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THE ACCURACY OF REAL-TIME COMPUTER SPECTRUM ANALYZER CALCULATIONS

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A practical estimate of the accuracy of calculations of a software-hardware system consisting of an audio-frequency generator and a combined spectrum analyzer is presented. It is shown, using the example of a number of model measurements, that the frequency, amplitude, and phase errors of the calculations are very close to the theoretical limits.
Key words*: spectrum analyzer, software-hardware measuring system.*

In this paper, we give a practical estimate of the accuracy of calculations of a software-hardware system consisting of an audio-frequency generator, a spectrum analyzer, an oscilloscope, an instrument for measuring noise and nonlinear distortions, a frequency meter, and a phasemeter, designed to investigate the characteristics of electrical four-pole networks and to model the processes occurring in them. The computer measuring instruments in the system take the form of two independent sections. The first consists of a multifunctional audio-frequency generator. The second part, called the Oscillometer, is a combined instrument, consisting of a spectrum analyzer, an oscilloscope, a frequency meter, dc and ac voltmeters, and instrument for measuring noise and nonlinear distortions, power, phase difference, and the amplitude distribution density of the input signal.

The two-channel multitone audio-frequency signal generator [1] includes an original SoundGen controlling program and a digital-to-analog converter. Two-channel 16-, 24- or 32-bit digital-to-analog converters of the standard audio system of the computer (the MME – MultiMedia Extension interface) with a sampling frequency F_s of up to 384 kHz are used as the converters. At the present time, two-channel 24-bit devices with F_s = 200 kHz, for example, the Juli@ audio-interface made by Ego Systems Inc. or the LynxTWO-B made by Lynx Studio Technology are available commercially. If necessary, an additional adaptation of the controlling program to other types of digital-to-analog converter, having their own buffer memory, is possible. The program-generated frequencies cover the range from 0.001 Hz to $F_s/2$.

The Oscillometer [2] consists structurally of the original OscilloMeter controlling program and analog-to-digital converters. In the basic version of the instrument, two-channel 16-, 24-, or 32-bit analog-to-digital converters of the standard audio system of the computer with a sampling frequency F_s up to 384 kHz, for example, the above-mentioned audiointerface, are used. Here the upper limit of the operating frequency band of the oscilloscope and the spectrum analyzer reaches values of $F_s/2$. If necessary, additional adaptation of the controlling program to other types of analog-to-digital converters possessing their own buffer memory is possible.

The SoundGen and OscilloMeter controlling programs are designed to operate in any IBM-PC or compatible computer in the Windows 95/98 NT/2000/XP operating system.

For a considerable number of the parameters measured by the combined computer spectrum analyzer, there are considerable differences in the definitions given in the literature. We will give definitions of these parameters in accordance with the algorithms for calculating them:

• *SNR* (Signal to Noise Ratio) is the ratio of the signal power at the fundamental frequency to the noise power;

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• *SINAD* (Signal to Noise plus Distortion Ratio) is the ratio of the signal power at the fundamental frequency to the overall power of the noise and harmonics;

• *SFDR* (Spurious-Free Dynamic Range) is the dynamic range as a ratio of the amplitudes of the fundamental signal and of the greatest secondary signal (noise or harmonic) with the exception of the constant component;

• *ENOB* (Effective Number of Bits) is the effective number of bits or the actual number of digits of the analog-to-digital converter, which takes into account the noise and the distortion

$$ENOB = (SINAD - 1.76)/6.02, \tag{1}$$

where *SINAD* is in decibels;

• *THD* (Total Harmonic Distortion) is the total harmonic distortions expressed as a percentage or in decibels and is equal to the square root of the ratio of the overall power of the harmonics to the power of the fundamental frequency:

$$THD = \sqrt{\frac{1}{P_{f_1}} \sum_{i=2}^{h} P_{if_1}}, \quad 0 < if_1 < f_N, \tag{2}$$

where P_{xx} is the power of the corresponding harmonic, i is the number of the harmonic, h is the maximum number of the harmonic which must be taken into account (specified in the measurements), and $f_N = F_s/2$ is the Kotel'nikov–Nyquist criterion;

• *IMD* (Inter-Modulation Distortion) is the coefficient of inter-modulation distortions for a two-tone signal with frequencies f_1 and f_2, equal to the square root of the ratio of the total power of the combination frequencies of the two greatest (fundamental) signals to the total power of these fundamental signals and expressed as a percentage or in decibels:

$$IMD = \sqrt{\frac{1}{P_{f_1} + P_{f_2}} \sum_{i=1}^{h} \sum_{j=1}^{h} P_{|if_1 \pm jf_2|}}, \quad 0 < |if_1 \pm jf_2| < f_N, \quad |if_1 - jf_2| \neq f_1, f_2, \tag{3}$$

where P_{xx} is the power of the signal of the corresponding combination frequency, i and j are the numbers of the harmonics, and h is the maximum number of the harmonic which must be taken into account (specified during the measurements).

