How Sansui Amplifiers are Designed

An effort to end of TIM Distortion.

by Basil Lane

Sansui

Introduction

Most of us take watches for granted. It is very easy to tell if they are accurate and we know that although the mechanism is complex, it is unnecessary to understand how it works. Thus, we tend to buy a watch on its features—its appearance, if it has hands, an LED or liquid crystal display or if it has quartz crystal control. We also know that the watch designers instruments will tell him all he needs to know about the performance of the watch.

Amplifiers may be compared to watches in some respects, since they too are precision devices and their circuitry is often complex and mysterious to the layman. However, the similarity ends there because the listener can often observe strange things happening to the sound produced via an amplifier that the designer, in the past, has been unable to identify with all the instruments at his command.

Of course, the watch is a rather simple, single function device. Its mechanism is designed to operate at a single rate and thus performance can be readily predicted or measured. The amplifier however, has to perform a wide range of functions and the signals it has to deal with range from the lowest organ note to the highest harmonic of a violin string; from the whisper of a childs voice to the crescendo of a full concert orchestra. Until recently, it has been almost impossible to provide a predictable laboratory substitute for all these signals, such that the full performance of the amplifier can be evaluated.

Experienced ears have often detected differences between two amplifiers that have had apparently identical specifications. The data that is usually published to describe the performance of an amplifier arises often from the use of what are called steady state test signals. These are simple repetetive tones or pulses which bear little resemblance to the sort of highly variable signal that arises from music or speech.

It is little wonder therefore that the basic static test methods failed to show why amplifiers sounded different to each other. Understanding why these differences occur is obviously a job for an expert designer. It is also his job to solve the problem and produce an amplifier free from all audible faults. But to appreciate the improvements in performance, the customer is often called upon to understand highly technical information presented in a jargon which can be tedious to read and difficult to understand.

In this booklet, the discoveries of one of the world's foremost authorities on audio amplifiers, Susumu Takahashi, are explained first in simple terms for the layman and then in more technical detail in the second part. Hopefully this will satisfy the interests of the tyro and also the curiousity of the more technically minded.

Part 1

What is TIM-D?

Transient Intermodulation Distortion (TIM-D) is a comparatively new spectre haunting the pages of hi-fi magazines. Since 1970, when it was first metioned in a research paper by Matti Otala, amplifier engineers have been seeking ways to measure and eliminate it.

Modern amplifiers usually have superb paper specifications, but still audio critics and expert designers have noticed that all is not perfect. Amplifiers unaccountably sounded different to each other, so whatever the problem was, it was not being revealed by normal so-called static signal tests.

Since 1970, a considerable body of research work has been undertaken to understand why this was happening and now we know that to a large extent, TIM-D was the villain of the piece.

As its name suggests, it is a distortion which is transitory in nature and is initiated by sudden sharp transients in the input to the amplifier. Typical of these would be a rim shot on a drum, the sound of a pistol being fired, or even a click in the record groove.

Especially if the signal is not a naturally recorded one, such as a record scratch, the input to the amplifier may well contain a range of frequencies extending well above the normal audio range. It should be understood therefor that a transient signal is one which is rich in high frequencies and that a fast rising signal is synonymous with a transient.

The undesirable audible effects observed therefore could never be adequately revealed by static tests. Nevertheless, solutions had to be found to the problem, and techniques devised whereby TIM-D could be measured.

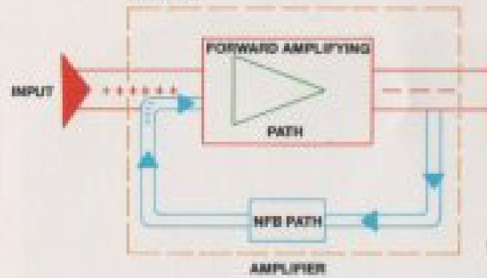
How TIM-D occurs

Frequency response is a way of describing the range over which an amplifier or even just a transistor will uniformly amplify a signal. Thus if an amplifier is said to have a frequency response from 20Hz to 20kHz, it means that for an input of fixed amplitude, the output will always be a fixed multiple, between the given low and high frequency limits. The multiple is called the gain of the amplifier.

