

## Measuring Distortion in Amplifiers

I have often been asked to define "velocity distortion". I hope this material will help to understand what I mean by this concept.

**A. Petrov**

During operation, all amplifiers introduce various kinds of distortion. There are linear distortions (changes in the amplitude of the signals and their phases without adding harmonic components, which usually takes place at the end of transients, i.e., in steady state) and non-linear distortions caused by the imperfection of both active and passive elements. And although linear distortions (deterministic distortions due to bandwidth limitation and uneven amplitude-frequency response) do not depend on the signal level and do not add new spectral components, they change the amplitude and phase relationships (due to the unequal speed of propagation of the spectral components) between the individual components of the signal and thereby distort the temporal structure, i.e.

The coefficient of non-linear distortion was first measured by K.Kupfmuller [1] and called them the clear factor (bounce factor). By the beginning of the 1950s, more than 20 different methods for measuring distortions had already been proposed.

According to the properties of the test signal and as their correlation with subjective quality assessments (SOC) grows, V. Rakovsky divided all these methods into five main groups [2]: );

- 2) methods of two tones (intermodulation distortion - IMI); H)
- methods with a discrete spectrum (multitone);
- 4) continuous spectrum methods (white noise, pink noise, pseudo noise);
- 5) methods with a working signal (Sapozhkov's compensation method).

Looking ahead, let's add another group here:

- 6) methods with a working Hafler-Carver signal.

Each of these groups has varieties that differ in the methods of recording nonlinearity, isolating distortion products, etc.

Matti Ojala devoted a lot of time to testing methods, his research formed the basis of the DIM-30 and DIM-100 standards.

Developers such as John Curl pay special attention to the frequency of the first pole and higher harmonics from the 7th. Since the OOS suppresses mainly the lower harmonics (2nd and 3rd) and almost does not affect the higher ones, John Curl tries to get by with a minimum OOS depth sufficient to bring the traditionally measured parameters to the minimum required by standards for Hi-Fi category amplifiers.

Lynn Olson in [3] writes:

"In electronics, the correlation between distortion percentage and subjective sound quality is almost zero. The lower order harmonics are almost inaudible compared to the higher ones, although they dominate the THD measurement figures! The meter arrow tells the developer complete nonsense.

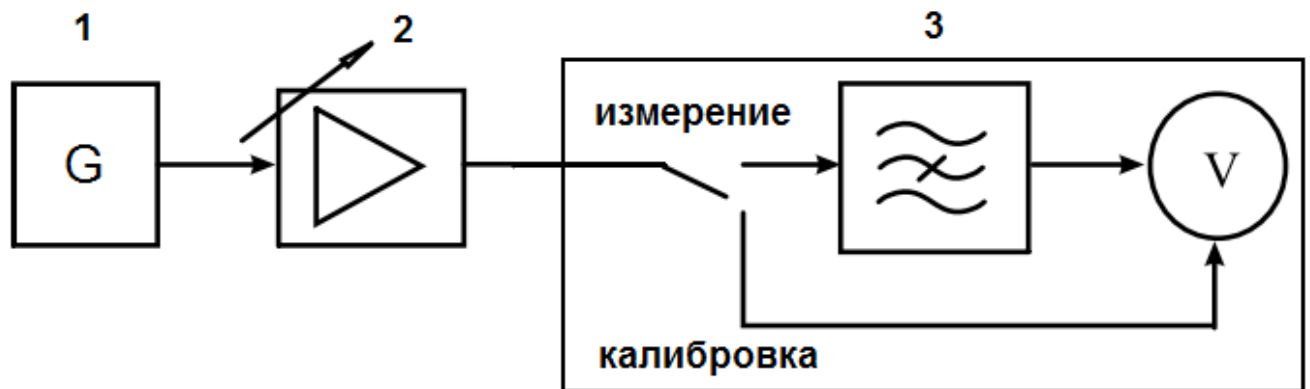
It's time to take a break from the myth of "euphonious distortion" and find those real subtle sources that create distortion in the amplifier that the human ear detects. Once we find measurement methods that actually help, rather than hide the truth from us, it will become easier to develop listener-friendly electronics."

Unfortunately, to this day, the first method of testing is the main one. It's like in a well-known joke - at night we are looking for the lost car keys under the lamp, and not where we lost it - because it is lighter there.

With the use of a single tone, bridge-based meters were used to suppress the main signal and highlight all harmonics at once, as well as methods of separate

measuring the voltage of each harmonic using spectrum analyzers.

However, non-linear distortion meters (NDI) based on notch filters are most widely used. The block diagram of such a meter is shown in fig. one

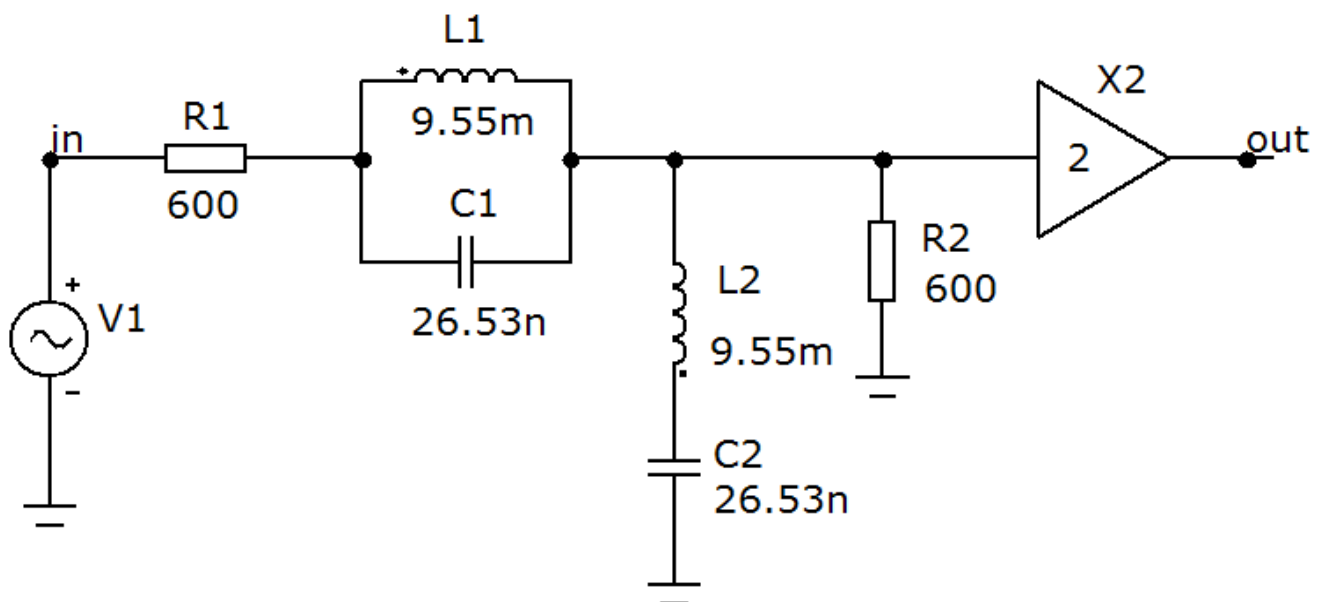


**1 - генератор, 2 - тестируемый усилитель, 3 - ИНИ**

Rice. 1 Block diagram of the harmonic distortion meter

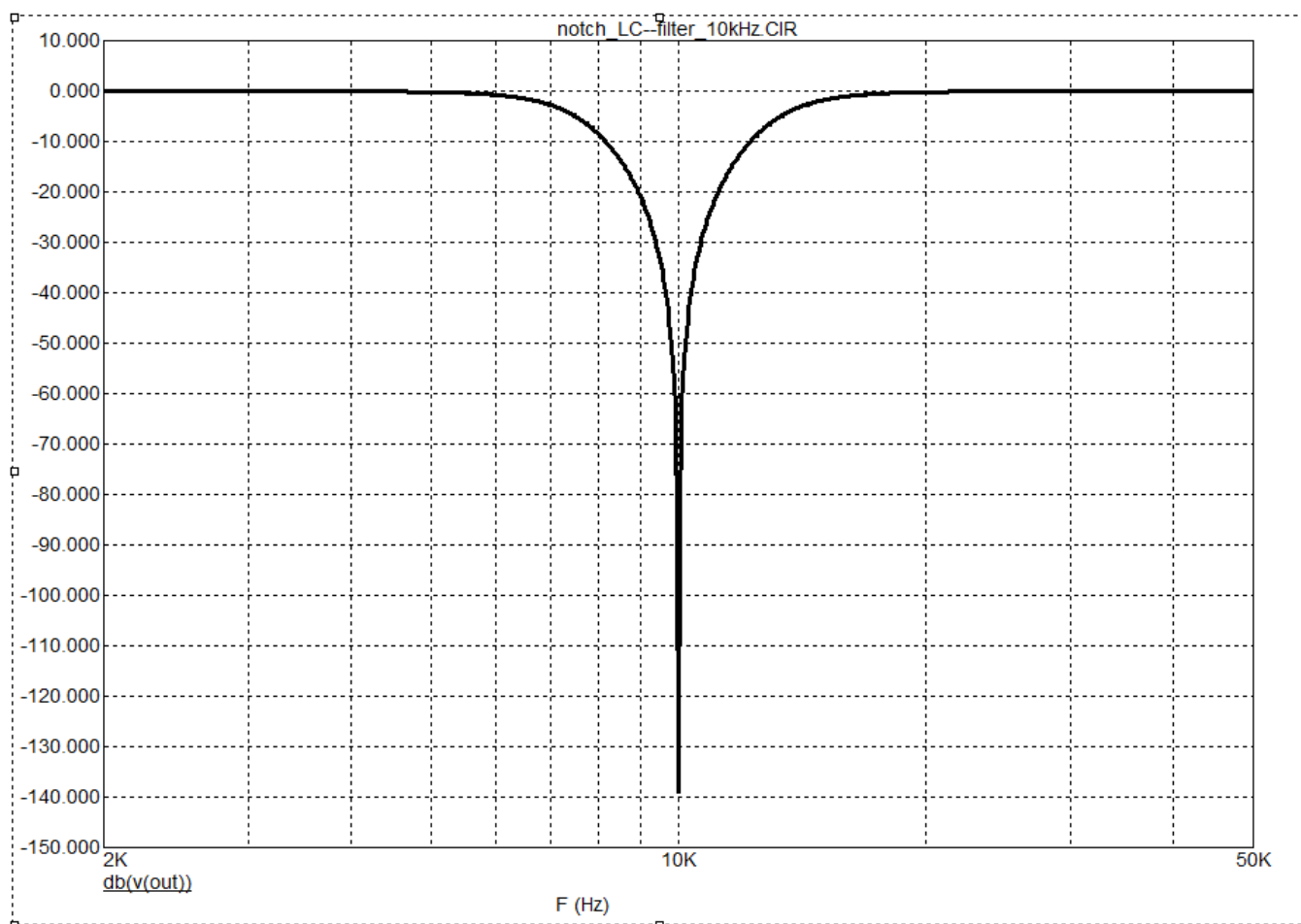
Let's try to figure out what is the reason for the low rating of this kind of meters, that they are treated as "probes" to control the compliance of amplifiers with their specifications during the production process (in the technical process).

There are a large number of notch filters, both passive and active, based on op amps. In order to evaluate the contribution of distortions to the useful signal at a level below 0.001% (-100 dB), the fundamental harmonic suppression level must be 10...20 dB deeper, i.e., not less than 110...120 dB. To begin with, consider the simplest passive filter of the 4th order at a frequency of 10 kHz based on LC elements, Fig. 2



Rice. 2 4th order Bessel notch filter

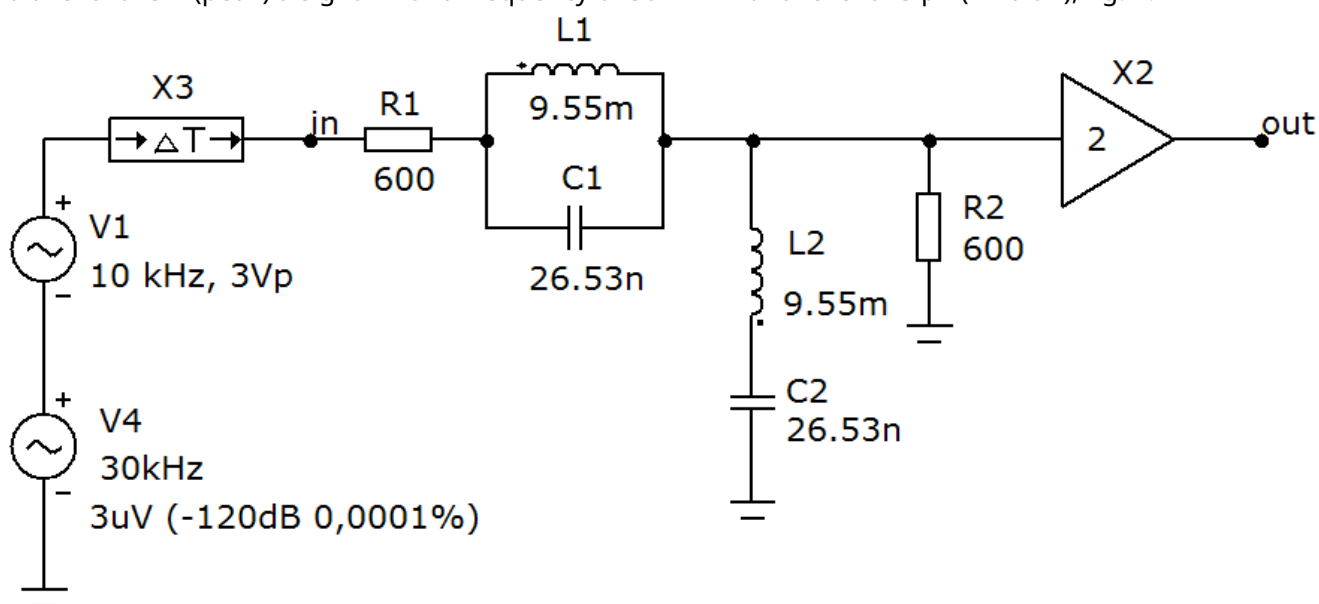
The frequency response of such a filter is shown in Figure 3



Rice. 3

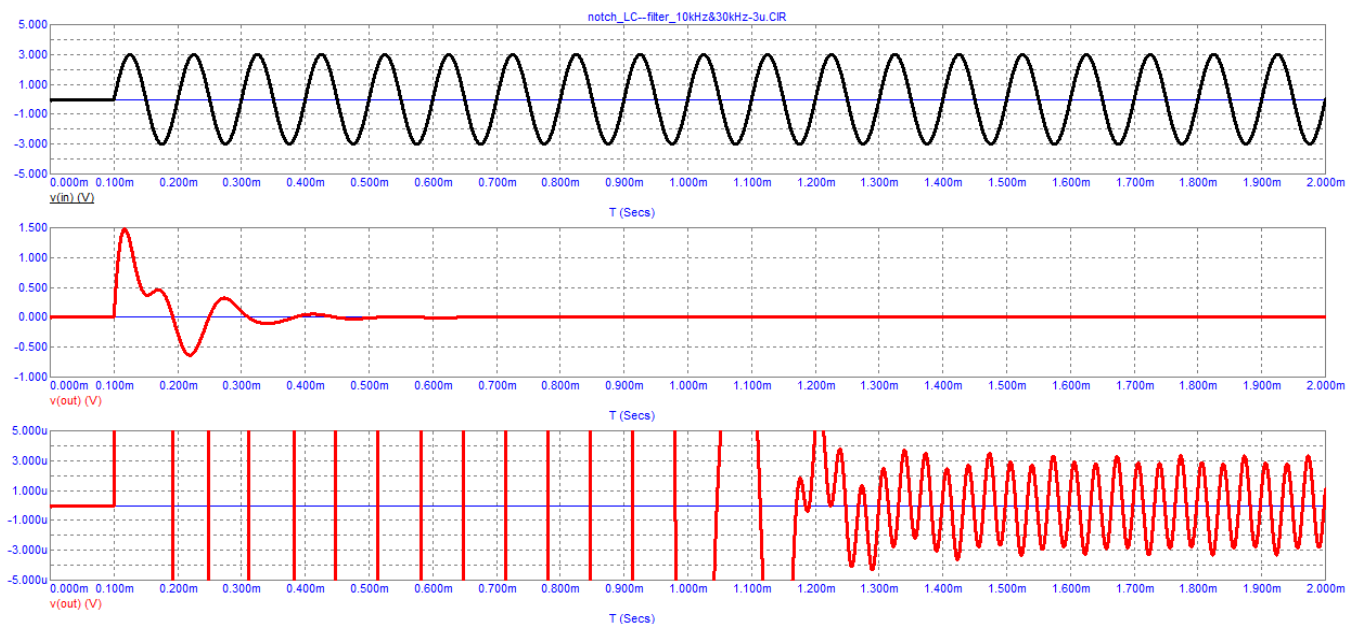
As can be seen from the frequency response graph, the suppression of the fundamental harmonic is almost 140 dB, and the nearest 2nd harmonic and subsequent ones are transmitted without attenuation.

The next parameter that interests us is the settling time, i.e. time after which the filter can adequately measure distortion. To do this, we add to the fundamental tone with a frequency of 10 kHz and a level of 3 V (peak) a signal with a frequency of 30 kHz with a level of 3  $\mu$ V (-120 dB), fig. 4.



Rice. 4 Scheme for measuring the third harmonic of a signal with a frequency of 10 kHz

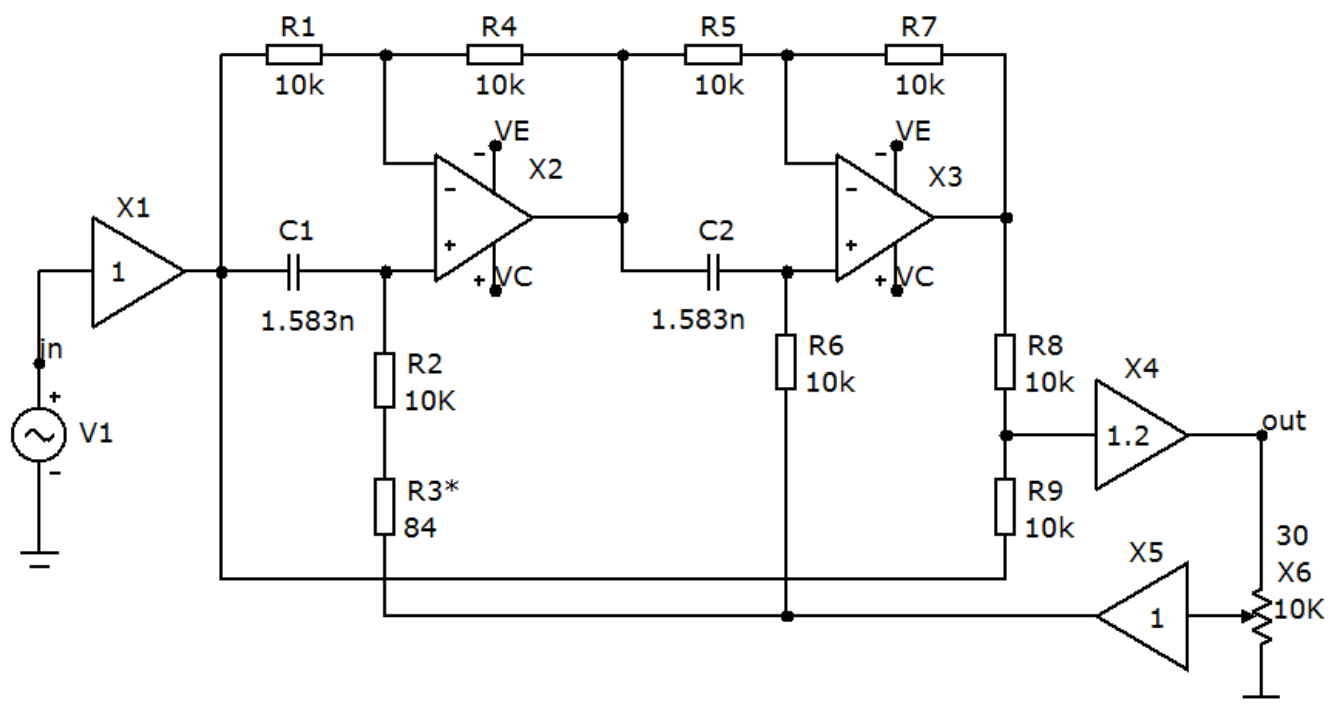
We start Transient/Analysis and look at the result, fig. 5



Rice. 5 3rd harmonic measurement result using the simulator

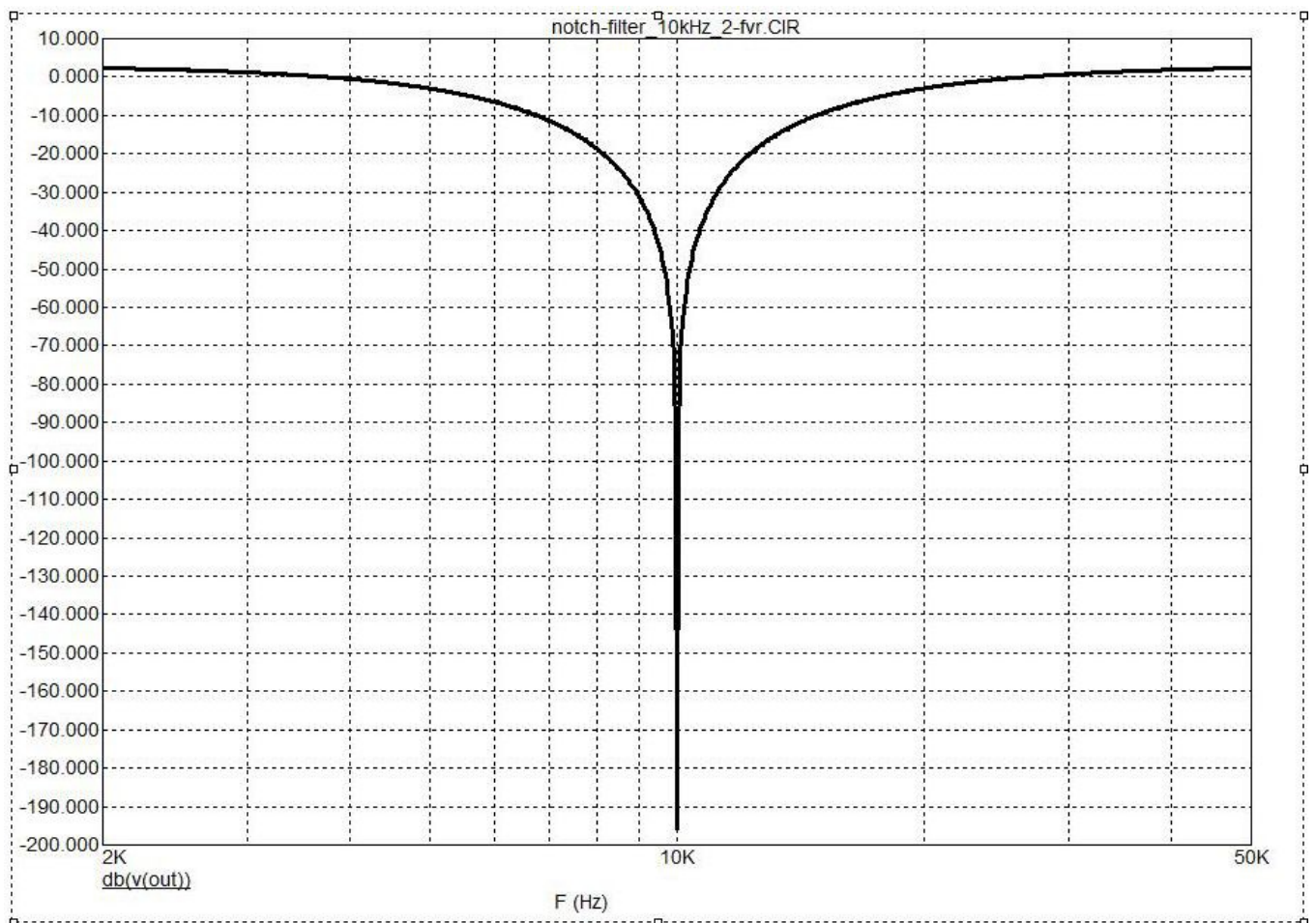
The test result shows that the transient distortion in the filter itself lasts about 1.5 ms, i.e. about 15 times longer than the period of the measured signal. Thus, it is not possible to measure distortion in the first periods (as it is possible to do with the help of the microcap program of the 9th version), and it is precisely in the first periods (the largest contribution in the first) that the main distortions occur in amplifiers due to transients, which erroneously referred to as linear distortion. Let me remind you that linear distortion occurs at the end of transients, i.e., in steady state.