It is of considerable interest to estimate the reliability and quality of calculations in the spectrum analyzer and of the measuring instruments combined with it. The determination of the degree of spectral purity of signals synthesized in the audio-frequency generator is also of considerable importance. For these purposes, the possibility is provided of feeding a signal, synthesized in the audio-frequency generator, directly into the OscilloMeter program, bypassing the digital-to-analog converter and the analog-to-digital converter, via a special driver (www.nrcde.ru/music/software/eng/vac.html). This approach enables one to model certain nonlinear processes, which occur in four-pole networks. Below we consider a number of examples which, if desired, can easily be reproduced. The adjustments of the spectrum analyzer (unless otherwise stated) are as follows: the regime for calculating the individual spectra of the channels, the number of readouts of the fast Fourier transformation unit 2^{16}, the Cos 8 min smoothing window [3], the signal sampling frequency F_s = 44.1 kHz, and the remaining parameters of the measurements are left untouched.

We will present the results of measurements carried out using the computer analyzer proposed above.

1. *A check of the amplitude linearity of the spectrum analyzer when simulating an ideal analog-to-digital converter.* A sinusoidal signal of frequency 1 kHz is applied from the generator. For 16 bit and 24 bit digitization, we obtain values of the *SINAD* of 98 dB and 146 dB respectively. This corresponds to the theoretical limit due to quantization noise. The theoretical limit of the *SINAD* can be calculated from (1) if we put the *ENOB* equal to the formal digitization. For a 32 bit digitization, the measured value *SINAD* = 190 dB, which gives an effective number of bits of 31.3. The losses are obviously due to rounding errors in the audio-frequency generator, where a 32-digit arithmetic is used when synthesizing the signals.

2. *A check of the amplitude linearity of the spectrum analyzer when simulating an ideal analog-to-digital converter using a composite signal.* A composite signal: 8.02 kHz (–14 dB) and 250 Hz (–2 dB), for measuring intermodulation dis-

tortions, is applied from the generator. For a 16-bit resolution, the measured value of the *IMD* = 0.00014%, while for 24-bit and 32-bit resolutions it is less than 10^{-6}% (the resolving threshold of the measuring instrument).

3. *Check of the instrument for measuring harmonic distortions (THD).* The following mixture is applied from the generator: a fundamental frequency of 1 kHz (–2 dB) and its harmonics from the second to the eighth. The amplitude of the harmonics falls off linearly from –60 dB to –90 dB as the number increases. For formal 16 bit digitization, we obtain *SNR* = 96 dB, *SINAD* = 56 dB, an effective number of bits of 9, and overall harmonic distortions amount to 0.15%. These results for the *SINAD* and the *THD* correspond completely to those calculated for this signal from formula (2). The value of the *SNR* corresponds to the signal at the fundamental frequency, digitized by an ideal analog-to-digital converter.

4. *A check on the instrument for measuring harmonic distortions (THD) when simulating an interfering signal.* A mixture of frequencies of 100 kHz and 3.01 kHz with amplitudes of –6 dB and –66 dB respectively is applied from the generator. For a digitization of 16 bits, *SNR* = 60 dB, *SINAD* = 60 dB, and the effective number of bits is equal to 9.6, and there are no overall harmonic distortions (0%). It is obvious that the *SNR* and *SINAD* are exactly equal to the ratio of the signals applied, which they should also be in the case of ideal digital-to-analog and analog-to-digital converters.

5. *A check of the instrument for measuring intermodulation distortions (IMD).* A mixture is applied from the generator, namely, two fundamental frequencies of 8.02 kHz (–14 dB) and 250 Hz (–2 dB) and their combination frequencies (sum-difference frequencies of the first-third orders): 7.27 kHz (–90 dB), 7.52 kHz (–80 dB), 7.77 kHz (–70 dB), 8.27 kHz (–70 dB), 8.52 kHz (–80 dB), and 8.77 kHz (–90 dB). The measured value of the coefficient of intermodulation distortions was 0.057%, which corresponds to the result of a calculation using formula (3) for this composite signal.