Curiously, the transistors used in a hi-fi amplifier often have a frequency response which is smaller than that of the amplifier which they form a part of. The paradox of that statement can be easily resolved when one realises that above the high frequency limit of constant gain, the transistor does not just stop amplifying. In fact the gain slowly drops away to unity at a point of frequency considerably above the limit of constant gain. If a circuit device could be used which sacrifices some of the orginal gain of the transistor, then it would be possible to extend the frequency response. We could in fact continue to extend the frequency response of the amplifier containing the transistor, by sacrificing more and more gain until the point was reached when the gain was unity.

The circuit device used to trade off gain for frequency response is called Negative Feedback (NFB). In applying this, several other advantages accrue. As the amount of NFB is increased, the gain reduces, the frequency response increases and the Total Harmonic Distortion (THD) of the amplifier will reduce. At the same time, the effects of component performance tolerances will also be reduced.

FIG. 1



PRINCIPLES OF NEGATIVE FEED BACK.

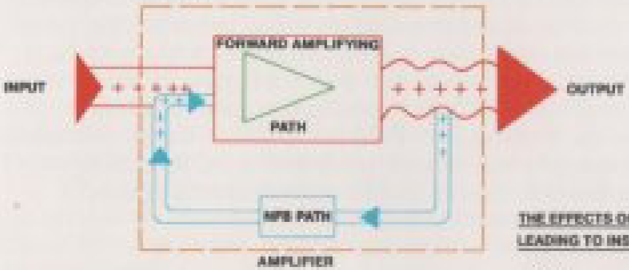
At first sight therefore, NFB seems to be a good thing. If we choose transistors with a very high gain and apply plenty of Negative Feedback, THD can be kept to a very low figure and the frequency response can be extended to accomodate the fastest rising signal we are likely to encounter.

What is not revealed by this simple explanation is that if the gain constancy of the circuit is to be maintained over the desired range of frequencies, this can only be done if the Negative Feedback is constant. In fact Negative Feedback relies on the output of the amplifying circuit always being an exact inversion of the input. If phase shifts occur then this changing relationship weakens the Negative Feedback until it either disappears or it adds to the input to turn the amplifier into an oscillator!

Unfortunately, phase shifts do occur at frequencies above the normal turnover frequency of the transistor and so to retain stability a compensation has to be applied. This takes the form of a simple filter connected between the output of the transistor and the input of the next stage. Although the filter ensures a consistent feedback, it has also been shown to have a detrimental effect contributing to TIM-D.

This is because the filter uses what is called a capacitor. An important feature of capacitance is that it can delay a signal rather as if a water

FIG. 2



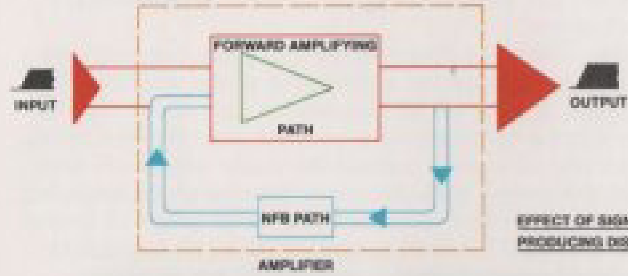
THE EFFECTS OF EXCESSIVE PHASE SHIFT LEADING TO INSTABILITY.

tank were placed in the path of a garden hose. The tank has to be filled before the flow of water can be completed.

In electrical terms, the signal passing through the amplifier is delayed by an amount that increases with frequency. It is certain therefore, that *situations can arise within an amplifier where the high frequency components of a signal will arrive at the output after the low frequencies.*

Negative feedback involves the use of a portion of the output, fed back to the input to control the overall gain of the amplifier. If it fails, because no signal is present at the output at a time when signals are passing into the amplifier, then the gain sacrifice made to increase frequency response will be nullified. As a result, the transistor will suddenly have to produce very large bursts of current which it is not designed for. The more NFB that is applied in the first place, the more disastrous are the results of its failure. Transient Intermodulation Distortion occurs under such conditions.