Of the active notch filters, there are options based on a double T-bridge, a Wien-Robinson bridge, etc. But the suppression depth of such filters barely reaches 40 ... 50 dB, which is clearly not enough. Consider a notch filter based on phase shifters, [4], fig. 6



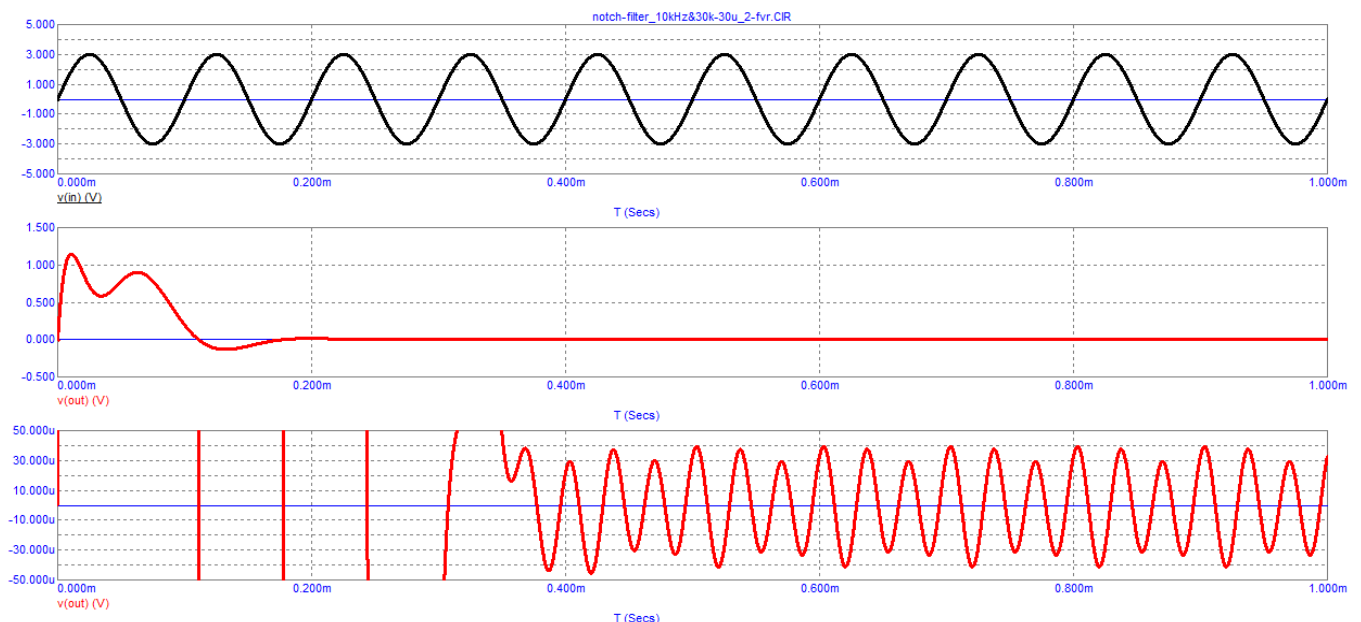
Rice. 6 Notch filter circuit on phase shifters

Such a filter presses the main tone with a frequency of 10 kHz more than 95 dB. To increase the degree of suppression, we turn on two such filters in series and measure the frequency response, fig. 7



Rice. 7 frequency response of 2 notch filters connected in series on phase shifters

Let's add a signal with a frequency of 30 kHz with a level of 30  $\mu\text{V}$  (-100 dB) to the test signal, and check the settling time, fig. eight



Rice. 8 3rd harmonic measurement result

The test shows that the settling time of such a filter is almost 3 times shorter. However, despite almost 50dB more rejection than the previous filter, the filter is unable to adequately measure the 0.001% (-100dB) distortion due to active element (op-amp) distortion. Models of high-speed precision op amps of the ORA627 type were used as the op amp. Thus, even with this filter, at least 4 pitch periods must be skipped. By the way, in the latest versions of the microcap (11th and 12th)

it is impossible to set less than 4 periods for miscalculating the amplifier for distortion (harmonic spectrum, total Kg).

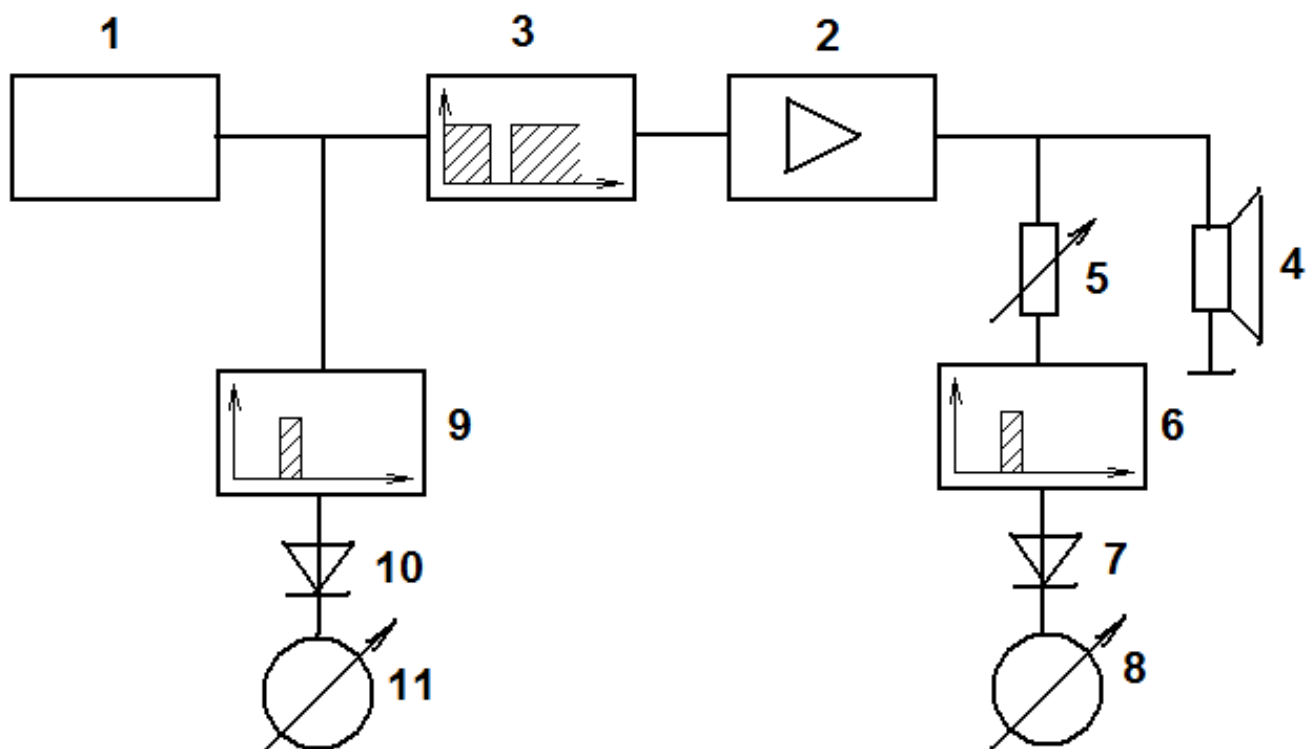
The sinusoidal voltage is generated by a constant amplitude vector rotating at a constant angular velocity.

This explains why the test results of this method do not correlate with the sound quality. After all, the main distortions occur at the moments of change in  $dV / dt$  (at the moments of change in the voltage vector or the angular velocity of its rotation), which takes place continuously in real sound signals.

The situation is somewhat better with two-tone and multi-tone methods, however, due to the fact that there are also simple fixed signals in the spectrum, the correlation with sound quality that was expected does not occur.

The spectrum of a signal is the result of decomposing a signal into simpler ones in the basis of orthogonal functions. As a decomposition, the Fourier transform, expansion in Walsh functions, wavelet transform, etc. are usually used. In simulators, the Fourier transform is mainly used.

At the beginning of 1950, W. Wolf applied for a patent "Method for measuring non-linear distortions" [5], fig. 9.



Rice. 9 Block diagram of the NI meter according to patent SU90185

The essence of the proposed measurement method is illustrated by the diagram. From a continuous spectrum sound generator of equal amplitudes *one* voltage is supplied to the input of the investigated path 2 across notch filter 3, blocking a narrow band of frequencies from  $f_1$  to  $f_2$ . From the exit of the tract 2 voltage is applied to the load 4 and in parallel through a voltage regulator 5 to strip filter 6 with a bandwidth of  $f_1$  to  $f_2$ . From the filter 6 voltage is applied to the detector 7 and then to the galvanometer *eight*.

Because the filter bandwidth 6 equal to filter stopband 3 (or somewhat worse than it), then at the output of the studied path, in the case of its linearity, there should not be currents whose frequencies lie in the filter passband 6, and a galvanometer *eight* should not deviate. The deviation of the galvanometer needle will indicate the appearance at the output of the studied path 2 currents in the frequency band  $f_1 - f_2$ , i.e. frequency currents, which, due to the presence of a filter 3, were absent at the input of the tract.

These generation of currents in the frequency band  $f_1 - f_2$  are the result of the appearance of harmonics from the lower frequency components contained in the input signal, and the result of the formation of combination frequencies falling on the band  $f_1 - f_2$ .

Deviation of the galvanometer pointer *eight* proportional not only to the degree of nonlinearity of the investigated path, but also the amplitude of the voltage acting at its input. To separately account for both of these factors, one more band-pass filter is introduced into the circuit *9*, detector *10* and galvanometer *eleven* with the same characteristics as similar elements *6*, *7*, *8*.

With voltage regulator *5* into the filter circuit *6* attenuation is introduced equal to path gain *2*. In this case, the deviation of the galvanometer *eleven* will be proportional emf useful signal in the frequency band  $f_1 - f_2$ , and the galvanometer *eight* - proportionally emf distortions, newly formed by the tract itself *2* in the same frequency band. Attitude galvanometer readings *eight* to the readings of the galvanometer *eleven*, with an appropriately selected attenuation of the voltage regulator *5*, will be a measure of non-linear distortions created by the investigated path.

For the convenience of reading galvanometers *eight* and *eleven* can be replaced by a ratiometer, deflection angle whose arrow is proportional to the ratio of the currents flowing through its coils.

Since sound energy is unevenly distributed over the frequency spectrum, it turns out that the distortions measured in the frequency band  $f_1 - f_2$  are not typical for other parts of the spectrum. Therefore filters *3*, *6* and *9* should be made tunable over the range of sound frequencies while maintaining the same width of their bandwidth (and, accordingly, delay). With a sufficiently fast tuning of the mentioned filters, the device will give an idea of the average energy distortion value for a time equal to the filter tuning period. In this case, it makes sense detectors *7* and *10* execute according to plan pulse meters with an appropriately selected time constant.

The main reason for filing an application is that the tests performed by applying a pure tone (or several pure tones) to the input of a quadripole do not take into account the spectral composition of a real sound signal. Therefore, the magnitude of those non-linear distortions that arise in the equipment when it is modulated by a voltage of a complex spectrum, characteristic of a real sound program, is far from fully reflected. The possibility of the occurrence of combination frequency currents, which sharply increase nonlinear distortions as a result of the amplitude-phase conversion of signals, is also not taken into account.

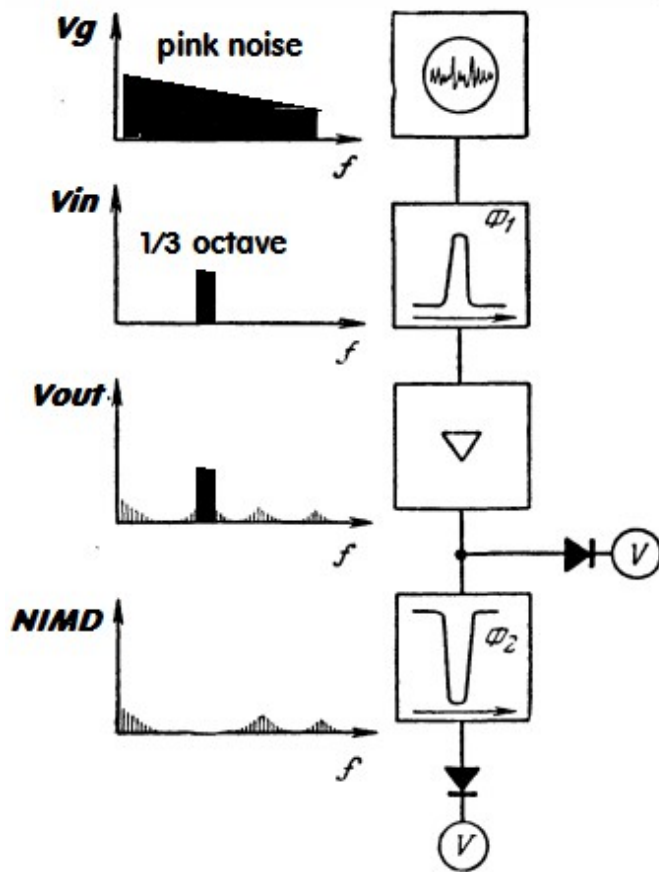
The most interesting thing is that according to this method, at the AES-126 held in 2009 in Munich, a well-known specialist in the field of electroacoustics prof. A. Farina presented the report and presented it as a new testing method [6].

However, the Wolf method turned out to have a significant drawback - despite the possibility of tuning filters over the sound range, it is impossible to determine exactly in which frequency band the amplifier has the greatest non-linearity.

This drawback was eliminated by a meter acting exactly the opposite [7]: 1/3 octave noise (white noise, pink noise or pseudo noise) is fed to the amplifier, and it is cut out at the output. As a result, distortion products generated by the noise signal remain in the spectrum, Fig. 10. And since the noise signal is modulated both in frequency and amplitude, it just provokes high-speed distortions characteristic of transients in the first periods of tonal signals.

Assuming that the high-frequency part of the audio range is most critical to the non-linear distortion of amplifiers, we can limit ourselves to several 1/3 octave bands starting from a frequency of 10 kHz. As test signals, you can use ready-made signals available on the test disks to check the frequency response (an example of the use of such signals is given in the appendix).

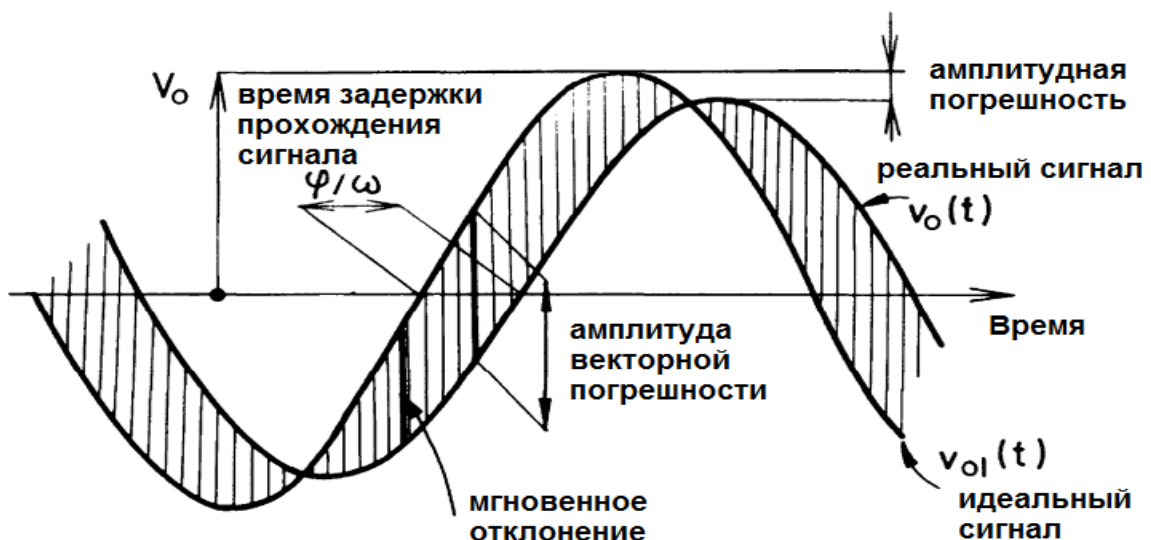




Rice. 10 Principle of operation of the INI using narrowband noise

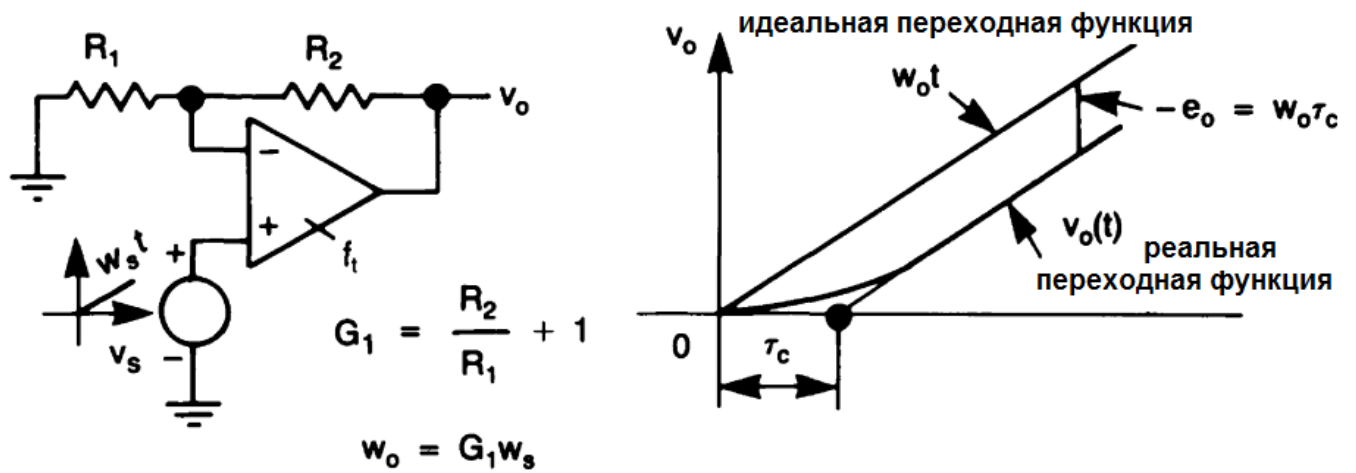
More details about the results of comparative tests can be found in [8]. The next step in the development of measuring technology was the direct subtraction of signals after scaling and phase matching of the input signal to the output phase, [9]. This method (the method of cross-correlation between input and output signals) was defended by Doctor of Technical Sciences M. Sapozhkov in 1954. The same method was also published in [10]. This method was later published in [11] and became known as the Baxandall method. In Russia, the popularizer of this method was I. Akulinichev, who published a series of vector distortion indicators in the Radio magazine.

12. All types of distortions are most fully reflected in [12], and are illustrated in Fig. 11 and



Rice. 11 Illustration of the relationship between vector, amplitude and phase errors when representing a signal as a function of time.





Rice. 12 The speed error of a non-inverting amplifier when excited by its linearly increasing voltage.

The types of distortions are deciphered on page 322 [12] in paragraphs 13 and 14, we will give them in full with minor additions

13. Speed error of first-order resistive operating circuit with corner frequency  $f_c = 1/2\pi\tau_c$ , excited by a linearly changing signal - such that the output signal changes at a rate  $\omega_{exit}$ , is equal to:

$$\epsilon_{exit} = -\omega_{exit}\tau_c \quad (one)$$

where

$$\omega_{exit} = 2\pi f_{\omega} U_{exit} \quad (2) \text{ (see p.14)}$$

$\tau_c = RC$  and is physically equal to the signal delay (time Propagation Delay  $t_{PD}$ ) in the passband

14. How speed error  $\epsilon_{exit}$  for non-harmonic, and the vector error  $\epsilon_v$  at harmonic effects express the same basic dynamic constraint associated with the presence of a corner frequency of the operating circuit  $f_c$ . In a first-order resistive operating circuit, these two errors are related by:

$$-\epsilon_{exit}/U_{exit} = \epsilon_v(f_{\omega}) \quad (3)$$

Substituting expression (2) into (1) we get:

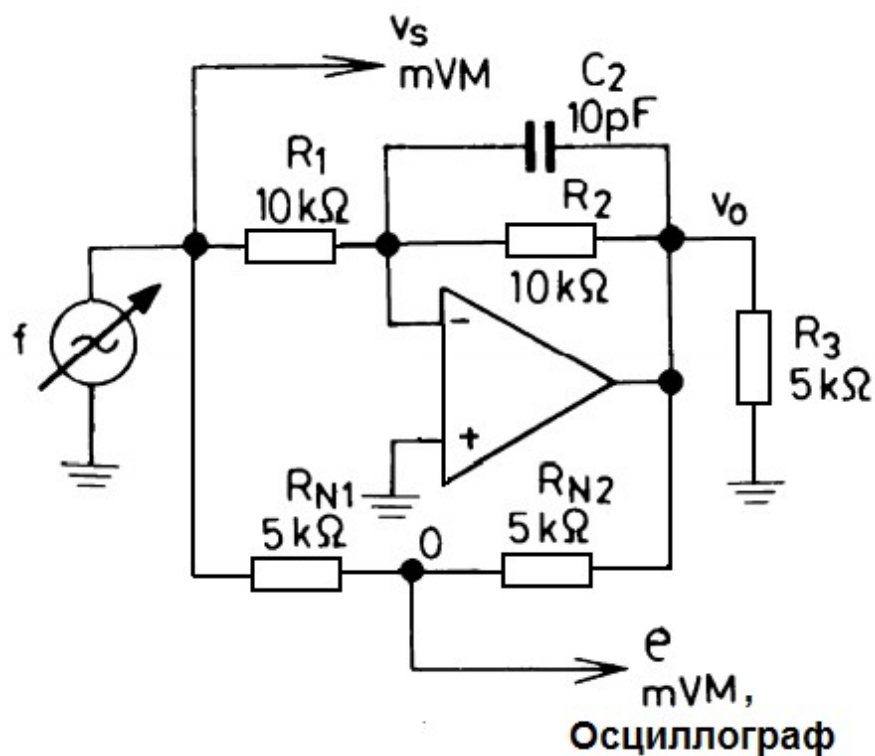
$$\epsilon_{exit} = -\omega_{exit}\tau_c = -2\pi f_{\omega} U_{exit}\tau_c = -2\pi f_{\omega} U_{exit}(t_{PD}) \quad (4)$$

Replacing the frequency with a period for the harmonic effect, we get:

$$\epsilon_{exit} = -2\pi U_{exit}\tau_c/T = -2\pi U_{exit}(t_{PD})/T \quad (5)$$

From paragraph 14 it follows that the speed error for non-harmonic effects and the vector error for harmonic effects are calculated using the same formula, which is not surprising, since the initial section of the sinusoid can be considered with good approximation as a non-harmonic effect.