6. *Measurement of the instability (jitter) of the clock generator, that is short-term compared with the measurement time.* The digitization is 16 bit. A sweep frequency with a mean value of 999.9995 Hz and a deviation of $\pm 5 \cdot 10^{-4}$ Hz and a sweep period of 20 msec is applied from the generator. The frequency modulation of the reference generator of the analog-to-digital converter is thereby simulated by interference from the power network. The equivalent relative frequency “instability” is then $5 \cdot 10^{-7}$. The jitter amplitude, referred to the sampling frequency of 44.1 kHz, therefore corresponds to 11 psec. The spectrum analyzer is switched to a “self-synchronized” signal repetition mode, by which we mean the digital product of the measured signal and its fundamental harmonic [2]. A similar operation is carried out in the synchronous detectors, henceforth similarly named. However, unlike detection, here the individual frequency components are not filtered, and the complete spectrum of the product of the signals obtained is calculated and displayed. In these measurements, a classical Hanning smoothing window [4] is the most effective. Moreover, one must include linear operation, for example, over five samples, averaging the results of the fast Fourier transform. For a 16-bit signal digitization, one can clearly observe the “parasitic” component of the frequency of 50 Hz and of amplitude –110 dB, and a dynamic range of 104 dB. Hence, this mode of measurement enables one to determine an extremely small short-term instability (jitter) of the clock generator of the analog-to-digital converter. Moreover, in this mode of operation one can obtain the source of the corresponding frequency fluctuations of the reference generator of the analog-digital converter.

7. Check of the built-in frequency meter. A sinusoidal signal with a frequency of 99.99999 Hz is applied from the generator. For the most accurate measurements of the frequency, Hanning [4], Rife-Vincent [5] windows or a single window (i.e., no window) are used. The dimensions of the fast Fourier transform unit is 4096 readouts of the input signal or more; consequently, the fast Fourier transform unit includes not less than ten periods of the signal. The actual error in measuring the frequency in this case does not exceed a few lower digits of the indicator, i.e., it amounts to $\pm 1 \cdot 10^{-7}$ in relative units. This high accuracy when using a discrete Fourier transformation is achieved by using interpolation of the results of the fast Fourier transform [6].

8. *A check of the built-in instrument for measuring the amplitude and power of the input signal.* The signal was a 24-bit signal. A sinusoidal signal of fixed frequency in the range 5 Hz – 20 kHz with different levels from –100 dB to 0 dB was applied from the generator. For the most accurate measurement of the amplitude and power, one must use a Flat-top 5A type smoothing window from [7]. The dimensions of the fast Fourier transform unit is 2^{16} readouts. For any frequencies and levels from these ranges, the deviation of the measured amplitudes and power of the input signal from those specified in the generator-oscillation source does not exceed ±0.002 dB.

9. *A check of the built-in instrument for measuring phase difference.* A sinusoidal signal with a frequency of 5 Hz – 20 kHz with different levels from –60 dB to 0 dB is applied from the generator. The phase shift between the chan-

nels of the generator is set arbitrarily in the range from –180° to 180°. We used Cos 8 min [3] and Rife-Vincent [5] smoothing windows. For any phase shifts, frequency and level in the ranges indicated the readings of the phase measuring instrument differ from the values set in the generator by not more than ±0.0001°.

The moduli of the errors of the calculation of the main measured parameters in the examples considered do not exceed the following values [2]:

frequency (of the measured value)	$5 \cdot 10^{-8} – 5 \cdot 10^{-7}$
amplitude	0.02 dB
power	0.01 dB
dynamic range of the signal/noise ratio; signal/(noise + distortion) ratio	0.05 dB
harmonic and intermodulation distortions (of the measured value)	1–5%
phase shift between channels	0.0001°

It must once again be emphasized that the examples given above are intended for estimating the accuracy of the calculation of these parameters using the corresponding algorithms. The actual accuracy of the computer instruments is limited by the quality (digitization, speed of response, linearity, and noise level) of the analog-to-digital and digital-to-analog converters employed.

The results obtained characterize the accuracy of the audiofrequency generator and combined spectrum analyzer as being very close to the theoretical limits. This enables the program to be used as a basis for constructing standard computer measuring instruments for a frequency band, limited solely by the speed of response of the analog-to-digital and digital-to-analog converters employed.

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