FIG. 3



EFFECT OF SIGNAL DELAY PRODUCING DISTORTED INPUT TO AMPLIFIER.

Super Integrated Amplifier AU-X1

Solutions

Several rather obvious solutions offered themselves to amplifier designers who have studied this problem. One was to use transistors with a much wider frequency response. This meant that phase shifts due to the transistor would occur at frequencies possibly outside the audio band and so the delay effects produced by compensation would be irrelevant since the input signal would rarely contain frequencies likely to be significantly delayed.

This has resulted in a rash of amplifiers with seemingly ridiculous frequency responses up into the region of several hundreds of kilohertz and capable of slew rates well beyond that needed to reproduce hi-fi signals. Slew rate incidentally, is literally a measure of the rate at which the output voltage or current follows the change in the input signal. It is measured in Volts or Amps per microsecond (V/μSec. or A/μSec.).

Using such exotic transistors is rather expensive and in any case, it can be likened to taking a sledgehammer to crack a nut. In practice, it also causes problems due to the possibility of stray radio frequency signals being amplified and passed on to the speaker to occasionally cause failure of the tweeter.

Another solution which is a little simpler and more elegant, is to limit the range of frequencies injected at the input of the amplifier. This can be done with a filter, cutting off all frequencies say, above 20kHz. However, this alone is not a complete solution since even the best filters, if set to cut off effectively all signals outside the audio range, will also produce an effect on the overall *audio frequency response* of the amplifier.

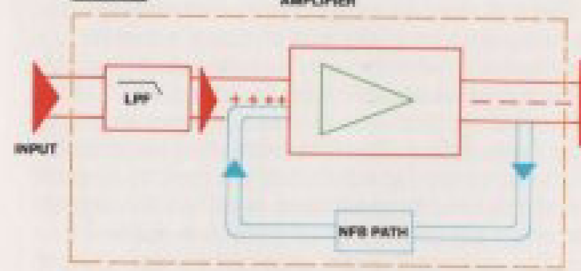
This leads us to the solutions offered in the latest range of Sansui amplifiers including the AU-X1, AU-919, AU-819, AU-719, ingretated amplifiers and the BA-F1 power amplifier and CA-F1 pre-amplifier.

As has been pointed out, the root causes of TIM-D arise from a) an inability of some of the transistors at the input and driver stages of a power amplifier to produce large bursts of current at times when delays cause a partial failure of the NFB and b) too high a delay in the signal path due to the presense of compensating filter capacitance.

Previously attempts to solve problem b) have failed because of the trade-off of stability against delay to high frequency signal components. Usually the main source of TIM-D is the stage that drives the output transistors, although if the input stage lacks adequate current range, TIM-D can sometimes occur there.

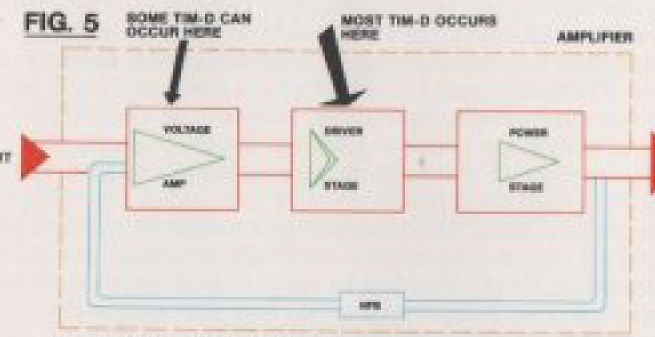
Sansui have avoided TIM-D in the early stages of their power amplifiers simply by making use of the superior characteristics of Field Effect Transistors (FETs). Improving the driver stage proved to be more difficult since the current swing and thus the slew rate demanded is considerably higher than for the input stage. A two-fold solution offered itself.