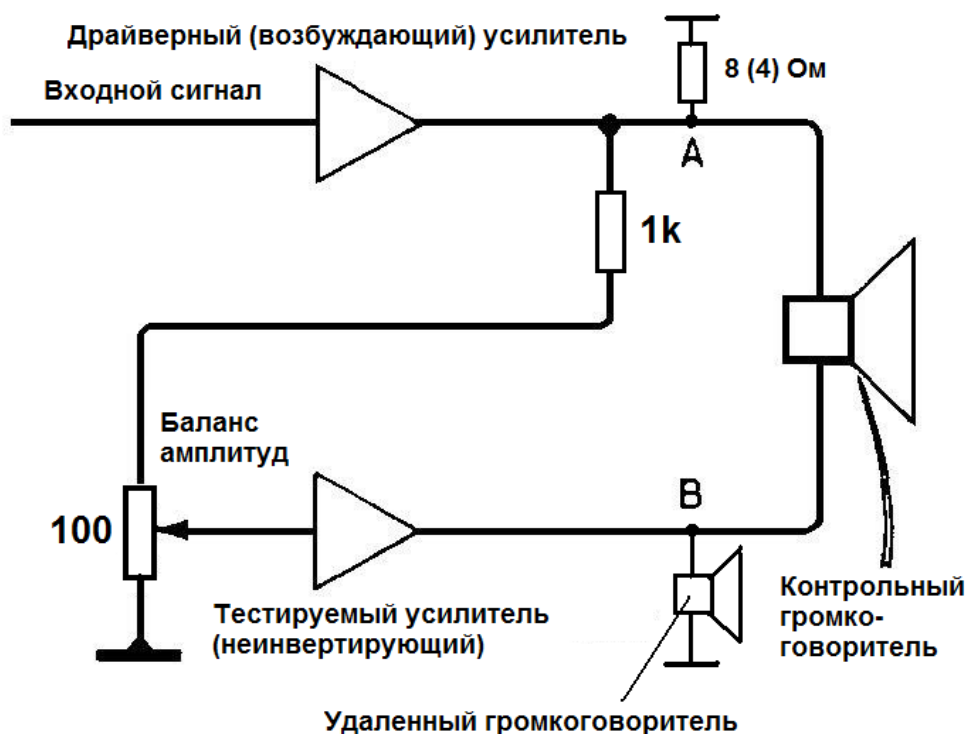
Methods for measuring the vector error consist in direct subtraction of the output signal from the input signal reduced in terms of level to the output signal. For example, for an inverting amplifier, it looks like this, fig. 13 [12]



Rice. 13 Scheme for measuring the vector error of the voltage inverter

Since the gain is minus 1, the result of the subtraction is carried out on resistors of equal size.

The developers needed a test that would correlate with sound quality. And David Hafler proposed such a test called SWDT ("straight-wire" differential test - differential test of "straight wire") [13], fig. 14.



Rice. 14 Block diagram of the Hafler vector error meter

The Hafler test is to compare the output voltage with the scaled input voltage (with the input voltage leveled to the output when viewed from the input)

amplifier under test to point A through the attenuator). Obviously, if the input and output are identical, there will be no signal in the monitor speaker (headphones). Any sound heard after careful balancing will be distortion. For the "null test" Hafler suggested a level of -70 dB for the middle frequencies (signal attenuation by 3000 times or up to 0.03%) and -60 dB (signal attenuation by 1000 times or up to 0.1%) for the higher frequencies of the audio range.

In one of the answers to questions on the article [13] on the application of the SWDT test, Hafler writes: "This is not a new idea. However, in the past, amplifiers have performed very poorly in this test, which has revealed all the shortcomings. The XL-280 may be the first amplifier to offer good zero signal with SWDT."

The idea is not really new, but audio amplifier designers want to see distortion products, not vector errors. But for this it is necessary to carry out a test taking into account the delay of the signal.

And although the Hafler test does not measure non-linear distortions, but only vector errors, nevertheless it is more informative than any other test.

Unlike the Sapozhkov-Baksandall method, this method does not use phase adjustment, which greatly simplifies the testing procedure, and on the other hand, it also takes into account the introduced phase distortions, which often lead to serious distortions associated with the amplitude-phase conversion of signals.

This idea was voiced in the report "Pitch, timbre, separation of sources and myths about sound reproduction through loudspeakers" by a well-known specialist in the field of psychoacoustics D. Griesinger at AES-132 held in Budapest in 2012. In his report, he showed that the phase relationships of the upper harmonics in the spectrum are extremely important, and also that when playing through loudspeakers, the structure of the amplitude and phase spectrum is always violated to one degree or another, so the sound is always poorer in timbre, clarity, clarity, etc. . than natural sound.

You may get the impression that the test result does not depend on the top amplifier in the circuit, because its task is only to give out the signal amplitude. In fact, it should be wide enough (high-speed) so as not to "blur" the subtle nuances of the sound material and not make it easier for the amplifier under test. Only in this case we will get more reliable information about the tested amplifier. If you use a "braked" amplifier (this may be one channel of the tested stereo amplifier used as a driver), then the tested amplifier may look better than it actually is.

Regarding the Hafler test, John Kerl said the following [14] (p. 40): "With all due respect to David Hafler. His amplifier may have passed his test, but this amplifier is not considered "perfect" by any standard and is usually "upgraded" by hobbyists, with "improvement" noted. It must be assumed that he meant the XL-280 amplifier.

At first glance, this statement may seem paradoxical. In fact, a lot depends on the test signals. If pure sinusoidal signals are used, the measurements will differ little from the steady state THD measurement, with the ensuing consequences. The most optimal signal for such testing is a triangular signal with a frequency of 10 kHz passed through a 100 kHz low-pass filter. Such a signal is very convenient as a test signal in all respects, the rate of rise / fall of the signal fronts is close to the maximum rise rate encountered in real audio signals, and at the tops of the signal there is a voltage change  $dV / dt$  characteristic of the subtle nuances of a real audio signal (reveals speed distortions twice over a period). *Appendixone*).

When using a sinusoidal signal, equal amplitudes at the amplifier outputs and no distortion, the difference signal will also be a sinusoid, but phase-shifted, the amplitude of which depends on the signal propagation delay of the amplifier under test (tPD):

$$a = 2\pi A * tPD / T \quad (6)$$

where

T is the signal period,  $\mu s$ ;

A is the signal amplitude at the outputs of the amplifiers,  
V tPD is the signal propagation delay time,  $\mu\text{s}$

From formula (6) it can be seen that the level of the vector error is inversely proportional to the period (directly proportional to the frequency). Therefore, at higher frequencies, it is most difficult to provide small vector errors.

Assume 3 kHz signal voltages at the amplifier outputs of 60 V (peak to peak) and a difference signal of -70 dB (attenuation by a factor of 3000), which corresponds to a vector error level of 20 mV (peak to peak). In this case, from formula (6) tPD is equal to:

$$t_{PD} = \frac{A}{2\pi f} \cdot \frac{0.02}{333} \cdot \frac{1}{6.28} \cdot 60 = 0.0177 \mu\text{s} = 17.7 \text{ ns}$$

Let's calculate the maximum allowable delay for a frequency of 20 kHz with a vector error of -60 dB, it will be only 8 ns.

I must say that these are quite stringent requirements for UMZCH, since the vast majority of amplifiers in operation have tPD from 200 ... 300 ns to 1.5  $\mu\text{s}$  or more.

If we focus on the amplitude accuracy of signal transmission of 0.01%, then in accordance with the conclusions of [12], the bandwidth of the amplifier with single-pole correction should be at least:

$$f_v = (100/\sqrt{2})f_{one} \approx 71f_{one} \quad (7)$$

We substitute the upper frequency of the audio range into formula (7), as a result we obtain the required bandwidth of 1.4 MHz. This means that the signal propagation delay time should not exceed:

$$t_{PD} \leq 1/2\pi f_v = 1/(6.28 \cdot 1400000) = 0.00000013 \text{ s} = 0.00013 \text{ ms} = 0.13 \mu\text{s} = 130 \text{ ns}$$

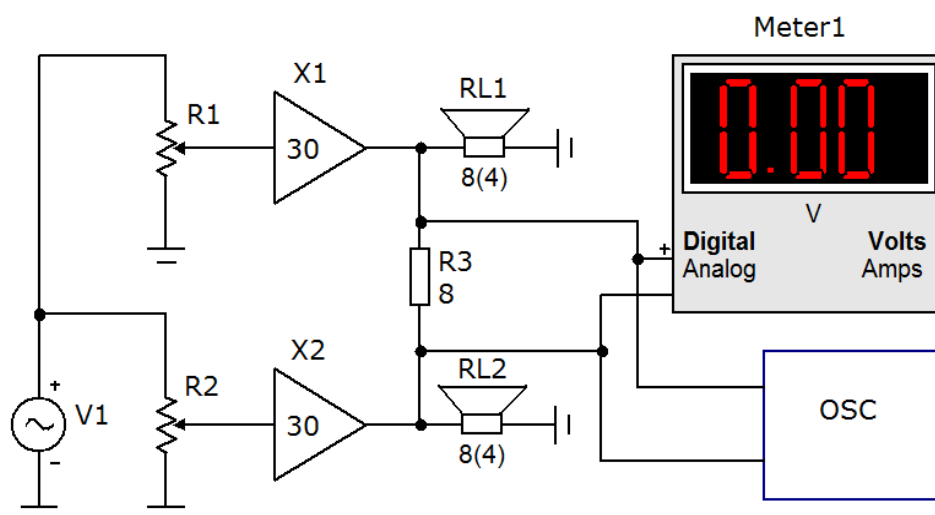
Thus, in accordance with the calculations by Jiri Dostal's formulas, the bandwidth of the amplifier must be at least 1.4 MHz, and the delay time of the signal in the amplifier itself (excluding the delay caused by the input RF low-pass filter) for high-quality amplifiers with OOS should not exceed 130 ns, but in this case it will not meet the more stringent requirements of the Hafler test.

In some modern top-end amplifiers, this parameter began to be indicated. For example, the technical specifications for the MIMESIS 9.2 amplifier indicate:

GROUP DELAY

- Propagation Delay: < 100 ns stable from DC to 200 kHz

A similar "null test" was also used by Bob Carver [15], fig. 15



Rice. 15 Block diagram for measuring the relative level of distortion with respect to the standard

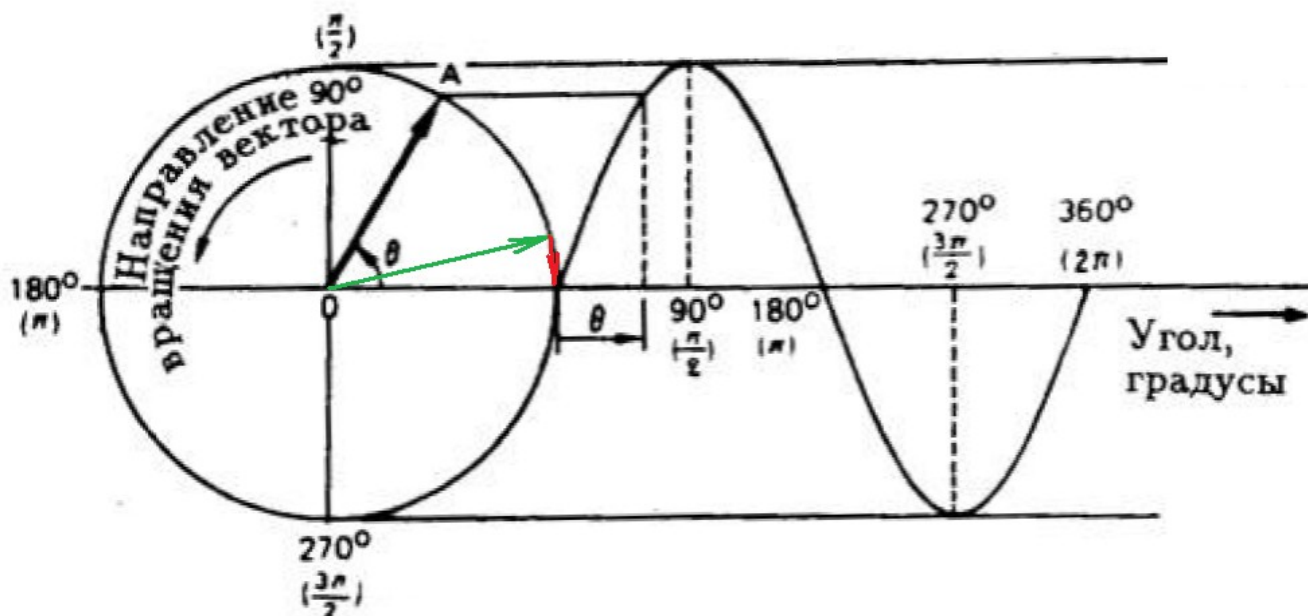
Unlike the original Hafler test, the Carver test uses a comparison between the output of the "reference" X1 and the output of the amplifier under test, X2. As the experiment showed, adjusting the parameters of an amplifier of the middle price category to the parameters of an expensive "standard" with an accuracy of -70 dB (0.03%) ensured the identity of their sound.

Since we are striving for perfect accuracy, when modeling with simulators, an ideal amplifier with a gain equal to the gain of the amplifier under test at the signal frequency can be used as a reference amplifier. Then, to measure insertion distortion, it is sufficient to delay the output signal of an ideal amplifier with an ideal delay line by the propagation delay time (tPD) in the amplifier under test.

The Carver method can also be used to measure the gain in distortion at maximum power (or any other) for example, with respect to an output power of 1 watt. To do this, amplifiers X1 and X2 must have the same parameters (for example, monoblocks of the same brand). At the output of one of them, we turn on the divider in parallel with the load (we take the resistor that is connected to the "common" equal to the load, and calculate the upper resistor so that at the connection point of the divider resistors there is a voltage corresponding to an output power of 1 W). It remains in the amplifier with a divider in parallel with the load to set the output power close to the maximum, and in the second amplifier, use the input attenuator to set the output voltage equal to the voltage at the division point of the resistors of the first amplifier. If the parameters of the amplifiers are the same (it is important that they are equal to tPD), then the vector distortions will be compensated and in the diagonal of the bridge we will get a pure increase in distortions. They can be evaluated using an oscilloscope, FIR or a spectrum analyzer.

It is known from psychoacoustics that human hearing is most sensitive to the rate of phase change, i.e. to the group delay (GDT) and its variations in the frequency domain. In fact, the constancy of the group delay is important not only in the audio range, but also far beyond it (up to 1 MHz and beyond).

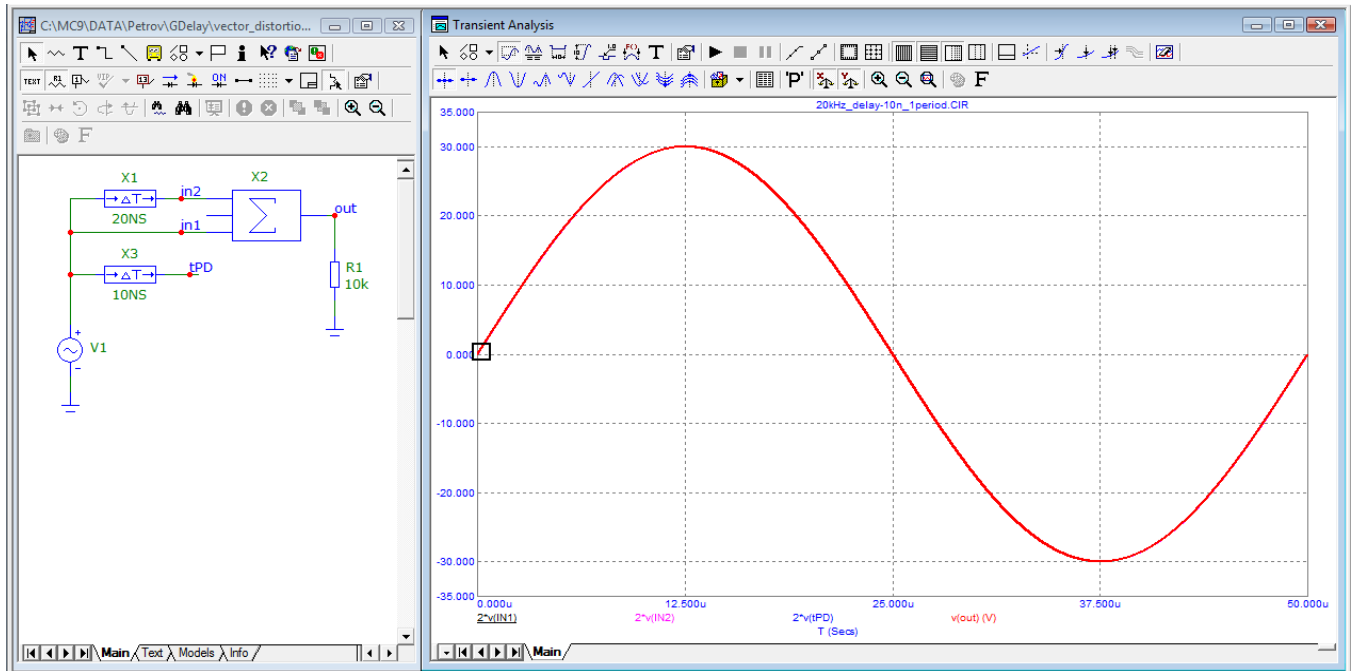
The relationship of vector errors with the delay time can be easily traced from the following figures, 16 - 18.



Rice. 16. Illustration of the mechanism of formation of a sinusoidal oscillation using a rotating vector.

The following illustrations show the relationship between vector error and propagation delay time. Datasheets for some high-speed op-amps list the propagation delay time as tPD (time Propagation Delay).

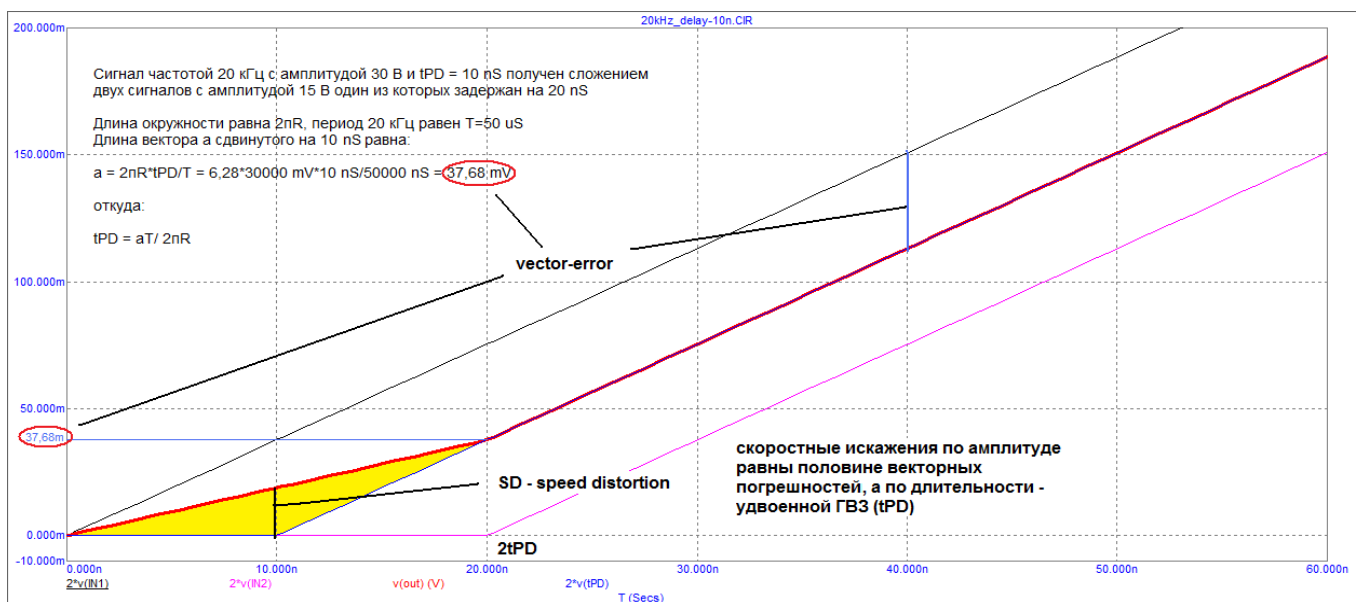
Figure 17 shows the ideal model of vector addition of sinusoids.



Rice. 17 Stress vector addition model

Figure 17 shows four sinusoids, but since they are delayed relative to each other for a short time, they visually merge. The oscillator signal with a frequency of 20 kHz has an amplitude of 15 V. When two signals are added, one of which is delayed for a short time, the amplitude of the output signal almost doubles and equals 30 V. In order for all four signals to be of the same amplitude, a multiplier of 2 is applied to the signals IN1, IN2 and tPD.

The initial section of the signal highlighted by the rectangle is shown in the following figure in a stretched form



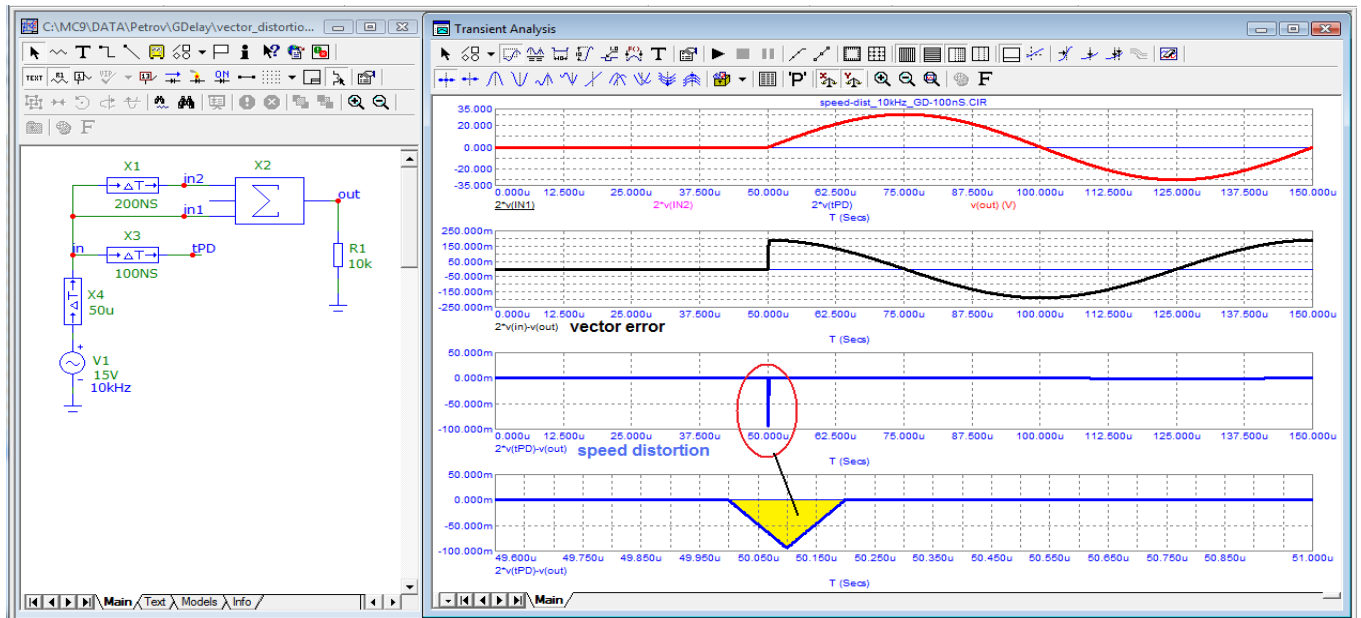
Rice. 18 The initial section of the result of the addition of 2 sinusoids

Figure 18 shows that when adding the original signal (black) and the signal (pink) delayed by 20 ns, an output signal (red) is formed with a delay of half that, i.e. 10 ns. Moreover, in the initial section, distortions inevitably occur over 20 ns, i.e., over  $2 * t_{PD}$ . Jiri Dostal mentions speed distortion in his book, but does not give a clear definition of what it is. I took it upon myself to name this kind of distortion

high-speed, since they are directly related to such a high-speed parameter as  $t_{PD}$  and occur at the moments of change in speed  $dV/dt$ . Ideally, these distortions are equal in amplitude to half the vector error (from the projection of the yellow triangle onto the horizontal axis), and in time  $2 \cdot t_{PD}$ . The amplitude and duration of high-speed distortion (yellow triangle) determines their contribution to the overall level of distortion.

