" we will continue in our efforts to build amplifiers with both superior specifications and audibly better sound quality.

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