FIG. 4



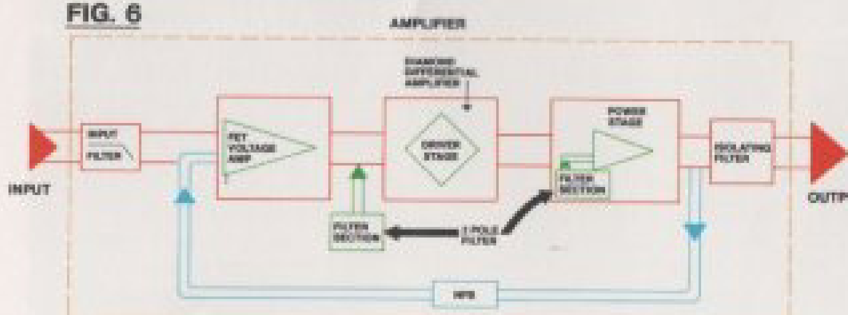
USE OF FILTER TO LIMIT SLEW RATE (FREQUENCY RESPONSE) OF INPUT TO MATCH THAT OF THE FORWARD AMPLIFYING PATH.

FIG. 5



MOST POWER AMPLIFIER HAVE THREE MAIN STAGES. THE INPUT VOLTAGE AMP CAN CAUSE TIM-D, BUT USUALLY THE WORST EFFECTS COME FROM THE DRIVER STAGE.

FIG. 6



SANSUI AMPLIFIER SHOWING ALL MEASURES TO PROTECT AGAINST TIM-D.

First, an entirely new circuit called the Diamond Differential DC (DD/DC) was designed. This has the advantage that it can produce very high slew rates without the expense and difficulty of using powerful and wide bandwidth transistors.

Secondly, to permit a high level of NFB (and thus very low THD), a new form of phase compensating filter was devised. In this design, the filter takes two bites at the problem. Two filter elements are distributed at different points in the amplifier to make what is called a two-pole filter. This has the merit that the circuit capacitance and thus the signal delay is kept small, whilst adequate phase compensation is applied to maintain good stability.

The combination of these features produced an amplifier in which the slew rate was high, the TIM-D was very low and because the feedback could be maintained at a large factor, the THD was also very low. However, this was not all. Since the NFB was high, the frequency response of the amplifier was considerably extended beyond that required.

As a result, filters designed to act well above the audio band have been added to the input of the amplifier. Additionally, an output filter has been fitted to isolate the amplifier from the effects of changes in the loudspeaker load and to eliminate stray radio frequency pick-up.

In Sansui amplifiers, the advantages of NFB have not been thrown away simply to obtain a low TIM-D and instead a complete solution has been provided which permit superb static signal characteristics *and* at the same time eliminating TIM-D. The overall result is an improvement in subjective performance as a result of paying close attention to the dynamic characteristics of the amplifier as well.

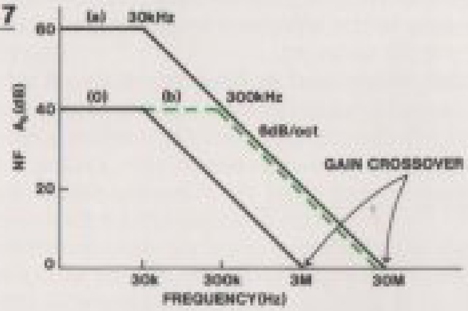
DC Stereo Integrated Amplifier AU-919

Part 2

In this brief section, more technical language is necessary to provide more detail of the circuit techniques used by Sansui. No apologies are made for this however, since it is intended for the more technically minded reader who will be more familiar with the terms and diagrams used. Nevertheless, every attempt has been made to keep the explanation as simple as possible.

Figure 7 illustrates how the application of NFB can extend the linear portion of the gain characteristic of an amplifier. To avoid instability, the gain of the stage has to be set to unity (the base line of the horizontal axis), well before any phase shift causes the outpet to reach 180 degrees shifted from the input.

FIG. 7



Straight DC Stereo Control Preamplifier CA-F1

DC Stereo Power Amplifier BA-F1

Fig.8 shows how a two-pole equaliser may be inserted in the amplifier and Fig. 9 shows the effects of choosing various frequency turnover points for the second pole. As can be seen, where the turnover frequencies of the two filter sections coincide, the overall system response shows only a slight 0.5dB peak at high frequencies.

Using such a filter enables the unity gain crossover frequency to remain the same as for a simple filter, or alternatively to select an increased turnover frequency. In either case both THD and TIM-D will be reduced, even though more NFB is applied than can be tolerated with a single pole simple filter.

