Compensation method of testing models of audio amplifiers at the stage of their design

The main parameters measured today, such as THD and IMD, do not correlate with sound quality.

Such parameters as THD and IMD show only the additional distortion products (harmonics, intermodulation products) introduced by amplifiers in steady-state mode, but do not reflect dynamic distortion occurring in moments of signal amplitude or frequency changes, nor do they reflect memory distortion.

Abstract

The use of a simulator to study the parameters of various amplifiers with global NFB has allowed to establish a correlation between the input signal delay time (time Propagation Delay) and the available information about the subjectively evaluated sound quality, which indicates the expediency of using this method at the stage of designing electronic amplifiers of high fidelity reproduction for audio signals.

With the help of simulator it is established that amplifiers with the smallest signal time Propagation Delay introduce the smallest additional distortions at the beginning and end of bursts, as well as at the moments of change of frequency or amplitude of signals, which is a continuous process in audio signals.

1. Introduction

The measurement of distortion is fundamental to the design and evaluation of audio circuits. Since the beginning of the development of audio amplifier circuitry, several distortion measurement techniques have been defined and are widely used to improve audio circuits to this day. However, the evaluation of top quality amplifiers by listening tests does not match the figures obtained by these methods and more and more people prefer tube circuits, or circuits without the use of common-mode OOS despite their low distortion figures.

Attempts have been made to identify new, more accurate measurements that better correlate with the subjective tests, but without much success. The explanation for this failure may lie in the fact that these new measurements are based on the classical use of steady-state sinusoidal signals without considering the nature of real audio signals. A real signal looks like anything but a standard 1 kHz signal. A sound signal varies pseudo-randomly in amplitude, frequency, and spectral composition. Questions of theoretical foundations of distortion in signals close to audio signals are fruitful and lead to the opening of new horizons in the design of amplifiers friendly to listeners. This work is a further development of [1].

2. Generalized theoretical analysis

2.1 Traditional theoretical analysis

Figure 1 shows a classical theoretical model of an audio amplifier. This model is the basis for measuring amplifier distortion. It consists of an ideal amplifier and two distortion generators: the linear distortion generator corresponds to the amplitude, phase, phase slope, and group delay resulting from the limitations of the frequency bandwidth of the real amplifier; the nonlinear distortion generator corresponds to the the nonlinear transfer characteristic of the real amplifier.

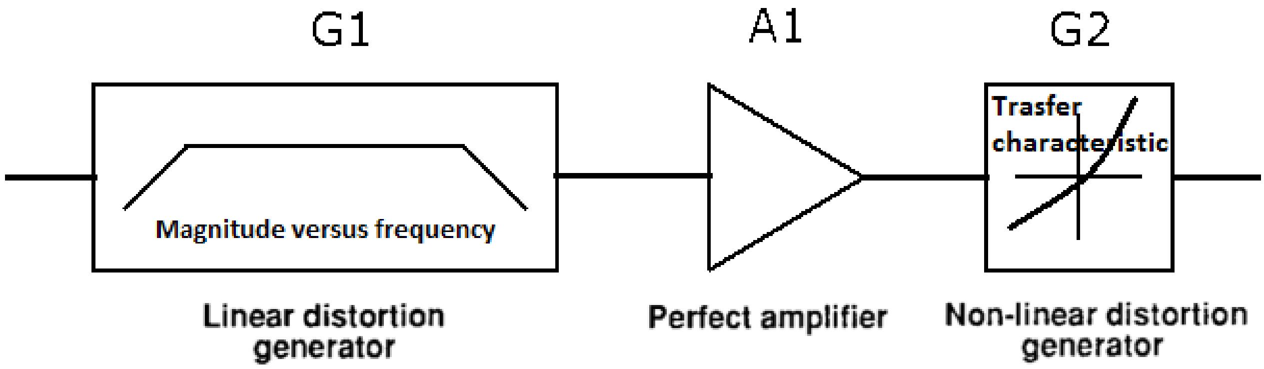

Figure 1. Classical theoretical model of an audio amplifier

The purpose of the current distortion measurement is to characterize the distortion generators. The band-limit and nonlinear transfer function are measured to fully characterize the circuit under test and determine its distortion for any audio signal. Characterization of the distortion generators is performed with sinusoidal signals.