The figure also shows a calculation in accordance with [12] showing a direct dependence of vector errors and high-speed distortion on  $t_{PD}$ .

In order to make the most of this ideal model, let us measure the speed distortions with its help, Fig. nineteen



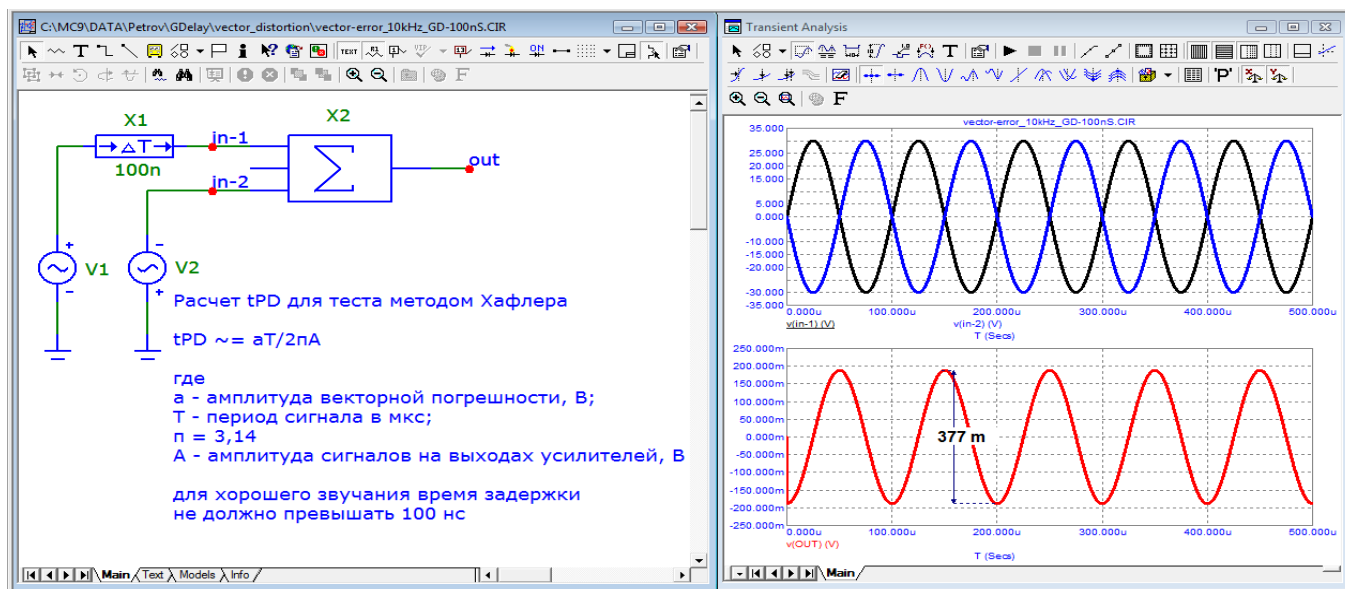
Rice. 19 The result of measuring the distortion resulting from the vector addition of two voltages

Figure 19 shows that when the output signal, delayed by 100 ns, is subtracted from the input signal, distortions occur with a negative sign with respect to the polarity of the first half-wave of the input signal and an amplitude equal to half the vector error and a total duration of 200 ns, i.e.  $2t_{PD}$ . These distortions have nothing to do with linear distortions (neither in phase nor in amplitude). These are the distortions I refer to as high-speed ones.

A sinusoidal voltage is formed by a vector of constant magnitude rotating at a constant angular velocity (see Fig. 16). With any change in both the vector amplitude and the angular velocity, the voltage  $dV/dt$  also changes. At these moments, additional distortions associated with the signal propagation delay time ( $t_{PD}$ ) occur.

For clarity of vector errors, let's assemble another model, fig. twenty





Rice. 20 The result of measuring the vector errors of the addition of two voltages

As a result of adding two signals in antiphase, one of which is delayed by 100 ns at the output, we obtained a vector difference in the form of a sinusoidal signal of the same frequency, the amplitude of which obeys formula (6):

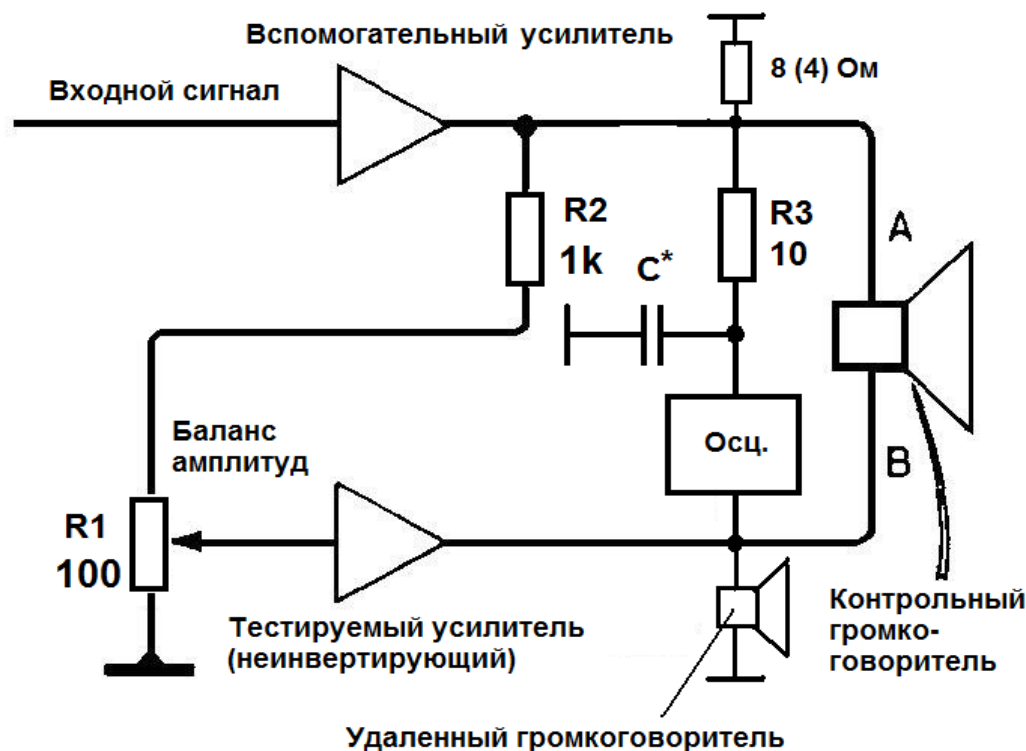
$$a \approx 2\pi A \cdot tPD / T \approx 6.28 \cdot 60 \cdot 0.1 / 100 \approx 0.377 \text{ V}$$

In real amplifiers, the amplitude of high-speed distortion can be somewhat lower than calculated - it depends on the value of tPD and its behavior both in the audio band and beyond it.

An example of the analysis of high-speed distortions of the Apex HD50 industrial amplifier model can be found in [16].

In order to isolate distortion products in more detail (in particular, switching distortions), it is necessary to use an ideal delay. But this can only be done in the simulator. In the gland, you can combine the Hafler test with the Sapozhkov test, Fig. 21.

A tunable delay line can be made with an acceptable error using a simple RC circuit whose time constant is tPD. The capacitor should be used high-voltage, for a voltage of at least 250 V, and the probe should be with a calibrated divider. To increase sensitivity when using a spectrum analyzer instead of an oscilloscope, you can use a wideband preamplifier. If measurements are made without a divider, then the calculated capacitance must be taken into account the capacitance of the connecting cable.

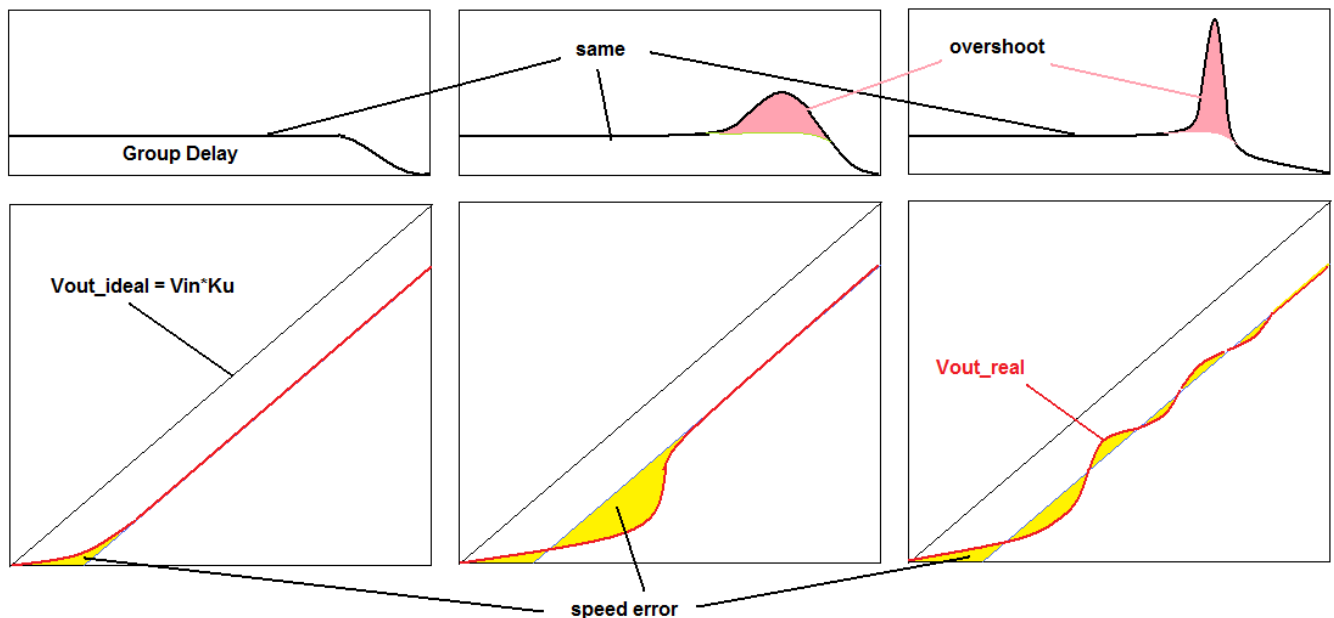


Rice. 21 Hafler block diagram combined with Sapozhkov vector distortion analyzer

As a delay line, in some cases, you can use the Boucher chain of the upper amplifier. A typical Boucher circuit consists of a 10 ohm resistor and a 100 nF capacitor connected in series, the time constant of such a circuit is 1  $\mu$ s. As practice shows, the signal propagation delay time in real amplifiers varies widely: from tens of ns to 1.5  $\mu$ s or more. Therefore, it is quite possible to use the Boucher circuit of the upper amplifier to form a delay by the calculated value by replacing the capacitor and adjusting the resistor  $R_3$  to the optimal value, taking into account the input capacitance of the oscilloscope cable.

If the delay  $t_{PD}$  is not more than 100 ns, then the high-speed distortions are already negligible and can be neglected, they will be at the level of other types of distortions. If the delay is large (maybe 1.5  $\mu$ s or more), then such a delay will compensate for high-speed distortions, but other types of distortions (crossover, nonlinear) will still show, with the possible exception of distortions caused by amplitude-phase conversion, since phase and amplitude errors will be largely compensated by the delay line on the RC chain.

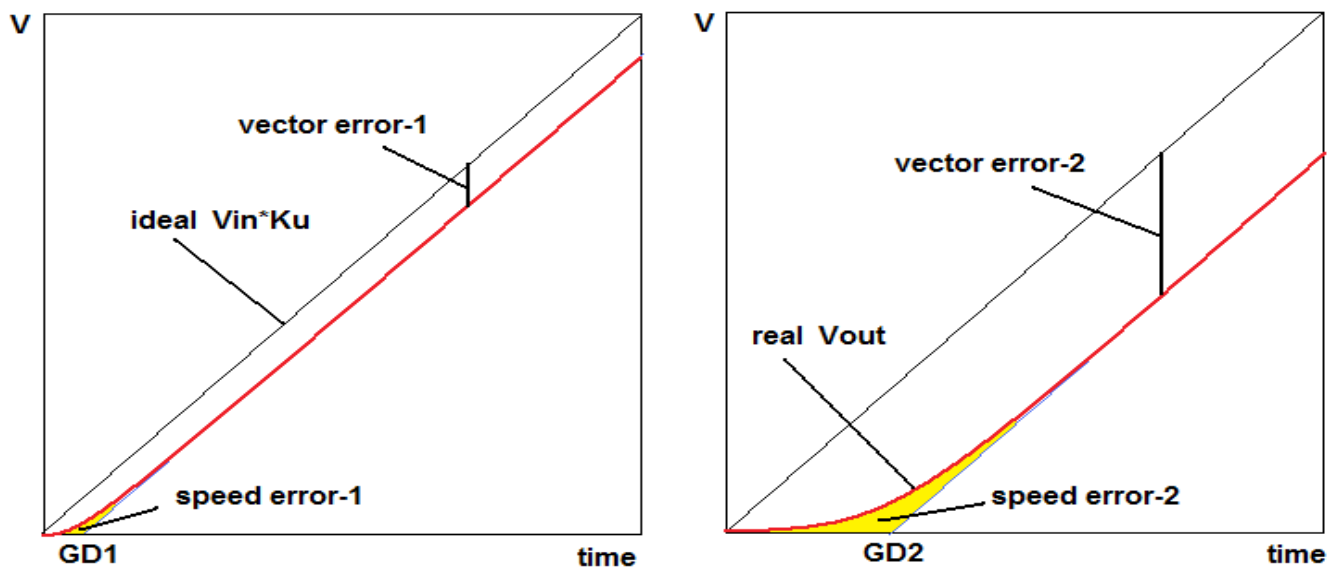
You can often hear from "theorists" that the manifestation of group delay in amplifiers is no different from the distortion introduced by a simple RC circuit. This can be said by someone who does not know that the RC circuit introduces a delay that smoothly drops to zero beyond the passband, while in amplifiers the behavior of the group delay ( $t_{PD}$ ) is unstable and very diverse (often with significant spikes in the positive region and dips in the negative region). It is the nature of group delay behavior that determines the duration and nature of transient processes and introduced distortions in the first periods (especially in the first half-cycle). Examples of group delay behaviors most frequently encountered in practice in UPT are shown in fig. 22



Rice. 22 Examples of practiced group delay behaviors in UPTs

In amplifiers with a closed input (with a coupling capacitor at the input and in an OOS divider), as well as in a UPT with a servo control system, the group delay behavior is much more complicated, often with a “fly away” to the negative region.

Figure 23 shows the dependence of high-speed distortion on the group delay of two amplifiers



Rice. 23 Two examples of the dependence of high-speed distortion on group delay

An amplifier test with a characteristic similar to that shown in Figure 22 (middle) is given in *application one*.

The Hafler test and the Hafler-Sapozhkov test are based on subtracting the scaled original (input) signal from the output signal. The absence of any (audible) residue proves that there are no mysterious, unmeasurable ingredients. Both methods can be successfully used to evaluate the performance of amplifiers.

The advantages of the latter methods include the possibility of using any signals as test signals, including musical ones, as well as their extreme simplicity, which does not require expensive special measuring equipment, which is no less important.

When using a triangular signal during balancing, a differential voltage close to a meander is achieved. In this case, the meander amplitude will depend on the signal propagation delay tPD as follows:

$$a = 4A(tPD)/T$$

(eight)

where

T is the signal period,  $\mu\text{s}$ ;

A is the signal amplitude at the outputs of the amplifiers,

V tPD is the signal propagation delay time,  $\mu\text{s}$

When using a test signal in the form of a meander, the difference signal should look like a straight line on which bursts of up-down amplitude occur opposite the signal fronts and the duration of which depends on tPD.

To quantify the introduced high-speed distortions (measure their amplitude and duration), it is best to use an oscilloscope.

Today, the design of amplifiers does not occur without their preliminary simulation. Vector errors can be measured according to Dostal, and all other types of distortion (nonlinear, crossover and speed) can be measured using a compensation method using an ideal delay line equal to the signal propagation delay time at the tested frequency. When making measurements, to improve accuracy, careful tuning should be done to determine both the gain and tPD as accurately as possible.

### **Methods for measuring dynamic non-linear distortions [17].**

One of the widely used methods for measuring dynamic non-linear distortion is the so-called "sine-rectangular signal" method. The measuring signal is a sequence of rectangular pulses with a repetition rate  $f_{\text{one}} = 3.18 \text{ kHz}$ , which is superimposed by a harmonic signal of frequency  $f_2 = 15 \text{ kHz}$ . The ratio of the voltage amplitudes of the signals  $U_{\text{one}} : U_2 = 4:1$ , the slew rate of signals is limited by the low-pass filter on RC links, the cutoff frequency of the filter is  $f_{\text{LP}} = 30$  and  $100 \text{ kHz}$  (respectively DIM-30 and DIM-100). The measure of non-linear distortion is the ratio of the effective values of the intermodulation products to the amplitude of the harmonic signal.

Another method is based on the use of a measuring signal, which consists of two harmonic signals of the same amplitude, slightly different in frequency. Such pair frequencies can be, for example,  $f_{\text{one}} = 14$  and  $f_2 = 15 \text{ kHz}$  or  $f_{\text{one}} = 19$  and  $f_2 = 20 \text{ kHz}$  (CCIF-IM). The measure of distortion is the ratio of the effective values of the amplitudes of the intermodulation products  $m f_{\text{one}} \pm n f_2$  to the amplitude of the main signals.

The third method - the sawtooth method - is based on the use of a sawtooth signal with a pulse repetition rate  $f_{\text{one}} = 20 \text{ kHz}$ , which periodically (with frequency  $f_2 = f_{\text{one}}/256$ ) reverses polarity. Distortions caused by a limited signal slew rate appear as periodic (with a frequency  $f_2$ ) offsets of the average values of the DC component of the voltage. The measure of distortion is the ratio of the levels of this low-frequency voltage and the sawtooth signal.

It should be noted that not all testing methods are considered. For example, the review did not include Hiiragi's inverse intermodulation method (RIMD) [18], as well as Hirata's testing method [19]

As for the use of NFB to reduce distortion, there are several authoritative statements about this. For example, Kiril Hammer said the following in an interview [20]: "Perfect performance is equally important. This is especially true for amplifiers with common feedback. The theoretical concept of negative feedback is very powerful, and the simplified mathematical equations that describe this concept always hold true. But they are only valid if the design takes into account the limitations of the concept. The delay from entry to exit must be zero! Obviously this is not possible in real life.

There are two ways to solve this problem:

1. You simply do not apply any negative OS at all in your project (giving up the benefits of the concept).
2. You speed up the amplifier to a few nanoseconds of time delay from

input to output (respectively 200 MHz bandwidth...), then the compensation errors are so small that they do not have any noticeable effect on the sound.

As soon as you decide to go the second way, a lot of new problems suddenly arise: thermal conditions, supply voltage stability, design suitable for high-frequency devices, the occurrence of noise, interference, etc."

Here it is appropriate to recall the statement of Martin Colloms [21]:

"Engineering can explain a lot about the world of sound. However, when that fails to explain something, the real fun begins. Some aspects of perceived sound quality are not explained by established theory. There is growing suspicion that some of these aspects are a loss of natural timbre; boring, less expressive performance; increased auditory fatigue; and the lack of life and energy in the reproduced sound - may be a consequence of the application of negative feedback.

Emeritus Professor Malcolm Hawksford paid much attention to the first path, proposing highly linear amplification stages with distortion compensation, including those with positive feedback (POS) for the output current [22].

Manufacturers producing two types of amplifiers are constantly competing with each other. And although the first amplifiers formally have a higher level of distortion measured on a sinusoidal signal in a steady state, in subjective tests they most often turn out to be the winners.

For example, Jeff Rowland believed that "amplifiers with OOS introduce distortion in the time domain" therefore he developed the "model 7" non-OS amplifier in three modifications that differ in both voltage amplifiers (VU) and output stages (VK) - technical solutions are not disclosed. A characteristic feature of these models is the constancy of the output impedance over the entire sound range, an extremely high load capacity (up to 150 A-peak) and an adequate response (in phase) to external influences in the form of back-EMF acoustics. And although they had slight differences in sound, they nevertheless received high marks from professionals and lovers of high-quality sound.

Suffice it to say that individual components of the company cannot be purchased even in the secondary market - amplifiers such as "Coherence", Model 7 and Model 9 have become rarities and a living legend among music lovers.

The second path is used by the legendary designer and consultant for several audio companies, John Curl. In doing so, he adheres to the following rules:

- output voltage slew rate not less than 100 V/ $\mu$ s;
- the frequency of the first pole is as high as possible in the sound range (preferably not lower than 100 ... 200 kHz);
- minimal feedback, and its absence is better;
- operation of the output stage in class A up to 10 W or more (mitigates switching distortion);
- as low as possible the level of high-order odd harmonics starting from the 5th and higher; if possible, avoid the use of an output choke to ensure stable operation, since a choke with an inductance of even a fraction of  $\mu$ H introduces a significant delay and, in combination with a reactive load, can create oscillatory transients;
- quality power for each cascade.

He also shared an experiment he demonstrated to his collaborators about 40 years ago. Otala's Electrocompaniet was used as an amplifier, and a Pioneer ribbon tweeter with a bandwidth of more than 45 kHz was used as a tweeter. As a test signal, he used a rectangular signal (meander) with a frequency of 5 kHz, processed by switchable first-order low-pass filters of 35 and 100 kHz. In this case, the rise time of the fronts changed from 10  $\mu$ s to 3.5  $\mu$ s. All employees heard the difference in sound. From this simple experiment, he deduced that human hearing is more sensitive to the rate of change of a signal than to the actual frequency response.

More details about his ideology can be found in [14], as well as in a number of statements on forums, in particular [23].

For example, in an interview [20] Nelson Pass said the following:

"The last ten years have shown once again that high quality amplifiers with per millionth distortion and other excellent performance are not very popular. It's like pure distilled water - it has no taste and most people don't want to drink it. »

Not surprisingly, many well-known manufacturers such as: Akai, Denon, darTZeel, Denset Beat, Krell, Lamm, Pass Laboratories, Pioneer, Rowland, Sony, Tandberg, Threshold and others have produced models without a general OOS. And companies like NAD began to raise the frequency of the first pole above the audio range, while limiting the loop gain.

In one of the interviews, V. Shushurin (Lamm) gave the following example:

"We have three amplifiers: one has 1-2% distortion, the other has 0.1% distortion, the third has thousandths of a percent. We listen to all this through a speaker system that has 5% distortion. In theory, we should not hear the difference - only the handwriting of the loudspeaker. But we can hear perfectly well that the amplifiers sound different. Paradox, right? The answer to this question is not so easy to find. It took me several years to figure out how to answer it." True, he did not share the answer ...

And this is not surprising - according to the Rakovsky scale, Kg is at the lowest level of correlation with sound quality. Much more important is the proportion of higher odd harmonics starting from the 5th and higher (especially the 7th and 9th).

John Kerl, referring to the 1941 German "Handbook of Radiotron Engineering" gives the following harmonic weighting factors [14]:  $N^2 / 4$  ( $N$  to the power of 2 divided by 4) for each harmonic. In this case, the second harmonic has a weighting factor of 1, and for example, the 7th harmonic is already 12.5 (22 dB higher).

Moreover, Kg does not take into account such a high-speed parameter as the signal propagation delay time (time Propagation Delay) and its behavior far beyond the audio range. But it is the group delay that has the greatest impact on the sound quality. The IRI measures distortion in a steady state (i.e., in a linear mode) and does not take into account speed and transient distortion.