The DD/DC circuit is an improvement of the conventional differential amplifier illustrated in Fig.10. The normal type is widely used because it has a high immunity to the effects of temperature variation and fluctuations in the DC supply voltages.

However, it is a weak link when used as the driver stage of a power

FIG. 8

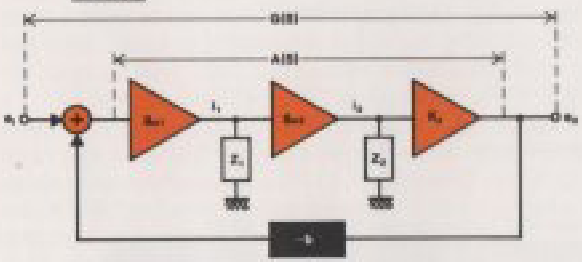
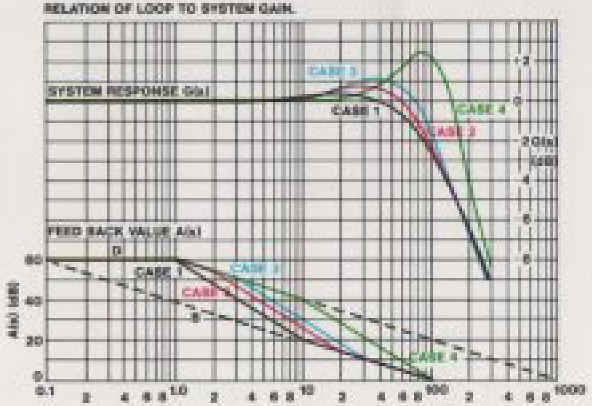


FIG. 9

RELATION OF LOOP TO SYSTEM GAIN.



amplifier, since its correct function depends upon the constancy of current supplied in the tail of the circuit. The output of the differential amplifier is obtained from the outputs of the two transistors and represents the difference in voltage between them. At rest the current from the tail of the circuit is divided equally between the two transistors and thus the voltage difference at the output will be zero. When the inputs to the transistors are subjected to a differential input voltage, the circuit balance changes so that the current divides *unequally* between the two, creating a voltage difference at the output.

The limit of the swing at the output is reached when one transistor is fully shut off and the other is carrying all the current available in the tail circuit. Limiting of current swing under extreme signal conditions is therefore a natural consequence of the constant current source in the circuit tail.

In Fig.11 it can be seen that the original DD/DC circuit has no constant current source and instead a further pair of differentially connected transistors has been substituted. Despite the lack of the constant current source, this circuit shows an even greater immunity to supply voltage variations and very large current swings can be produced on demand without circuit saturation.

With modern power amplifiers, there is an increasing tendency to make use of the MOSFET type of transistor for the output stage. These have the advantage of great linearity and behave somewhat in the same fashion as a valve. Also in common with a valve, they have a very high input impedance. As a result, it had been thought they would not place the same current demands on the driver stage as conventional devices.

However, Sansui have discovered that the large input capacitance of the device is quite capable of 'soaking' up large currents and thus a powerful driver stage *and* the two-pole filter were still necessary. Making use of these transistors, the two-pole filter and the DD/DC driver stage, Sansui have been able to easily obtain an increase in overall amplifier slew rate from 80V/μSec. to 800V/μSec.

FIG. 10

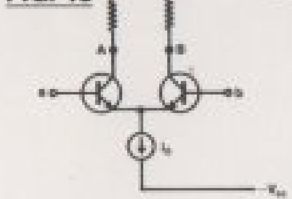
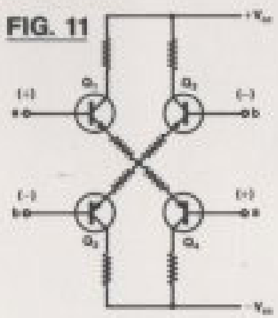