This approach is rigorous and valid as long as the actual model. The validity of the distortion model is widely recognized, even though this model does not take into account known distortion phenomena:
- SID (Slew-rate Indused Distortion) rotational or transient intermodulation distortion,
- memory distortion,
- dynamic distortion due to the complex nature of real signals,
- distortion in the time domain.

The reason for this is that outdated conventional measurement methods using standard measuring instruments and steady-state sinusoidal signals are used. However, the use of more complex test signals in the form of bursts and triangular-shaped signals in distortion measurements using the compensation method shows that linear and nonlinear distortions can be combined in a more complex way than in the classical amplifier model. Thus, their nonlinearity cannot be adequately analyzed with sine waves and therefore with classical distortion measurements.

There was an attempt to evaluate the quality of amplifiers using the SWDT (straight-wire differential test) [2]. However, this test measured vector errors rather than distortion. Although the test condition was to have a signal delay of only 8 ns at the test frequency, the positive result of the test did not guarantee the sound quality of the tested amplifier because it did not take into account the distortion products in the time domain and the behavior of the Group Delay just outside the audio range.

2.2 New theoretical analysis

Careful theoretical analysis of sound circuits reveals many possible causes that make the characteristics unstable and especially variable depending on the signal. These variations often have time constants, causing memory phenomena, and also lead to degradation of important signal components responsible for the naturalness of the sound.

To analyze distortion, a new circuit testing model (a new compensated testing method) can be proposed, including an ideal delay line X1, an additional ideal amplifier A2, and an adder X2 (Fig. 2).

Ideally, in order for the amplifier not to introduce distortion into the transmitted signal, it is necessary that the output voltage curve accurately repeats the input voltage curve on an enlarged scale. In this case, a time shift Δt between the input and output voltages is inevitable, equal to the time the signal passes through the amplification device (time Propagation Delay).

The condition for undistorted signal amplification can be written as:

$$Vout\ (t) = Vin\ Ku\ (t - \Delta t) = Vin\ Ku\ (t - tPD)$$

where

Ku - is the gain at the testing frequency;
tPD – signal propagation delay time at testing frequency [3]

For this, it is necessary that the ratios of the amplitudes and phases of the harmonic components of the output voltage be the same as those of the input voltage. This means that both the changes in amplitude and the time delay of all harmonic components should not depend on frequency, and this is possible only with a constant group delay.