As for the effect of harmonic components on the timbre of sound, this was described more than 80 years ago [24]:

"The second harmonic adds clarity and brightness, but nothing more, since the general principle is that adding an octave cannot make any difference in timbre or distinctive musical quality. When the second harmonic has the same strength as the first, it has almost the same effect as adding an octave connector on an organ or harmonium, or playing in octaves instead of individual notes on a piano.

The third harmonic again adds some brightness due to its high tone, but it also makes a difference in timbre, thickening the tone and adding to it a breathy or nasal character that we can recognize as one of the main ingredients of the clarinet tone.

The fourth harmonic, being two octaves higher than the fundamental, adds even more brightness and perhaps even shrillness, but no more than that, for the reason already explained.

The fifth harmonic, in addition to adding even more brilliance, gives the tone a richness, somewhat similar to a horn, and the sixth adds a subtle piercing nasal quality.

All these six harmonics form parts of a common root note chord and therefore agree with that note and with each other.

However, the seventh harmonic introduces an element of dissonance. The same is true for the ninth, eleventh, thirteenth and all higher odd harmonics; they also add dissonance as pitch harshness, and thus bring roughness or harshness to the compound sound. The resulting tone quality is often described as "metallic"

Back in the 1950s, Theo Williamson wrote that for a high-quality amplification of an audio signal, it is sufficient that  $K_r$  be no more than 0.1% at maximum power. In this case, the harmonic content is virtually undetectable in the most sophisticated listening tests (apparently he was referring to tube amplifiers with a short spectrum, this does not apply to modern transistor amplifiers). He also noted that phase shifts between the harmonic components of a complex signal in dynamics, especially on attacks of sounds, have a significant impact on the naturalness of the sound.

Amplifiers with a low first pole are known to have a loop gain phase of 90

deg. in almost the entire audio range. Here is what S. Ageev writes about this [25].

"When the phase angle of the loop amplification is close to  $\pm 90$  or  $\pm 270$  degrees, the amplitude non-linearities of the original amplifier are almost completely converted into phase ones (i.e., into parasitic phase modulation, albeit weakened by  $|bK|$  times). In this case, parasitic amplitude modulation practically disappears, and the results of intermodulation distortion measurements can be 20 ... 30 dB more optimistic than the spectrum analyzer (and hearing in the case of UMZCH) actually shows. Unfortunately, that's just the way things are. with most OUs and many UMZCHs."

Attempts have been made to artificially widen the region of constant loop gain by using a two-pole correction. But this only gives an improvement in the measured parameters (THD), the sound quality only deteriorates due to changes in the group delay characteristic.

As for the speed parameters of modern amplifiers, in particular the slew rate, for example, the top-end NAD M3 amplifier has a slew rate (SR) of 1000 V/ $\mu$ s; HK Citation XX - 500 V /  $\mu$ s; Denon POA 2400 - 500 V/ $\mu$ s; HK PA-2400 - 280 V /  $\mu$ s; KR-8050 - 200 V/ $\mu$ s; KR-770 - 180 V/ $\mu$ s; HK Citation 22 - 160V/ $\mu$ s; Sansui G-22000 - 175V/ $\mu$ s etc. - the list goes on and on...

I hope that the presented material will allow us to take a fresh look at the long-known information both on the requirements for high-quality sound amplification amplifiers and on their testing methods to obtain the maximum correlation of measurement results with sound quality.

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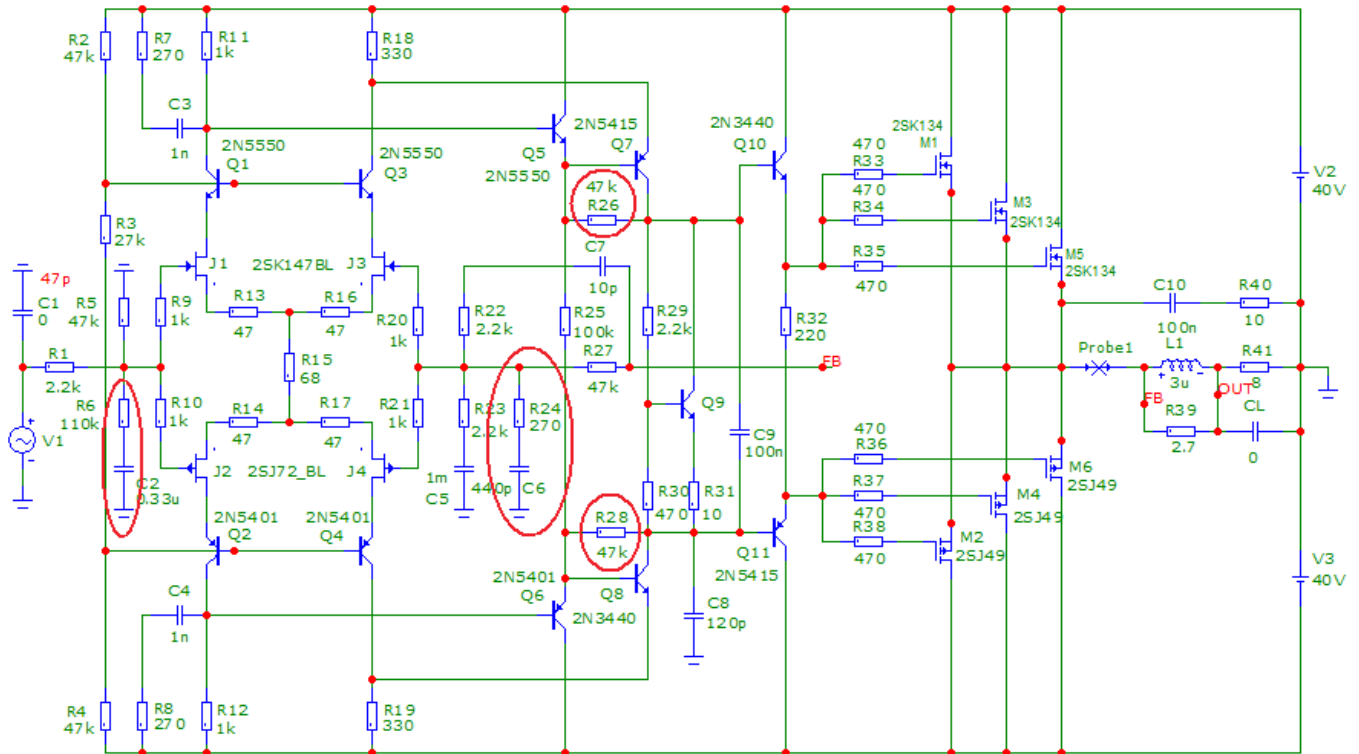
Petrov Alexander Afanasyevich

## Annex 1

### SWDT Hafler in action

As already noted in the article, Hafler's idea is to reduce vector errors in the high frequency region up to -60 dB (1000 times) and below, and in the medium and low frequencies below -70 dB (3000 times).

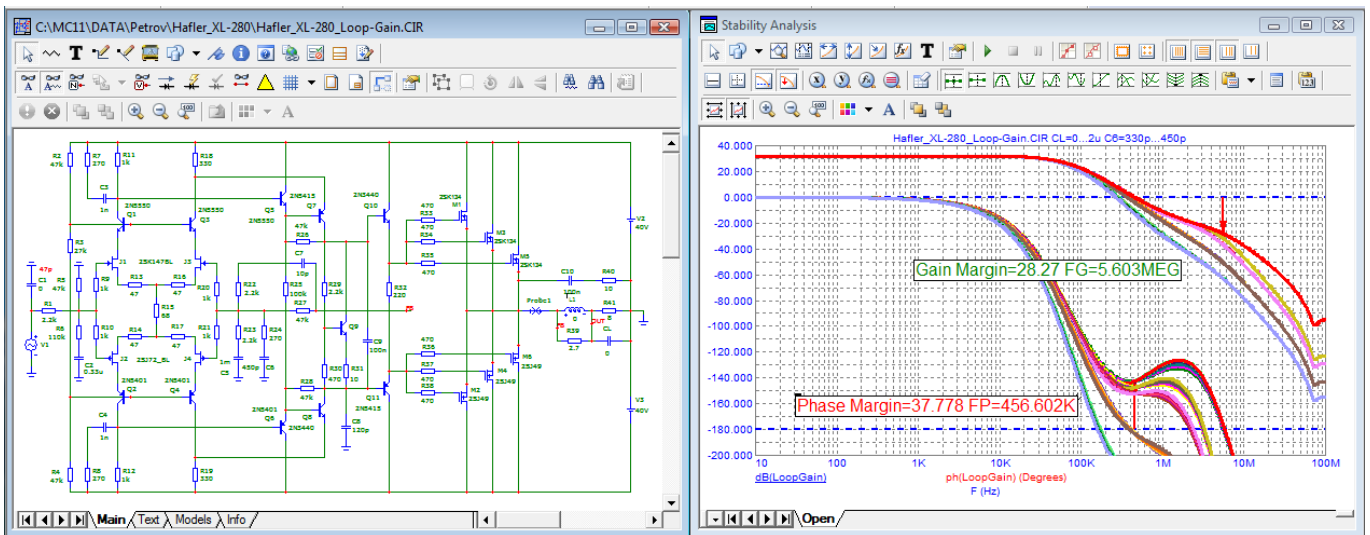
Consider D. Hafler's approach to design and testing on the example of his demonstration amplifier Hafler XL-280 Excelinear, fig. one



Rice. 1 Schematic of the XL-280 amplifier

With the help of local OS R26, R28, Hafler raised the frequency of the first pole to 30 kHz. In this case, the loop gain in the entire sound band is constant and slightly more than 30 dB. Using the circuit R6, C2, he compensated for the influence of the capacitor C5 in the OOS divider. With the help of a tunable capacitor C6, the adjustment of the signal propagation delay in the audio band is introduced. Capacitor C6 together with capacitors C7 and C8 create a resonance at a frequency of 450 kHz with a rise in frequency response up to 6 dB.

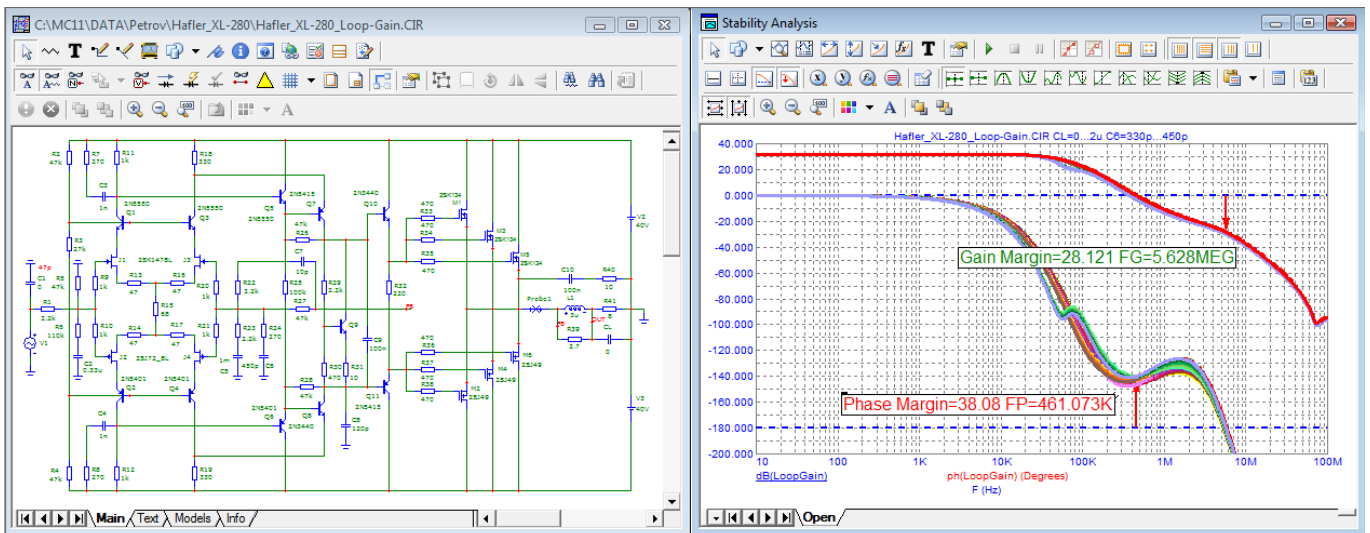
Let's take a loop gain diagram with a reactive load from 50 nF to 2  $\mu$ F in the entire tuning range of the capacitor C6, which is responsible for the delay time of the signal, fig. 2



Rice. 2. Bode diagram of the original circuit (L1=0)

From the Bode diagram, it can be seen that with a capacitor in the load of 0.5  $\mu\text{F}$  or more, the amplifier model without inductance at the output is unstable (there is no phase margin).

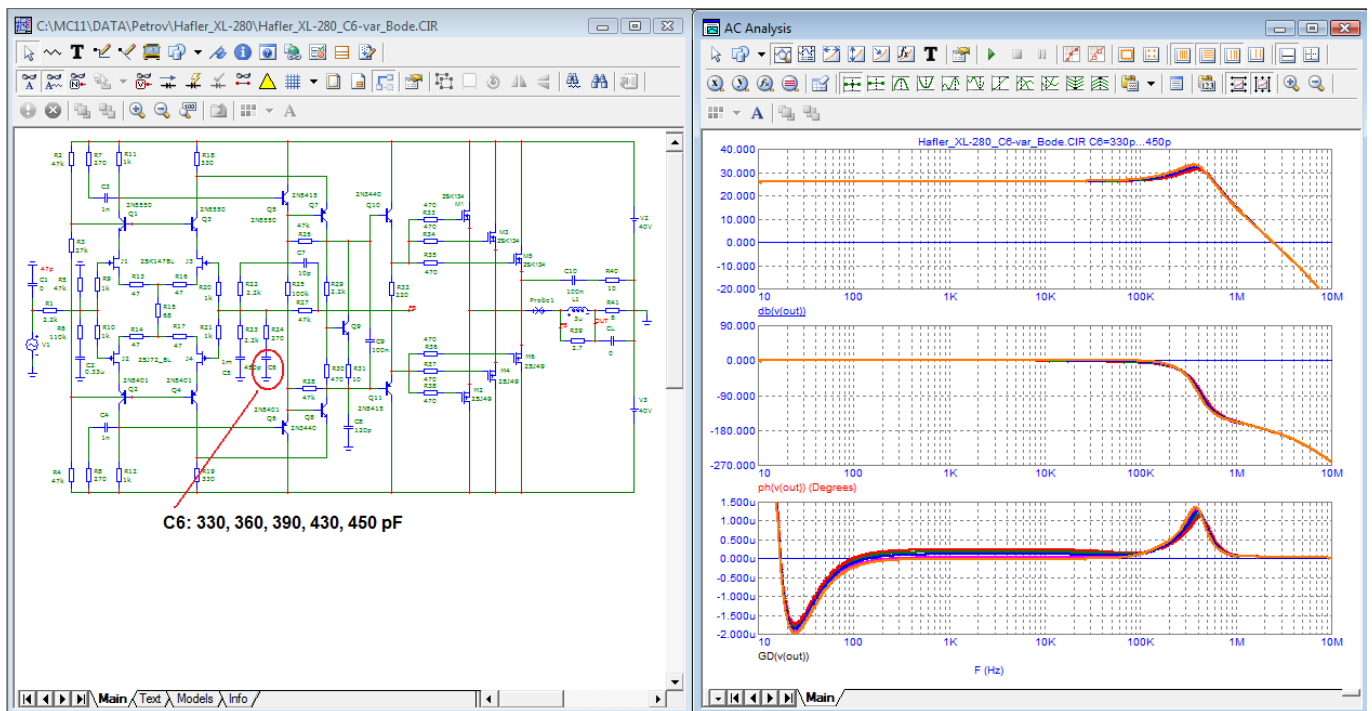
Let's use the standard technique to ensure stability with the help of a throttle, fig. 3.



Rice. 3. Loop gain with choke

The optimal value of the inductance of the inductor turned out to be 3  $\mu\text{H}$ , and the optimal shunt resistor was 2.7 Ohm.

Let's take a Bode diagram, fig. 4

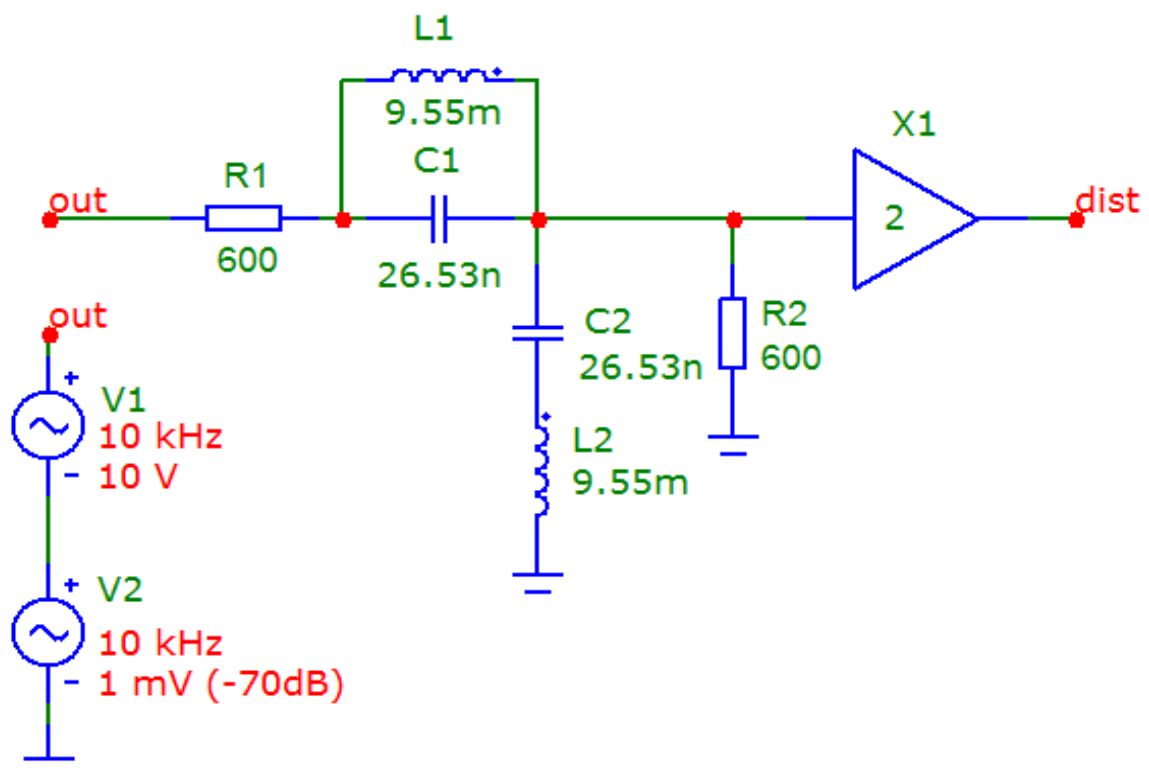


Rice. 4 Bode diagram and tPD adjustment limits

It can be seen from the Bode diagram that in the frequency range from 200 Hz to almost 100 kHz, the group delay is constant, however, at a frequency of 450 kHz, the group delay rises up to 1.3  $\mu$ s, and in the LF region (approximately 25 Hz), the group delay takes a negative value up to minus 2 ms. It is the group delay surge at a frequency of 450 kHz that determines the high-speed distortion!

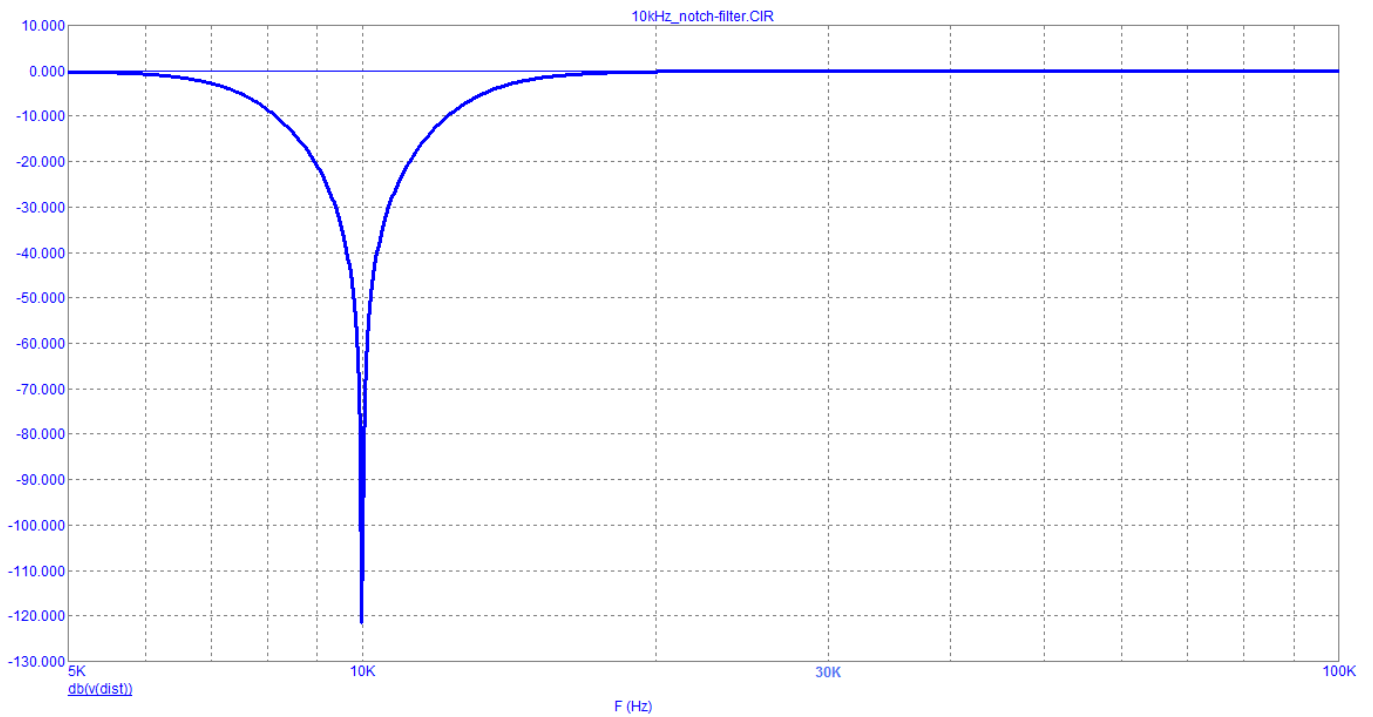
Without capacitor C6, the group delay is 550 ns (which is a lot) up to 100 kHz, then there is a smooth decline. Thus, without capacitor C6, the group delay is more than 4 times higher than the level allowed for high-quality amplifiers.