FIG. 11



Conclusions

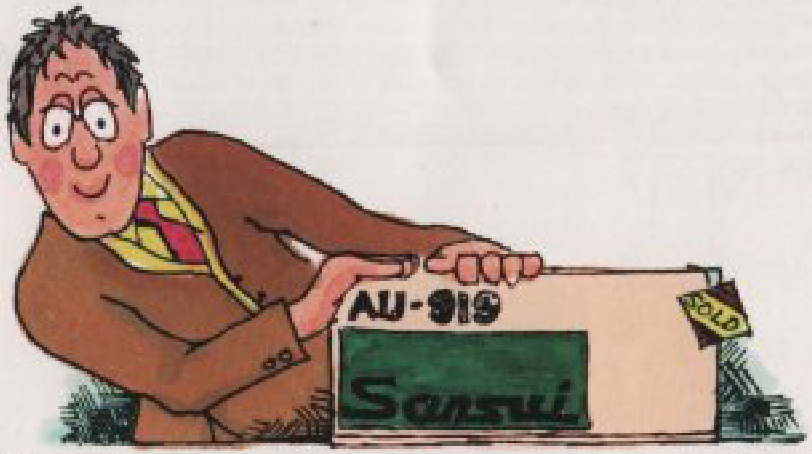
The discoveries and improvements mentioned in this booklet started with subjective experiments showing that instruments did not always tell the full performance story. As a result of this, Sansui developed new and elegant techniques to measure TIM-D and evolved simple circuit techniques to avoid it.

The interdependance of NFB and TIM-D has been shown to be directly linked with the *internal* slew rate of the amplifier. High slew rate is not just obtained from using exotic high frequency transistors, but may be obtained through the application of two-pole phase compensation filters and the use of the DD/DC circuit in the driver stage.

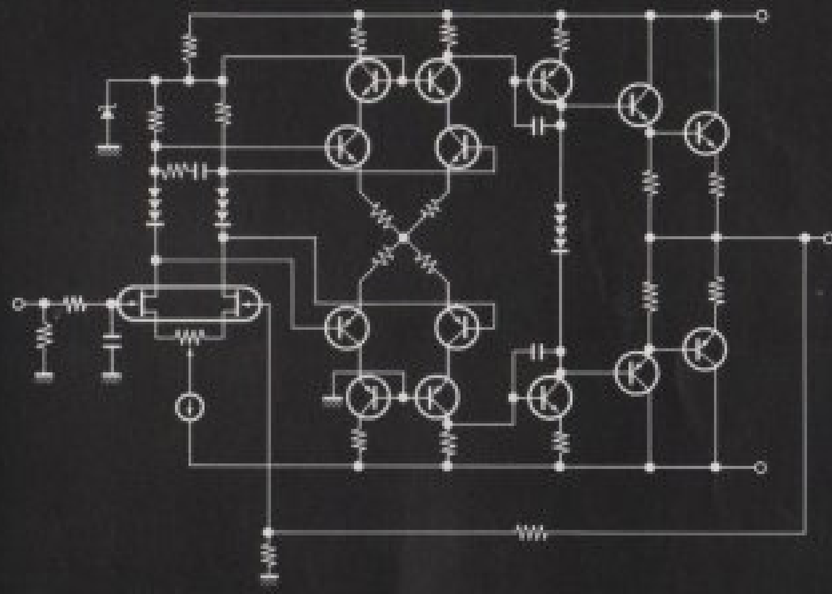
The high internal slew rate is required simply to ensure an accurate NFB under all audio signal conditions and therefore filters may be used at the input and output of the amplifier, set at frequencies well above the audio range. These serve the purpose of preventing undesirable inteference from entering the amplifier whilst keeping the overall slew rate of the amplifier well above the requirements of the fastest audio transient likely to be encountered.

From this, it can be seen that measuring the overall slew rate of an amplifier may not necessarily indicate the true internal slew rate and thus its freedom from TIM-D. Conversely, slew rate alone is not a sufficient criterion on which to judge the true quality of a power amplifier. In the end, it will always be listening tests that determine the ultimate quality of reproduction.

All Sansui's new amplifiers are designed with these criteria in mind giving them not only a superb static signal performance, but also outstanding dynamic performance and negligible Transient Intermodulation Distortion.



THE DD/DC CIRCUIT (POWER AMP SECTION)



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