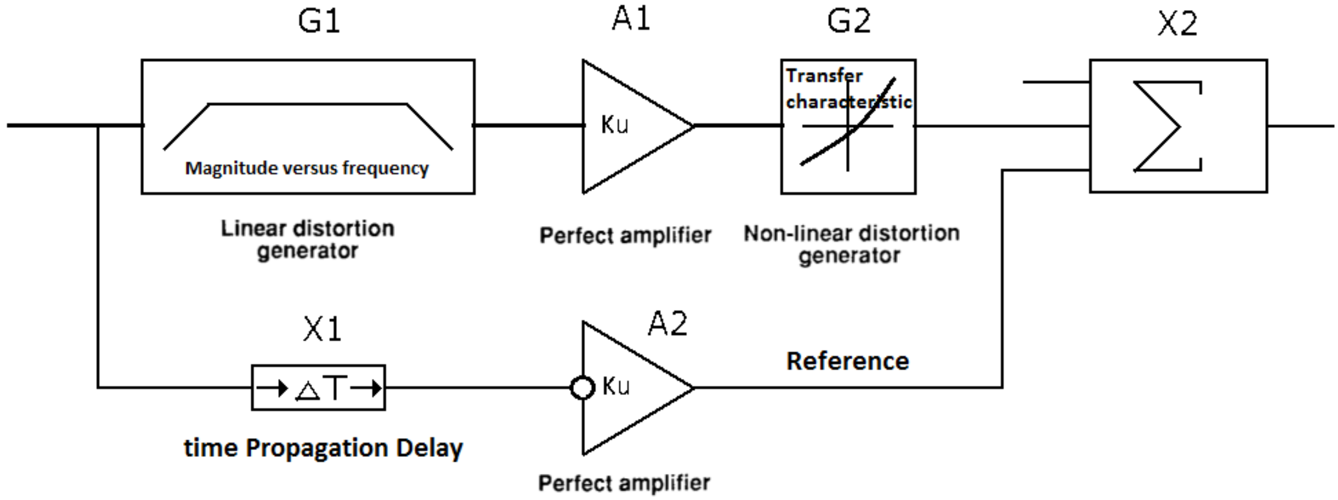

Figure 2. Compensation method of distortion measurement

In compensation measurement, the input signal is the reference and should not be subjected to any processing other than scaling and time shifting.

Any amplifier has a gain (Ku) and a time Propagation Delay (tPD). The input signal multiplied by Ku and time shifted by the time delay tPD is taken as a reference for comparison. The approximate value of the delay is taken from the horizontal section of the Bode plot. The final value is specified in the process of testing by special signals (see item 3). Let's call the input signal multiplied by Ku the reduced to the level of the output signal or simply the reduced signal.

Then by subtracting the reference from the output signal and get the distortion products, shown in Fig. 2. In this case, we immediately get all distortion products in full scale, including memory distortion, which are not detected by any other tests (THD, IMD, etc.). In the simulator, the ideal amplifier A2 can be replaced by a simple multiplication operation, and the adder X2 is an addition or subtraction operation (as necessary).

Non-NFB amplifiers consist of two main assemblies:
A voltage amplifier (VAS) and a current amplifier - output stage (OPS) which most often does not need any inductance at the output.

Both of these nodes have low signal delay times (no more than 50 ns) and are usually quite broadband since they are not covered by correction to ensure stable operation and significantly limit the bandwidth.

Typical signal transit delay times in amplifiers with NFB are in the range of 0.2... 5,5 µs and only in rare samples fall below 0.1 µs.

The following types of distortion should be distinguished:
- nonlinear distortions in the form of harmonics (additional harmonics in the spectrum);
- intermodulation products in the case of two or multitone signals;
- linear distortions: changes in signal amplitude and phase without adding harmonic components (occurring in steady-state mode);
- transient distortions - distortions associated with the time constants of the input circuits and NFB circuits. During transients, harmonic signals become quasi-harmonic and are enriched with additional harmonic components [4];
- temporary distortions of a resonant nature associated with the presence of inductance at the amplifier output and the reactive nature of the load. This type of distortion can also produce an unwanted additional

level of harmonic content;
- memory distortions (distortions related to thermal processes in semiconductors and electrical time constants);
- dynamic intermodulation distortion of the type SID (slew-rate induced distortion), arising due to insufficient rotation speed at the moments of signal deviation from the sinusoid: changes in frequency, changes in amplitude, or both at the same time, which occurs continuously in audio signals.

Conventional distortion measurement methods only detect the first two types of distortion, which correlate poorly with sound quality.

The last type of distortion depends on the signal transit time in the amplifier. This type of distortion is the most unfavorable, as it leads to sound degradation, to loss of microdynamics - it "smears" fine details of the sound material like an mp3 compressor.

Amplifiers with common NFB are prone to all types of distortion, while amplifiers without NFB are least prone to the last type of distortion.

3. A new set of test signals.

Compensation method of distortion measurement is not new, one of the first mentions of this method is described in [5]. The following references can be found in [6] and [7].

The disadvantages of these methods include the fact that RC-chains were used as a delay line for the input signal. Such delay lines introduce transient and linear distortions into the test signals as standards, which is unacceptable.

To detect all kinds of distortions it is necessary to use an ideal delay line that is possible in simulators in the process of debugging circuits, as well as to use instead of stationary sinusoidal signals more complex test signals.