Let's measure the distortion introduced by the amplifier model at a frequency of 10 kHz. To do this, we use a notch filter, Fig. 5



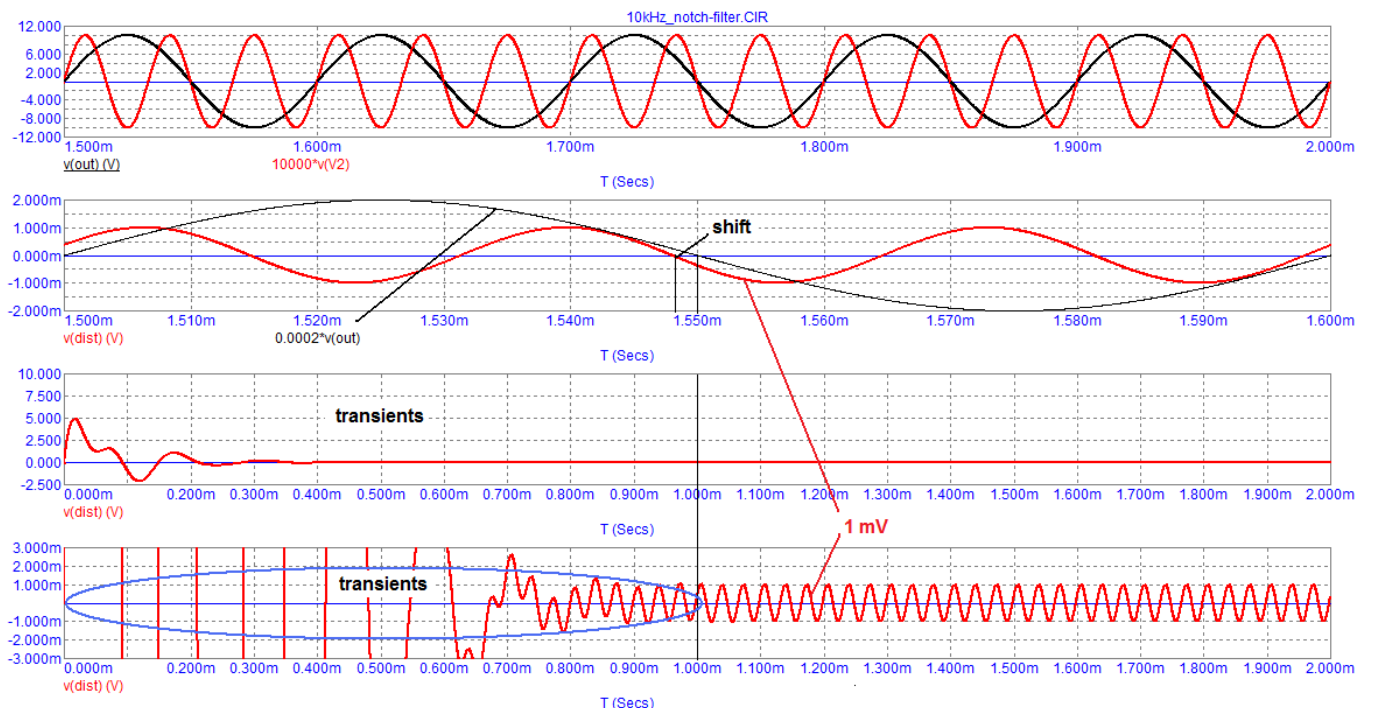
Rice. 5. 10 kHz notch filter circuit

The Bode diagram of such a filter is shown in Fig. 6



Rice. 6

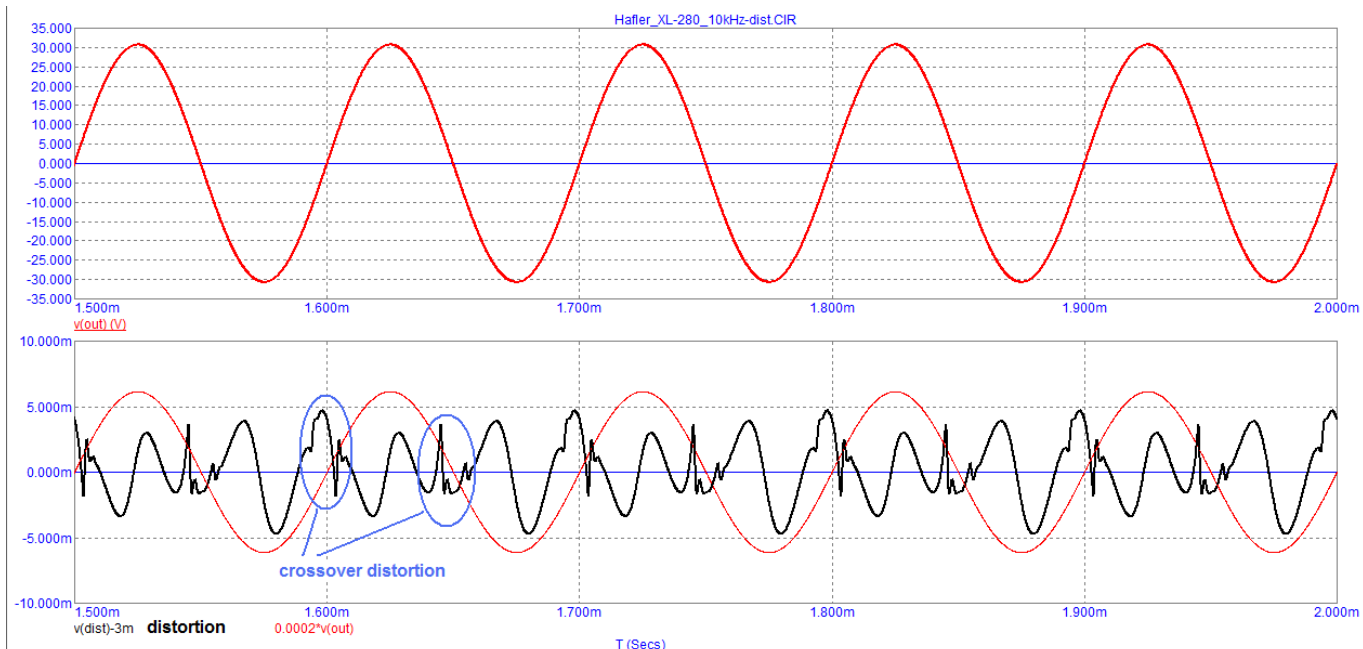
From the Bode diagram, it can be seen that the signal attenuation at a frequency of 10 kHz is more than 120 dB. Let's apply a signal with a frequency of 10 kHz at a level of 10 V (peak) and a signal with a frequency of 30 kHz (3rd harmonic) at a level of 1 mV (peak) to the input of such a filter and see how the filter copes with the measurement of an artificially added 3rd harmonic, fig. 7



Rice. 7 3rd harmonic measurement result

As can be seen from the test result, transient processes in the filter itself take about 1 ms (at least 10 periods of the main signal). The third harmonic is isolated with high accuracy in amplitude and a small phase shift (which is natural).

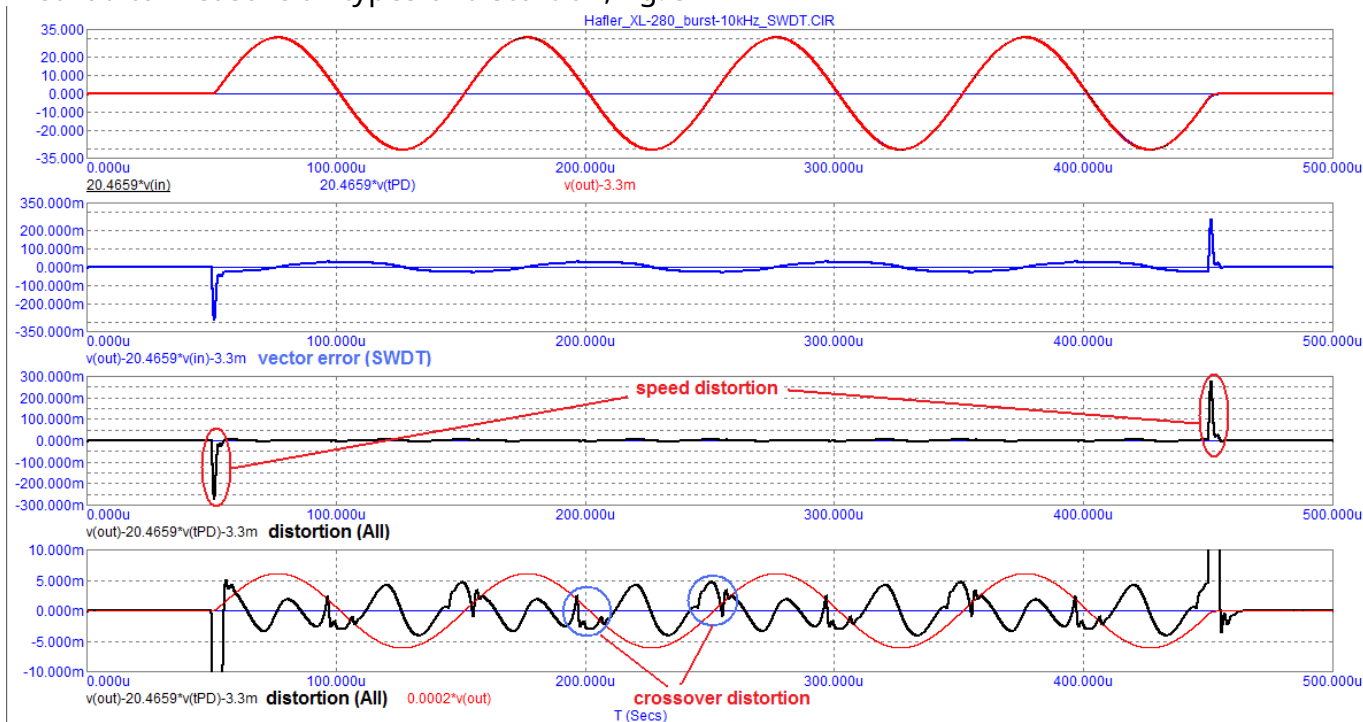
We apply such a filter to evaluate the distortion introduced by the amplifier model, Fig.



Rice. 8. The result of measuring the distortion of the model at a frequency of 10 kHz

In the distortion spectrum, we see mainly the 3rd harmonic and commutation distortions commensurate with it in amplitude.

Now let's use the Hafler test to measure vector distortion and the compensation method to measure all types of distortion, fig. 9



Rice. 9 SWDT - Hafler test

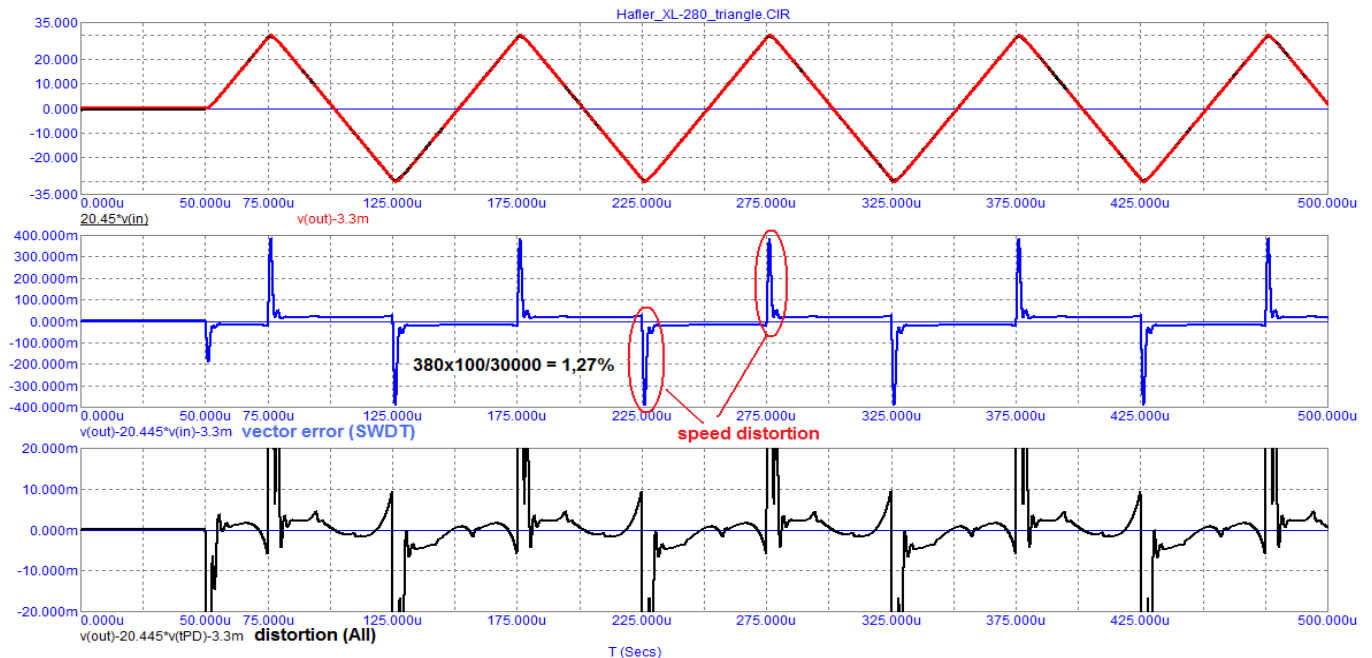
In the second graph (blue) we see the vector errors obtained by subtracting the scaled input signal from the output voltage. They fully meet the requirements of Hafler - suppression of -60 dB (1000 times), i.e. up to 30 mV from 30 V. If we use the Hafler method formally using fixed frequencies, we will get a result that differs little from the measurement of harmonic distortion in steady state. We also made sure that the results of measurements using the notch filter and the compensation method are the same (4th graph). However, at the beginning of the burst and at its end, we see significant distortion with an amplitude of about 250 mV. This is the speed distortion that is not determined by other methods.

Thus, Hafler was clearly mistaken in believing that it was the magnitude of the vector

errors associated with the delay time of the signal is responsible for the distortion introduced by the amplifier. He did not understand the reasons for the occurrence of additional distortions associated with the behavior of the group delay beyond the passband.

The distortion that occurs at the beginning of a sine wave is called "first cycle distortion" (FCD) by Graham Maynard. <https://www.diyaudio.com/forums/solid-state/32758-cycle-distortion-graham-post379900.html>

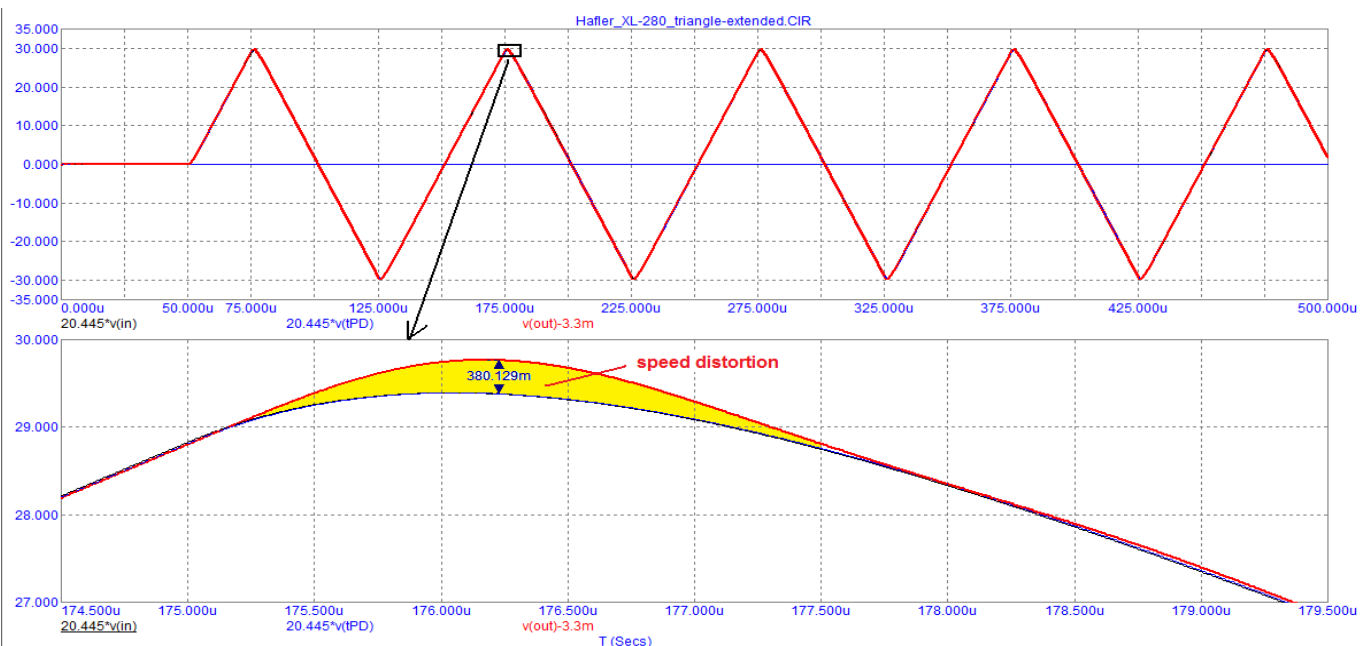
The next signal that is very convenient for use in the SWDT test is a triangular signal with a frequency of 10 kHz, fig. 10.



Rice. 10. SWDT test with triangular signal.

The second graph shows the Hafler vector errors (blue). Opposite the tops of the triangular signal, we see surges with an amplitude of 380 mV or 1.27%. The lower graph (black) shows all types of distortions obtained by the compensation method. The curvature of the line between the high-speed distortions indicates the linearity of the transfer characteristic.

Let's stretch a section of one of the vertices and look at the speed distortions "under the microscope", fig. eleven.

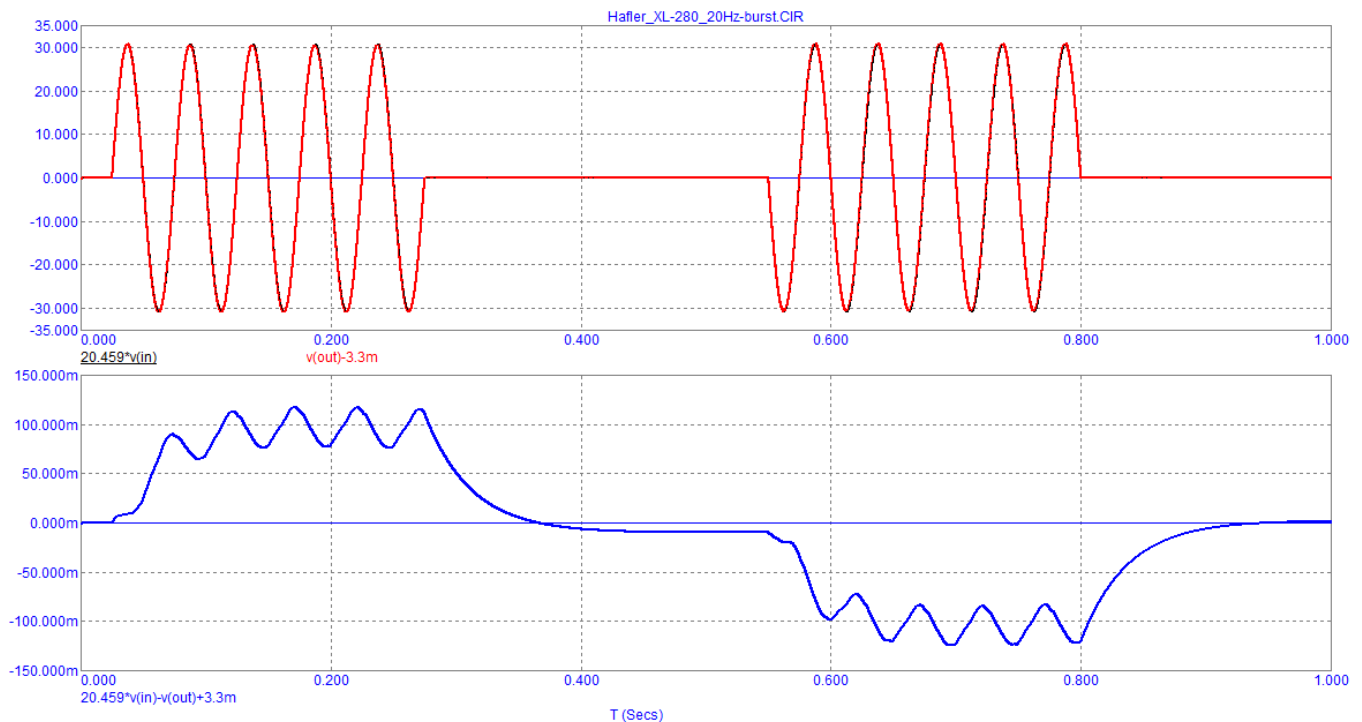


Rice. 11. High-speed distortion



The slew rate of such a signal at the output of the amplifier with an amplitude from peak to peak of 60 V and a duration of 50  $\mu\text{s}$  will be equal to:  $60/50 = 1.2 \text{ V}/\mu\text{s}$ , i.e. tens or even hundreds of times lower than the slew rate of the output voltage of the amplifiers.

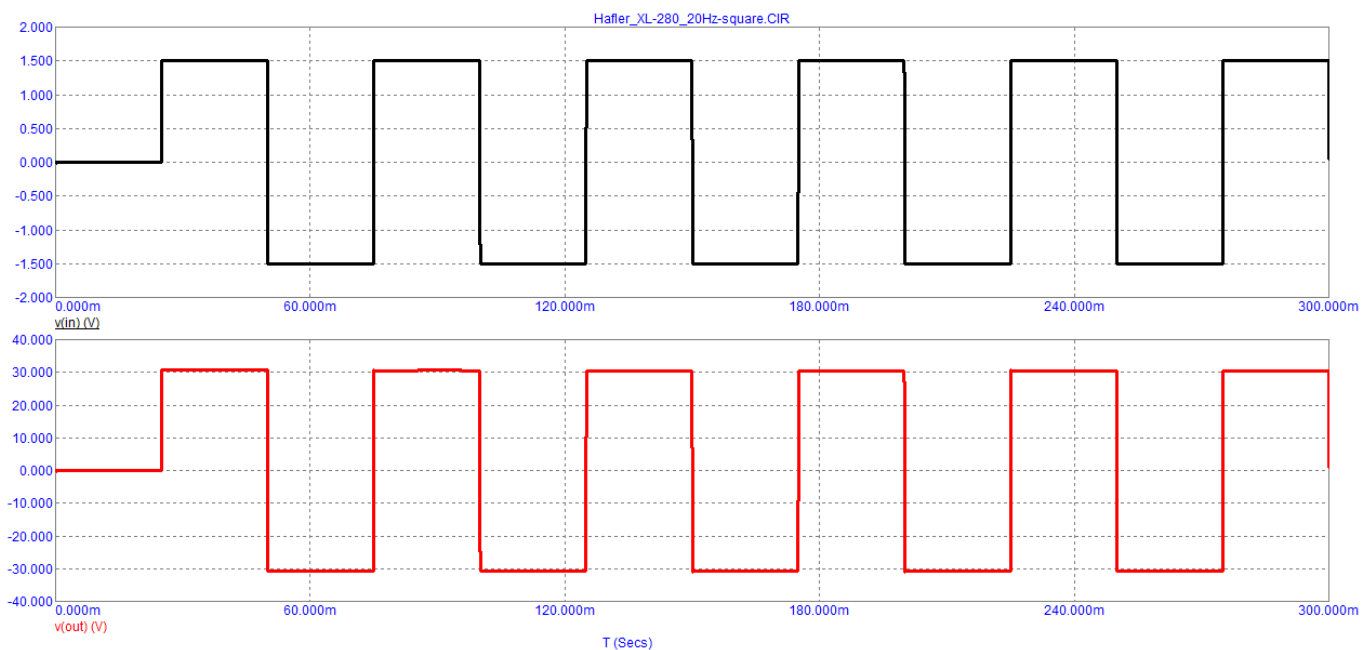
Let's check how the signal with a frequency of 20 Hz is amplified, fig. 12



Rice. 12

The burst test showed that an offset of up to 100 mV appears in the signal, which depends on the polarity of the first half-cycle. According to the "gold-eared" offset  $\pm 25 \text{ mV}$  is already audible and adversely affects the sound quality. It is possible that the Doppler effect manifests itself to some extent (it can be calculated, of course, but laziness) due to the additional modulation of the MF-HF components of the signal.

Let's check the amplification quality of a rectangular signal with a frequency of 20 Hz, fig. thirteen.