When testing real amplifiers, a coaxial cable with a delay per linear meter of about 5 ns can be used as a delay line. Thus, you may need a cable coil of 10 meters or more. Adjustment of the delay line can be done using a resistor of 10 Ohms or less and a trimming capacitor. For example, a 10 Ohm resistor and a 1 nF capacitor will give a delay of 10 ns.

Such signals can include the following signals:
- 10 kHz bursts (2-3 periods) with alternating polarity of the first half-period;
- 10 kHz bursts of different amplitudes without phase discontinuity;
- 10 and 20 kHz bursts with no phase break, including those of different amplitude;
- 10 kHz triangular signals.

Before such signals are fed to the input of the tested amplifier model, they are subjected to first-order LF processing with a cutoff frequency of 100 kHz (as in the DIM-100 test).

It makes sense to insert a 50 μs delay line in front of the generator signal, and to leave space in the calculation to observe possible transients at the end of the second burst.

The use of short bursts (two periods are enough) following each other with an interval of 1...3 periods allows to identify in pauses both transient distortions and memory distortions, which can be dependent on the polarity of the first half-period in the bursts.

As a reference of distortion products of the first two test signals we can use the distortion products extracted on a sinusoidal signal in steady-state mode with the help of a 10 kHz notch filter, fig. 3

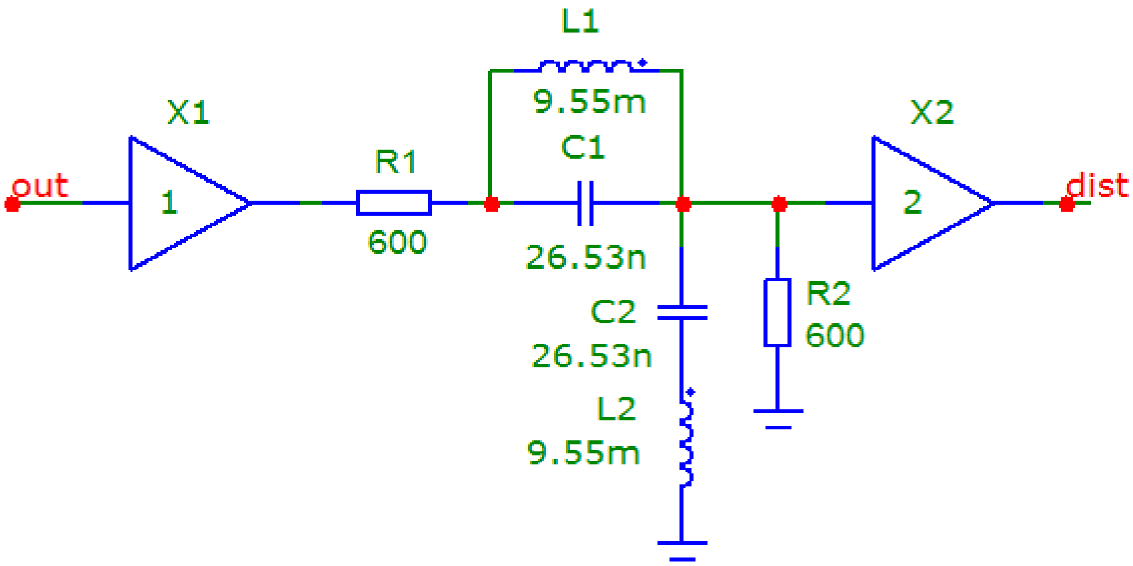

Figure 3. Notch filter for 10 kHz frequency

The Bode diagram of such a filter is shown in Fig. 4

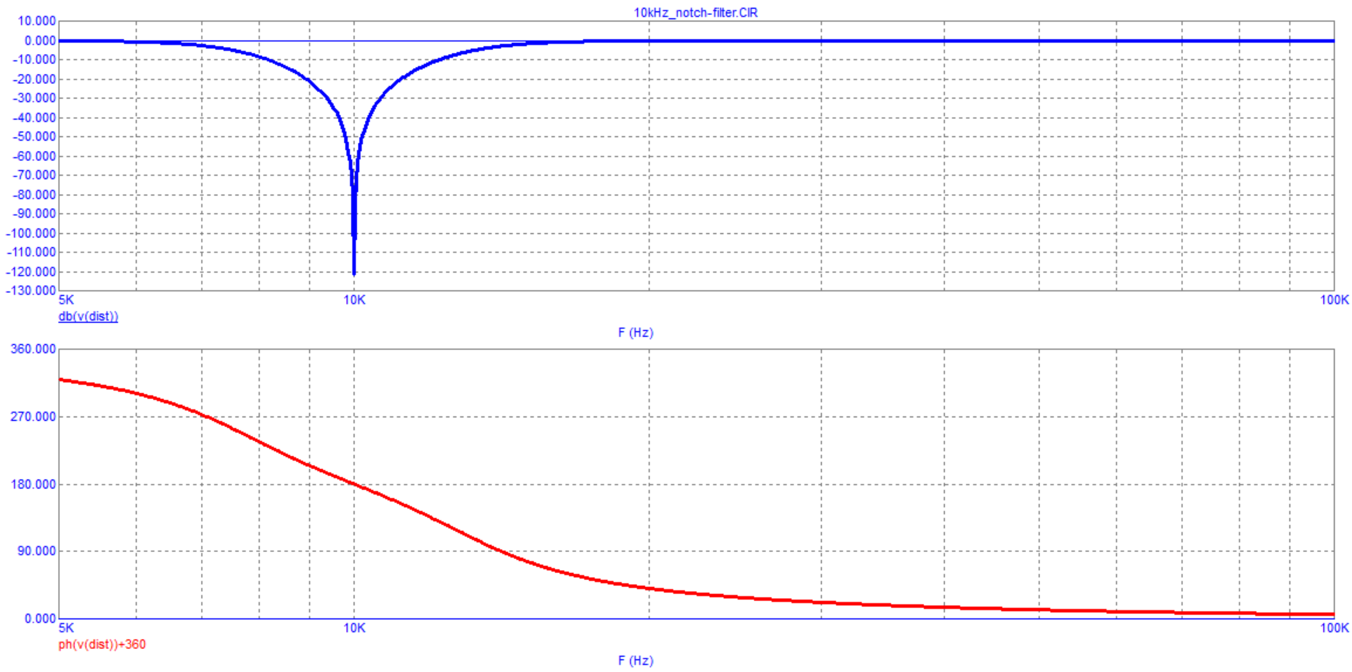

Figure 4. Bode diagram of a notch filter.

The Bode diagram shows that starting from the 2nd harmonic of the test signal the filter gain is equal to 1, and the maximum phase deviation is less than 45 degrees for the 2nd harmonic of the signal.

When using such a filter, for example, in the Micro Cap program run the Transient test for 2 ms and watch the distortion products in the interval of 1.7 ....2 ms (i.e. at the end of transients in the filter).

It is safe to say that from the second period of the 10 kHz signal, the distortion spectrum of the vast majority of amplifiers (especially DC amplifiers, which do not have transient distortion due to the presence of capacitors at the input, output, and in the circuit NFB) differs little from the steady-state spectrum. Therefore, when conducting a test by the compensatory method, the task of the engineer using this method is to obtain the maximum coincidence of distortion products of the second period of bursts and subsequent with the distortion products at the output of the notch-filter both in amplitude and in shape. This is achieved by refining both the gain and the ideal delay line.

4. Correlation with listening tests

In [1] it was proved that amplifiers with no memory distortion (in particular tube amplifiers) compare favorably in sound quality with amplifiers in which such distortion is present.

An analysis of information was carried out on amplifiers without NFB from such companies as Densen, Threshold, Pioneer, Nakamichi, Akai, as well as on amplifiers without using the global NFB from such developers as Charles Hansen, Nelson Pass, Jeff Rowland, Mike Malinowksi, Vladimir Lamm, etc. .