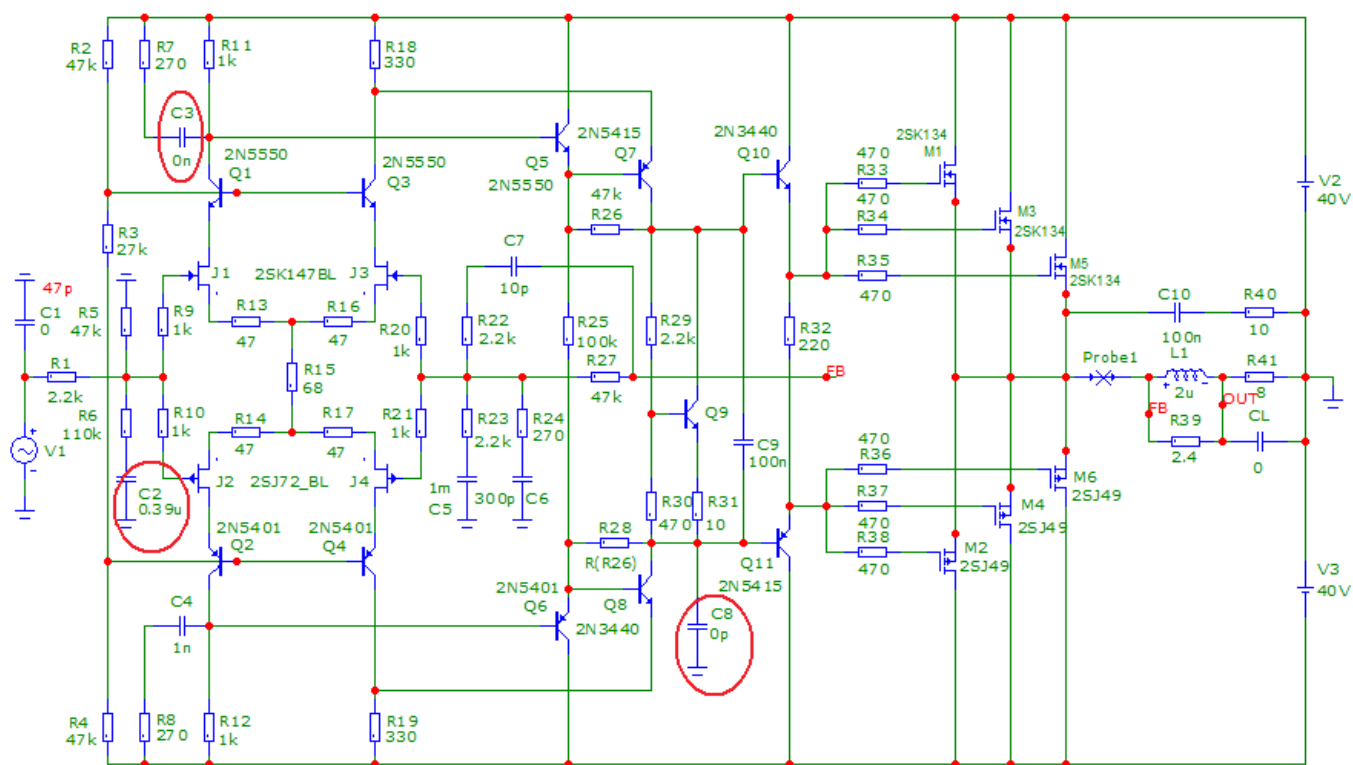


Rice. thirteen.

Figure 13 shows that the signal shelves are horizontal, i.e. there are no pronounced distortions. And this is not surprising, because this is a direct current amplifier (UCT).

*Note. Before applying the signal to the input, it is passed through a low-pass filter with a cutoff frequency of 100 kHz.*

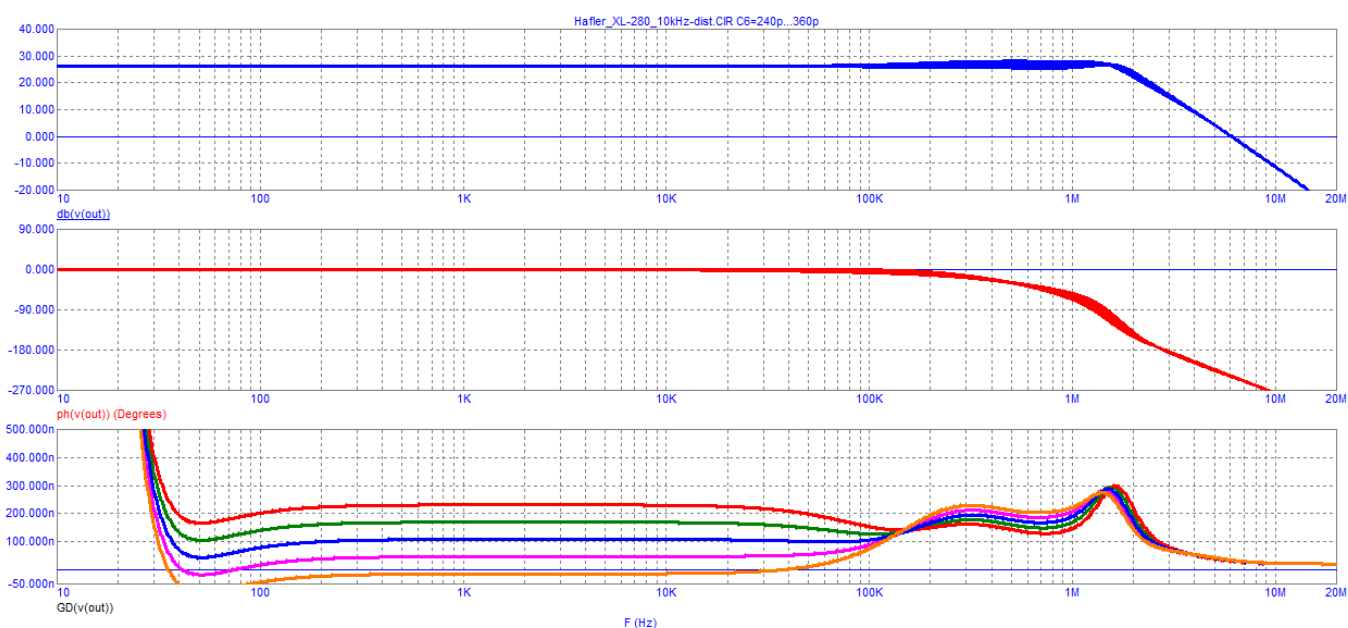
Let's try to refine the model in such a way as to reduce the group delay surge responsible for high-speed distortions, Fig. 14



Rice. 14. Scheme of the modified model

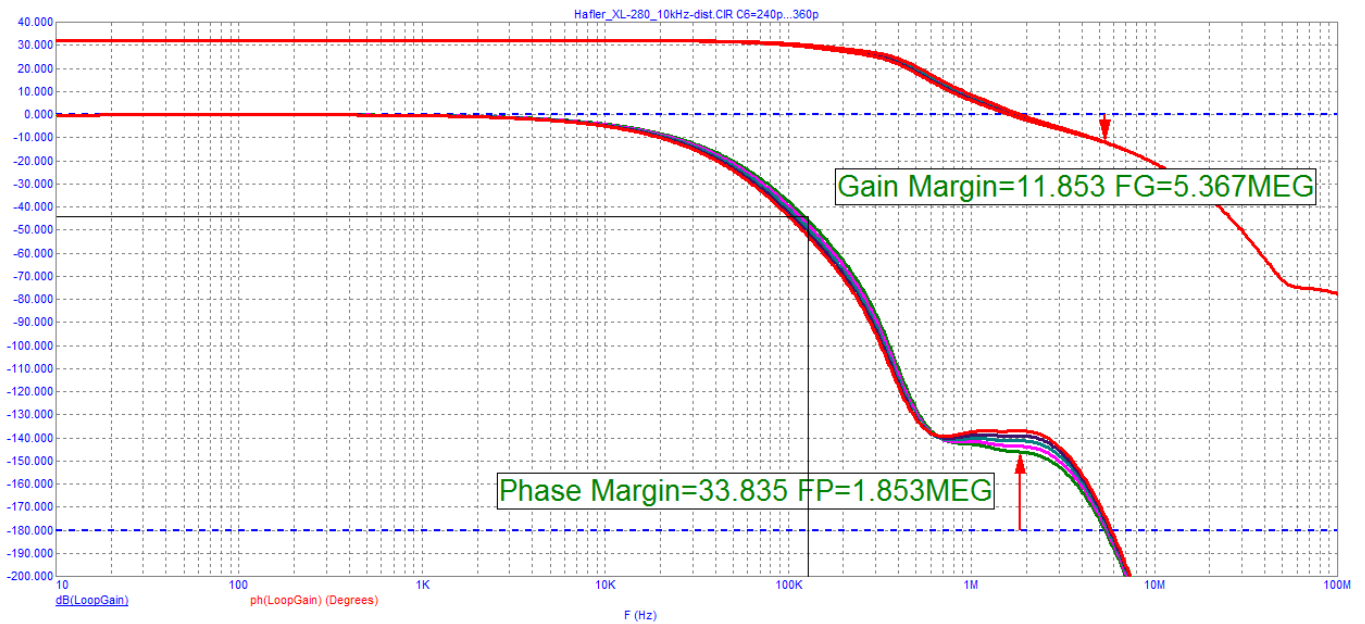
In order to expand the horizontal section of the group delay in the LF region, the capacitance of the capacitor C2 was increased. Removing correction capacitors C3, C4 and C8 reduced the overshoot at 450 kHz from 1.5  $\mu$ s to 300 ns and shifted it to 1.5 MHz, fig. 15, and also increased the frequency of the first pole from 20 kHz to 120 kHz (6 times). At the same time, the modes of the cascades for direct current remained the same.

In fact, capacitor C8 is not needed, since the input capacitance of the paired output transistors is converted from the output of the buffer stage (Q10, Q11) to its input.



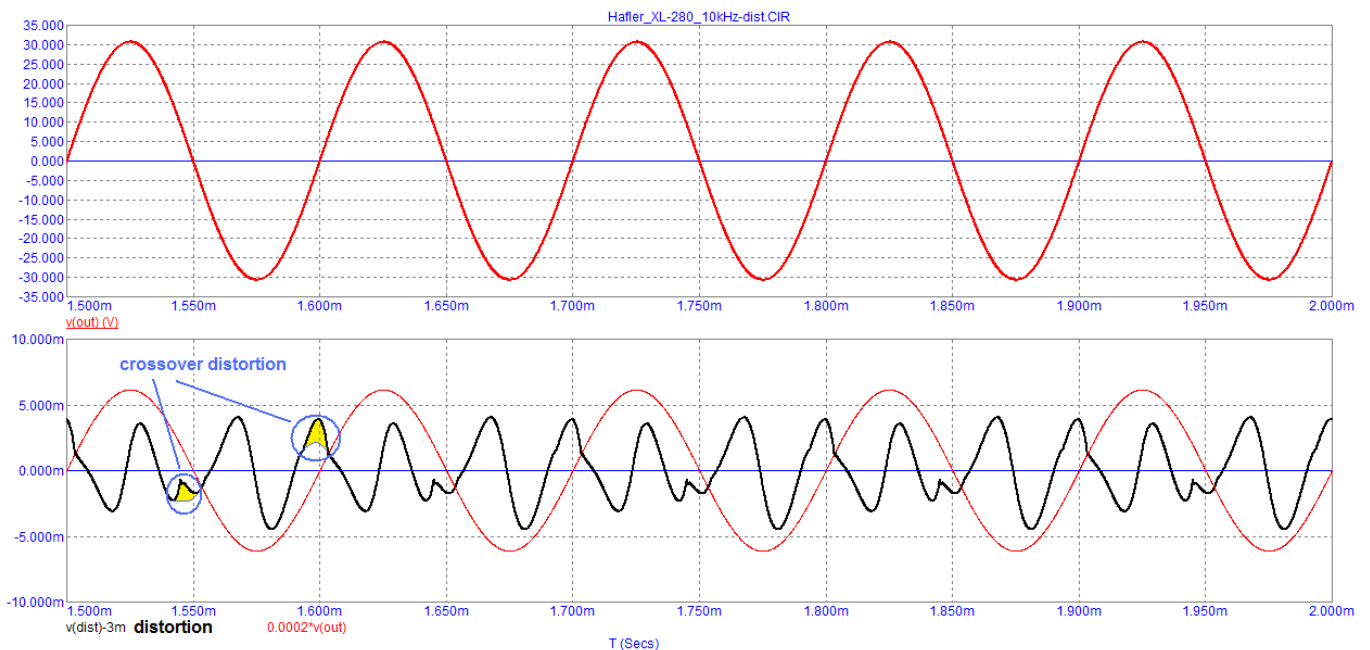
Rice. 15.

Let's check the loop gain and stability margins, Fig. 16



Rice. Fig. 16. Loop gain graph for all possible destabilizing factors (SHG tuning and reactive load from 50 nF to 2  $\mu$ F).

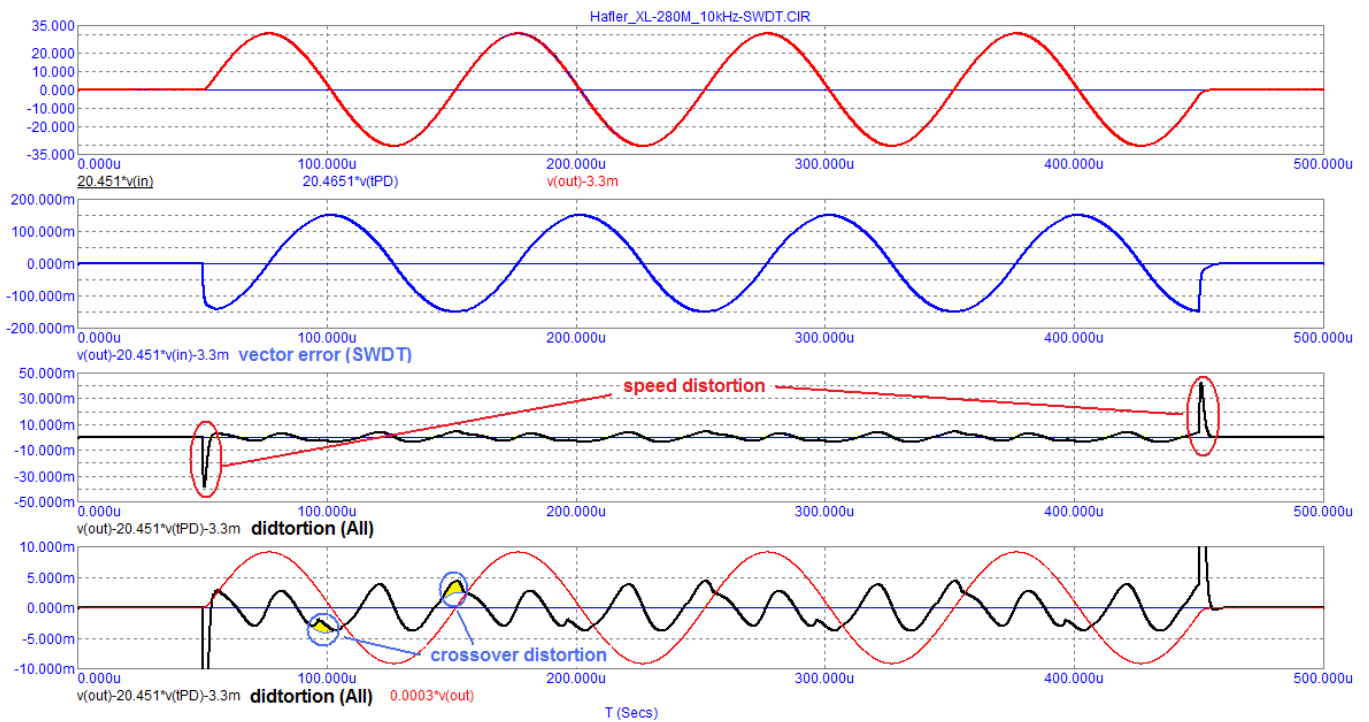
Let's use a notch filter and measure the distortion at a frequency of 10 kHz, fig. 17.



Rice. 17. The result of measuring distortion with a notch filter

As follows from the test, the spectrum of lower harmonics and its level remained the same, mainly the 3rd harmonic at a level of about 4 mV (peak). However, the level of switching distortions has significantly decreased and their spectrum has improved, the distortions have become more "smooth" due to the increased frequency of the first pole.

Let's carry out the Hafler test and measure the distortion by the compensation method, fig. eighteen



Rice. 18. The result of measurements using the Hafler test (third graph) and the compensation method (fourth graph)

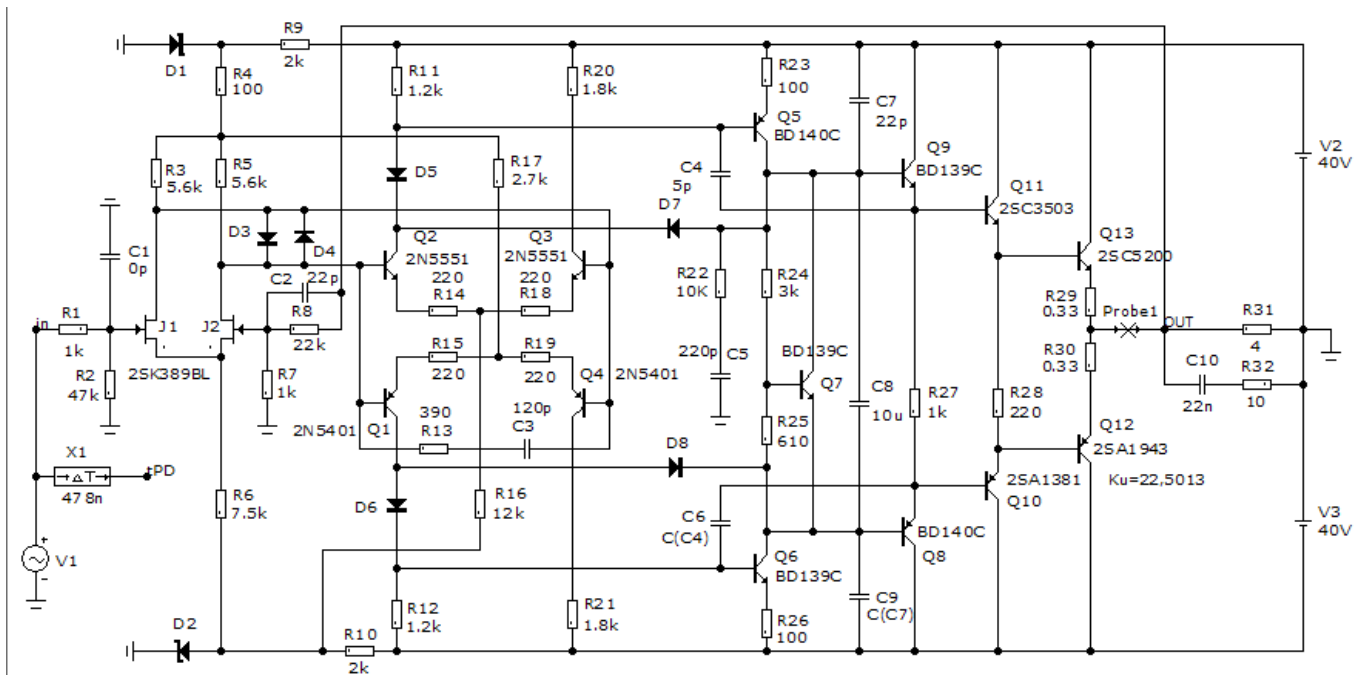
Compared to the test of the original circuit, the group delay increased from 13 ns to 80 ns by more than 6 times, hence the vector errors increased by the same factor (up to 150 mV (peak)). At the same time, high-speed distortion decreased from 250 mV to 40 mV (6 times). In the steady state, the distortions are the same as those measured with a notch filter).

Switching (crossover) distortions in many class AB amplifiers are well audible, it is not in vain that many prefer class A amplifiers. And if you consider that in a real musical signal, zero crossings more often coincide with low-frequency signals (occur relatively rarely), and high-frequency components and their harmonics are much change  $dV / dt$  more often (on which high-speed distortions depend), then I hope it becomes clear what the sound quality actually depends on.

**Alexander Petrov**

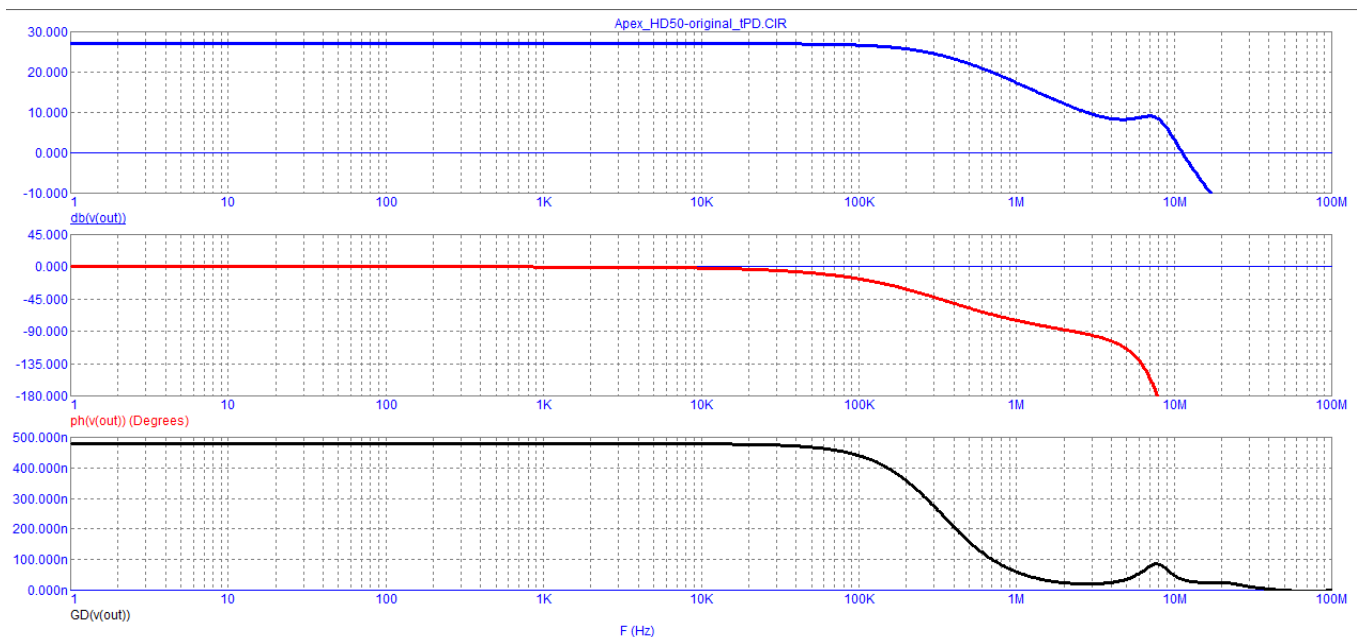
## Appendix 2

Let's check the influence of signal propagation delay time ( $t_{PD}$ ) on real amplifier models. As such an amplifier, let's take Apex HD50, fig.22.