Despite the relatively high level of distortion compared to deep-NFB amplifiers, it is these amplifiers that provide the best sound quality. And it is not surprising, because the signal delay in such amplifiers is negligible and the harmonic spectrum is short and falling.

At the same time a lot of work has been done with models of amplifiers with NFB. The signal delay time in such amplifiers varies within 0.2...3.5 µs. Moreover, such amplifiers have a low frequency of the first pole, their output impedance is not constant in the audio bandwidth and has a phase shift in accordance with the phase of the loop gain, which also does not contribute to sound quality. The result was repeated with inevitable stability: the lower the signal delay - the faster the NFB, the wider the small-signal band, the better the microdynamics, the better it copes with switching and other types of distortion, the less models of such amplifiers are subject to dynamic intermodulation distortions of SID type in moments of change of both frequency and amplitude of the input signal, and the lower the noise floor in the IMD measurement and hence the higher the correlation with sound quality.

In [8], the causes of SID were analyzed and it was concluded that one of the reasons for their occurrence is excessive depth of NFB and its insufficient performance. The optimum depth of the automatic feedback at sufficient speed (Group Delay not more than 100...120 ns) was recommended to be 30...40 dB.

5. Conclusion.

A compensatory method for measuring all types of distortion has been presented. The method has been realized and tested on a large number of audio amplifier models and has shown high efficiency.

Nowadays, when no audio amplifier design can do without preliminary modeling in a simulator and optimization of such basic parameters as phase margin and gain on a reactive load, it is reasonable to measure Group Delay and analyze its behavior far beyond the audio range. The decay of the Group Delay should be smooth, without significant emissions and should not be negative.

When designing amplifiers with NFB, group delay greater than 100 ns should be avoided in the frequency range from 5 kHz to 300 kHz. A group delay output higher than 100...150 ns in the frequency range up to 300 kHz is undesirable. Let's assume a slight increase in the group delay at low Q in the region of 1 MHz and above, followed by a smooth decrease. A high-Q rise in group delay may indicate that the amplifier is prone to parasitic oscillation. The most preferred amplifiers should be DC amplifiers (some amplifiers will list "DC" in their specifications).

If the amplifier is designed competently and its Group Delay horizontal line is at least from 5 kHz and up to several hundred kHz, the equalization of the reference and output signal is automatic up to 15...20 harmonics of the audio range. In THD at best 10 harmonics are taken into account, and often even up to 80 kHz (i.e. up to 4 harmonics for 20 kHz).

As for the permissible THD level, psychoacoustics gives the following answer: "The thresholds of auditory sensitivity depend significantly on the nature of nonlinearity: when lower (second, third) harmonics appear, the hearing thresholds for tonal (single, sinusoidal) signals are 0.1%, for piano music 1-2%, for pop music up to 7%." [9].

The application of this method at the development stage will significantly improve the level of developments.

Literature:

1. Gérard PERROT, Measurement of a Neglected Circuit Characteristic, AES-100, may 1996
2. Nulling Out Amp Distortion by David Hafler (Feb. 1987), https://gammaelectronics.xyz/audio_02-1987_null.html
3. С.Н.Кризе, Усилительные устройства, Связьиздат 1958
4. Л.М.Финк,   Сигналы, помехи, ошибки... , М, Радио и связь, 1984

5. М.А. Сапожков, Корреляционный метод измерения нелинейных искажений передачи, Акустический журнал, Том 2, вып. 3, 1956
6. Jan Lohstroh and Matti Otala, An Audio Power Amplifier for Ultimate Quality Requirements, IEEE TRANSACTIONS ON AUDIO AND ELECTROACOUSTICS, VOL. AU-2 1, NO. 6, DECEMBER 1973
7. P.J. Baxandall, Audible amplifier distortion is not a mystery, Wireless World, november 1977
8. An Overview Of SID and TIM, Walter G. lung, Mark L. Stephens, and Craig C. Todd
9. I. Aldoshina, Hearing thresholds (part 2), Sound engineer 2000-06

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