Rice. 22

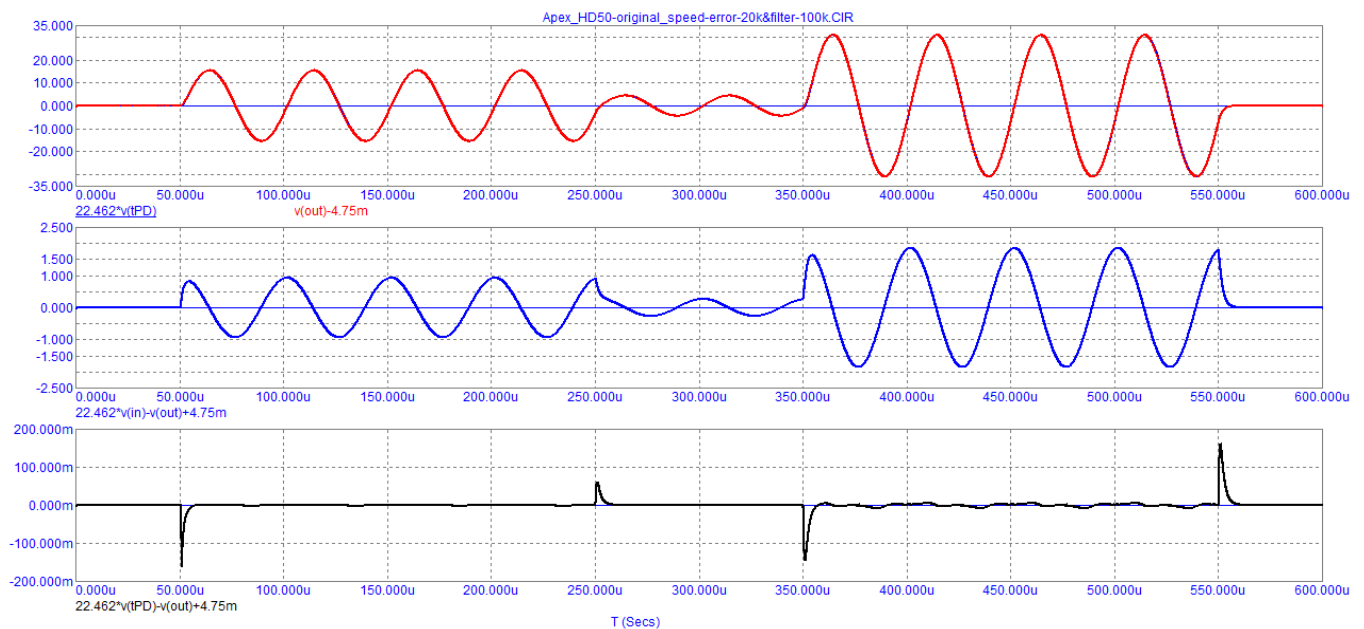
The bode diagram is shown in fig. 23



Rice. 23

From the Bode diagram, you can see that  $t_{PD}$  is almost 500 ns (478 to be exact).

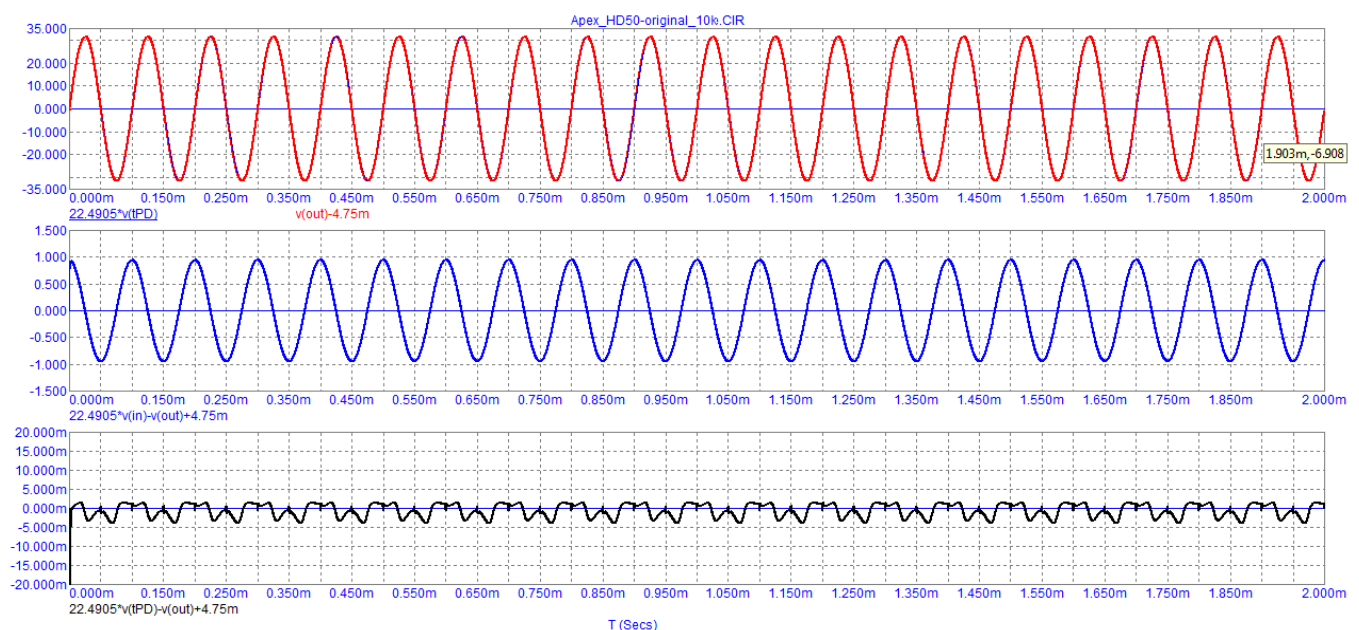
Let's apply to the input of the amplifier bursts with a frequency of 20 kHz of different amplitudes and following one after another without a phase break. In this case, the output signal of the generator will be passed through a low-pass filter with a cutoff frequency of 100 kHz, Fig. 24



Rice. 24

At the beginning of the first burst of 15 V (peak), we see short-term high-speed distortion with an amplitude of more than 150 mV, which is 1% of the amplitude of 15 V. This level of high-speed distortion also appeared at the end of the last burst (0.5%).

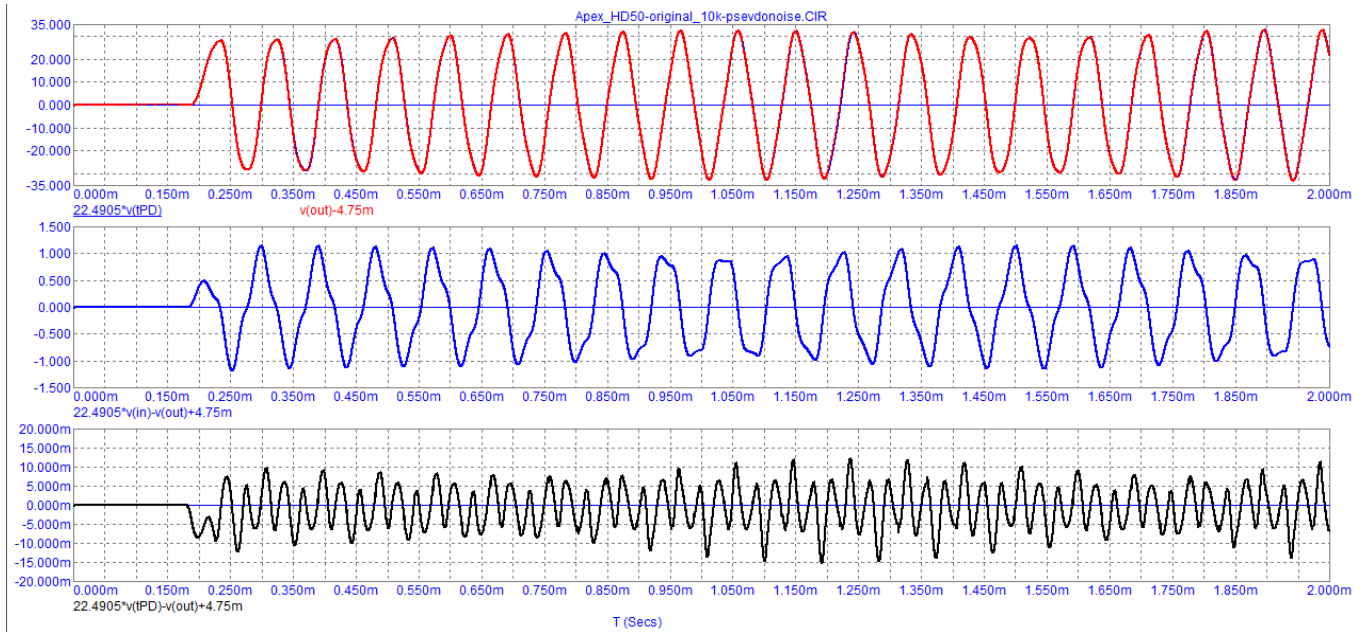
To check how the amplifier will behave with a signal close to the sound, let's apply 1/3 octave pseudo noise with a frequency of 10 kHz to the input of the amplifier. Let us preliminarily set up the model (we carefully select the coefficient Ku and specify tPD) on a sinusoid with a frequency of 10 kHz, Fig. 25



Rice. 25

The result of measuring distortion by directly subtracting the output signal from the delayed input signal gave a level of low order harmonics with an amplitude of up to 3 mV (0.01%) and a small level of crossover distortion (opposite to zero crossings).

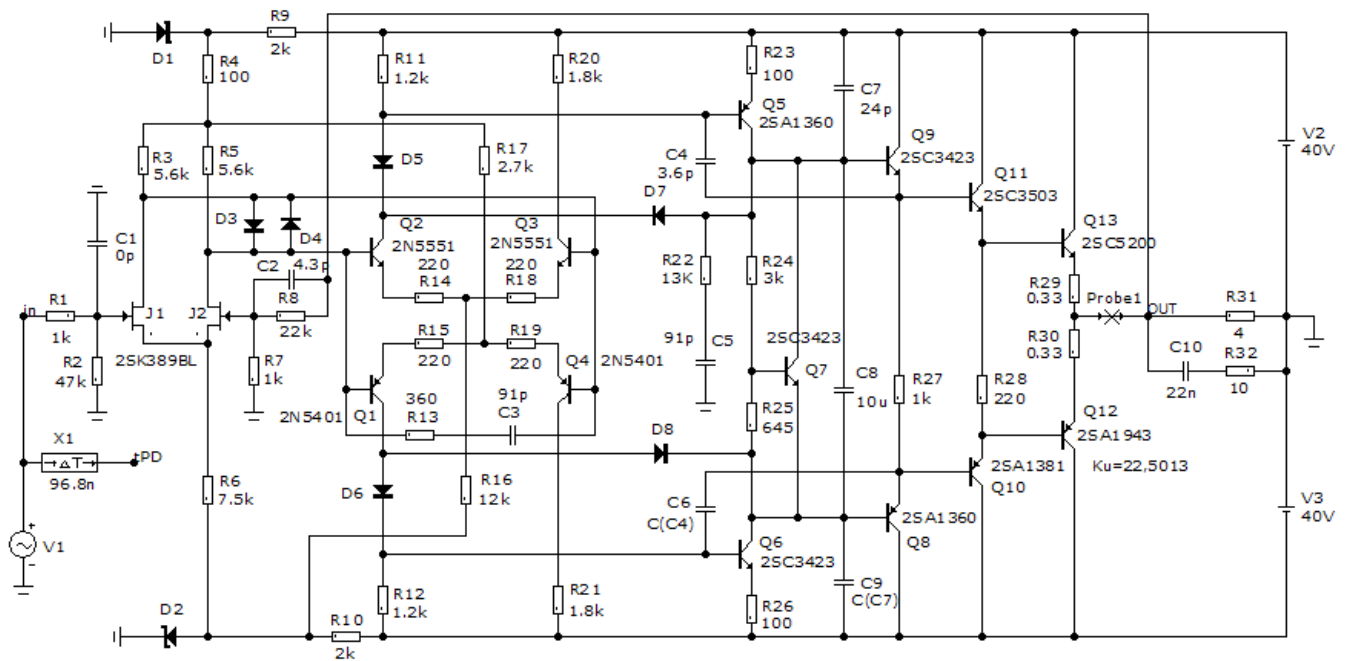
Without changing the settings, we will change the sinusoid to pseudonoise. We use pseudonoise in the form of a wav file, which, before being fed to the input of the model, is passed through a 4th order Bessel low-pass filter with a cutoff frequency of 30 kHz, Fig. 26



Rice. 26

The level of distortion products at pseudo-noise increased by about 5 times, i.e., up to 0.05%.

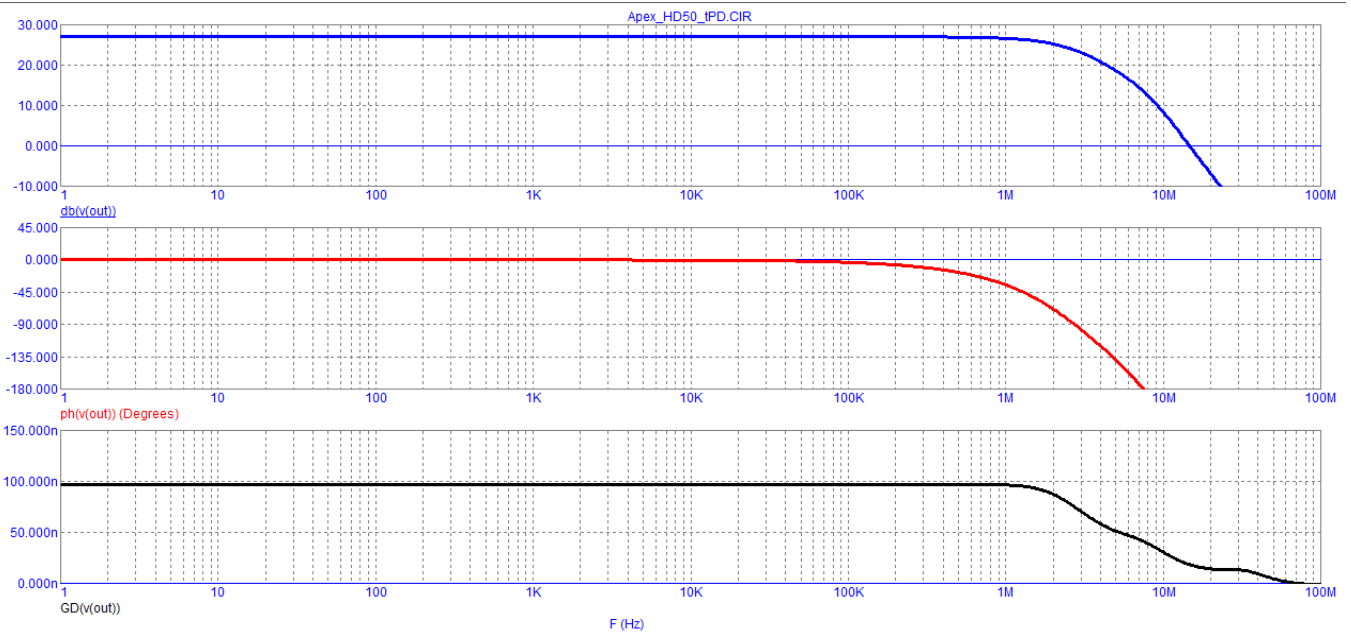
And now, without changing the modes (the quiescent current of the output transistors is 110 mA), we will finalize the model in order to reduce tPD, fig. 27



Rice. 27

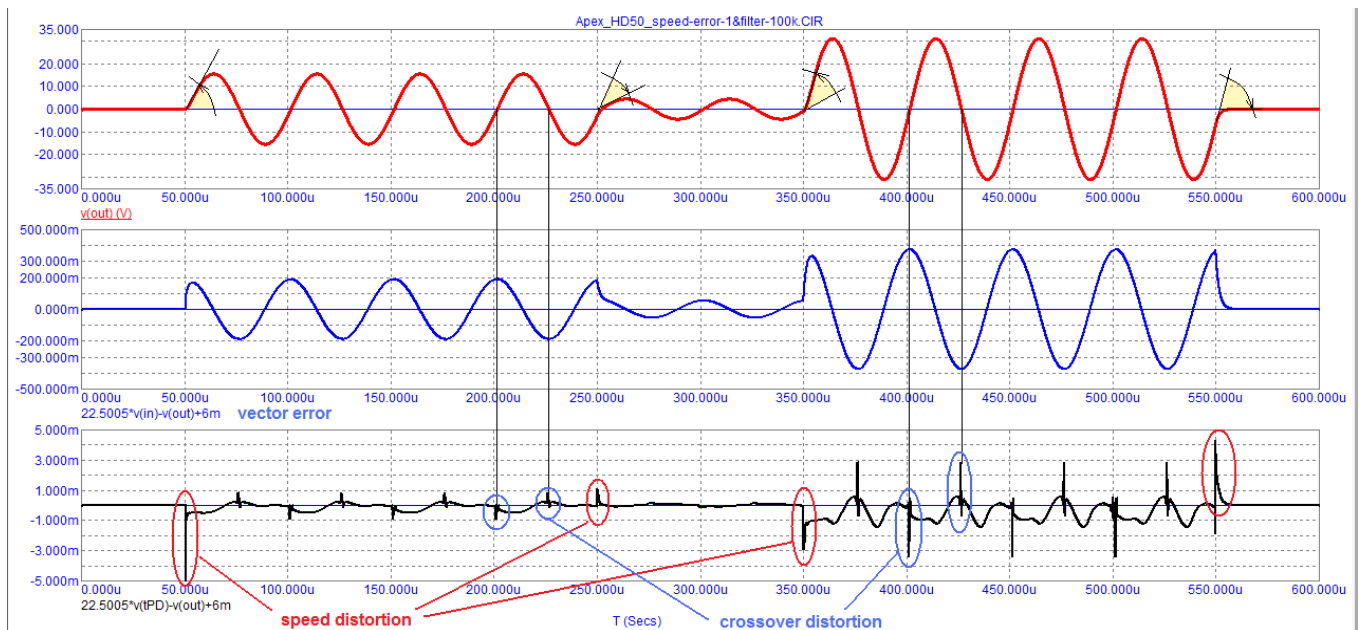
The Bode diagram of the modified version is shown in Figure 28





Rice. 28

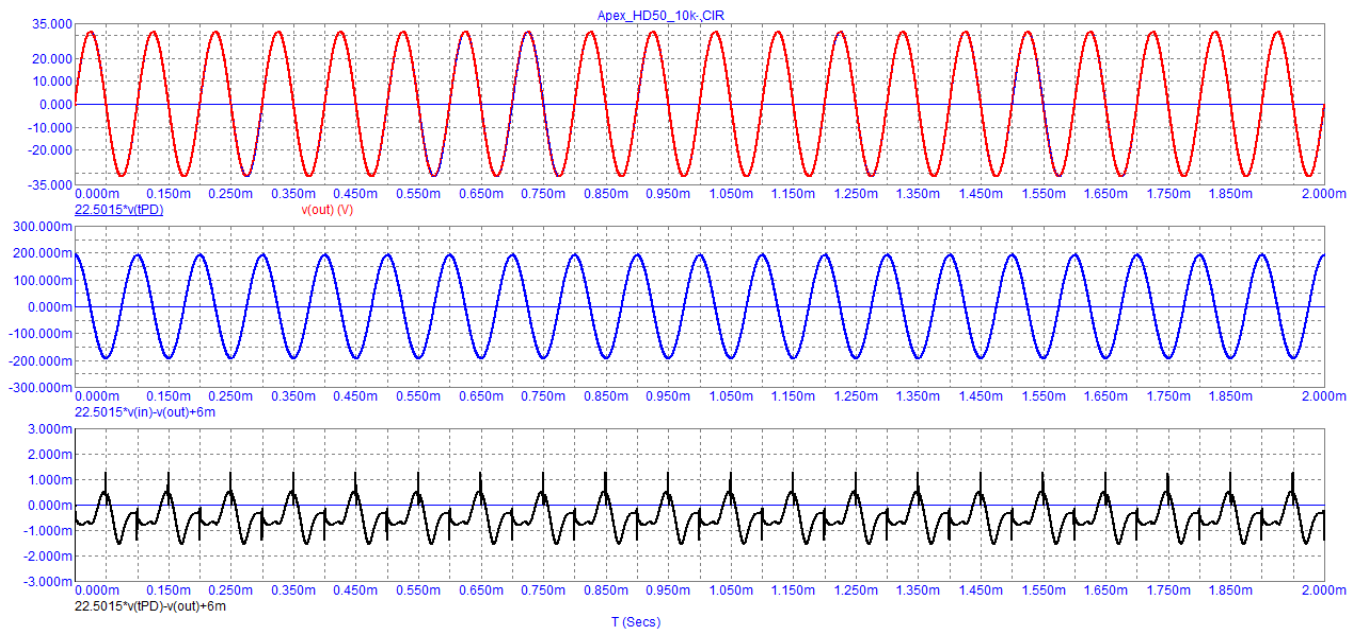
From the Bode diagram, it can be seen that tPD has decreased by almost 5 times and has become slightly less than 100 ns, and linearly more than 1 MHz. Let's do the same tests with the modified model. The measurement of high-speed distortions on bursts with a frequency of 20 kHz is shown in fig. 29



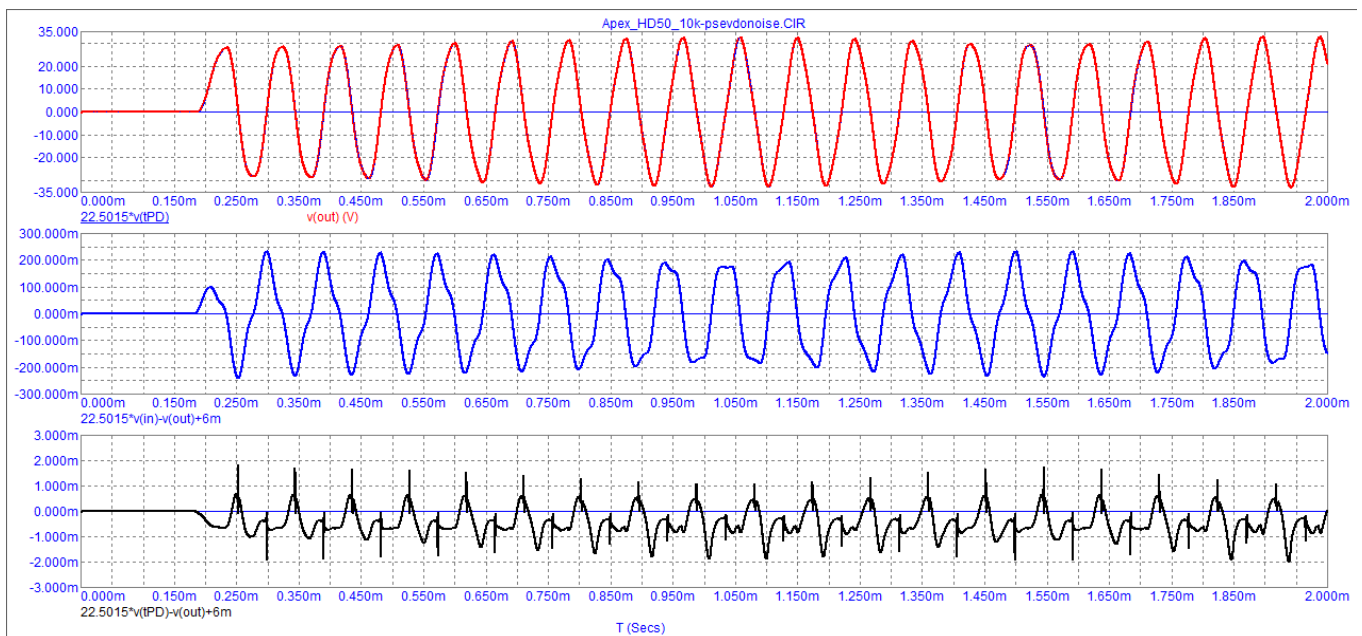
Rice. 29

It can be seen from the test result that the amplitude of high-speed distortion at the beginning and end of bursts decreased from 150 mV to less than 5 mV (more than 30 times), the level of harmonic components also decreased, and commutation distortion came to the fore, which almost reach the level of high-speed distortion. In real signals, of course, there will not be such a level of signals with a frequency of 20 kHz, therefore, the actually introduced switching distortions will be significantly lower.

Let's do similar tests on pseudonoise with a frequency of 10 kHz, fig. 30 and 31



Rice. 30 Vector errors and distortions of the model obtained by direct subtraction on a sinusoid with a frequency of 10 kHz



Rice. 31 Vector errors and distortions of the model obtained by direct subtraction on 1/3 octave pseudonoise with a frequency of 10 kHz.

If you look closely at the third graphs of both tests, you can easily see that they are almost identical. This suggests that with a signal propagation delay of 100 ns or less (provided that the group delay is linear and does not have significant spikes in the 0.2 ... 1 MHz band), high-speed distortions practically do not appear.

Alexander Petrov