METHOD FOR DISTORTION MEASUREMENTS USING PSEUDO-RANDOM

SIGNAL

PACS reference: 43.58.Ry

Djurek, Ivan; Somek, Branko; Maletic, Mladen

Faculty of EE and Computing, Dept. of electroacoustics

Unska 3

HR-1000 Zagreb, Croatia Tel.: +38516129833

Fax.: +38516129680

E-mail: ivan.djurek@fer.hr

ABSTRACT

Distortion measurements should give us objective evaluation and comparison of different

electroacoustical devices. Modern methods for distortion measurements give us partial

information about certain devices, because used test signals don't have characteristics of

natural signals - speech and music. Proposed combined audio test signal (CATS) consists of

three sawtooth signals, and has characteristics of natural signals. Frequency components of the

CATS spectrum are unevenly distributed, so intermodulation components fall between

components of the original spectrum at the input of the measured system. Using peak search

method intermodulation components can be extracted form original signal, and their comparison

can be made.

INTRODUCTION

Objective measurements of acoustical and electroacoustical systems are very important part of

this field of electrical engineering. Conventional methods give us only partial information about

these systems because of the characteristics of the test signals, which don't have

characteristics of natural signals. Goal is to make a signal, which is repeatable, easily controlled

and it is similar to the natural signals – speech and music. Proposed combined audio test signal basically consists of three sawtooth signals. Its probability density function is bell-like, peak factor is between 4-5, and the normalized rate of change is not grater then 75 mV/µs/V_P. Measurements with MLS signals are in the moment very popular ways of measuring electroacoustical systems. Compared to periodic-impulse excitation (PIE), with similar impulse repetition frequency, they have much larger excitation energy, for the same peak value at the output. MLS signals also have much higher noise immunity. Generally, for measuring LTI systems we can use every synthesized test signal, that is repeatable and that has wide spectrum. This signal must have energy at all frequencies of measured LTI system. Spectrum of CATS is wide, and if it's applied into nonlinear system, distortion components fall between original spectrum components. In order to avoid distortion components to fall into original spectrum, equal spectrum shift must be applied.

SYNTHESIS OF THE SIGNAL

Basic part of the composed audio test signal is sawtooth signal, with random varying rising times. The signal is synthesized in such a way that is not possible that two saws have same rising times. It that way there is not repetition of certain harmonics in signal. Probability density function of one sawtooth signal is rectangular. In order to get bell-like distribution, three synchronized sawtooth signals are combined. Second sawtooth signal is made with combining duration of two saws in first sawtooth signal. If first signal has 12 saws, then second would have 6. Duration of both sawtooth signal sequences is the same. The third sawtooth signal is made with same principle from second one. If second sawtooth signal has 6 saws then third one would have 3. Final composed audio test signal, in time and frequency domain, is shown in figure 1a, and 1b. Probability density function of this signal is bell-like (figure 2), and similar to the probability density function of natural signals, which have Gaussian distribution. Repetition frequency of sawtooth signal sequence determines lower part of the spectrum. In figure 1b first sawtooth signal has 24 saws, second 12, and third 6. Basic combination of three sawtooth signal have duration of approximately 3 milliseconds, which gives the lowest frequency of 300 Hz. When several sequences are combined, and lined up one after another, composed audio

test signal is modulated by the repetition frequency, so that all frequencies in lower part of the spectrum are harmonics of repetition frequency (figure 1b). In this bwer part, the spectrum is not so thick, and difference between two harmonics is determined by repetition frequency. This is basically multitone signal, which can be used for distortion measurements.

This kind of multitone signal can be also obtained with use of standard MLS signal. Problems with MLS signals are that they don't have bell-like amplitude distribution, and distortions are happening only at one amplitude level. With proposed composed audio test signal distortions happen at all amplitude levels. Also, rate of change of CATS is similar to the rate of change of natural signals.

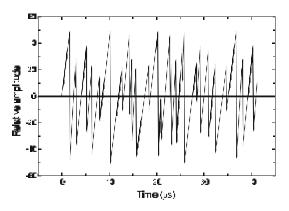


Figure 1a CATS in time domain.

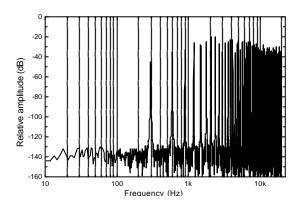


Figure 1b CATS in frequency domain (FFT points: 65536).

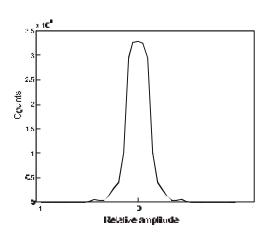


Figure 2 PDF of CATS.

MEASUREMENT METHOD

Distortion generated harmonics form multitone signal fall between harmonics of original signal. If all frequencies in spectrum stay in harmonic relationship, distortion generated new harmonics will fall into original spectrum. That is way spectrum components must be equally shifted for certain frequency. We have done this with use of Matlab. First, DFFT is used, and signal is transformed into frequency domain. In order to have better frequency resolution, number of FFT points is increased to over 100,000. To have enough points for this calculations composed audio test signal lasts over 4 seconds. Spectrum is shifted for certain amount, which can be changed, and transformed back to time domain with IFFT. Spectrum of shifted signal is shown in figure 3. This signal is used for distortion measurements.

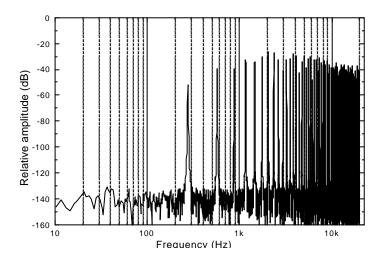


Figure 3 Shifted spectrum of the CATS (20 Hz shift).

Measurement method is similar to the basic total harmonic and intermodulation distortion measurements. Same as in these methods, newly generated harmonics must be extracted from distorted signal, and compared to the original spectrum. This is done with peak search algorithm, which is used for analysis of the DFFT result of distorted signal. To get better frequency resolution, DFFT must be calculated with large number of FFT points. Equation used for distortion calculation is basically the same as for THD measurements:

$$distortion level = \frac{\sqrt{U_{harmonics}^2}}{\sqrt{U_{original}^2 + U_{harmonics}^2}} \cdot 100\%, \tag{1}$$

where $U_{harmonics}$ is level of distortion an intermodulation harmonics, and $U_{original}$ level of harmonics of the original signal without distortion components.

MEASUREMENTS

Examples of distorted composed audio test signal and MLS signals are shown in figure 4a and 4b. Distortion is simulated with third order power series function:

$$f(t) = x(t) - 0.05 \cdot x(t)^3$$
 (2)

Distortion components are marked in figures 4a and b. Distortion level was calculated with use of equation (1). DFFT is done for different number of FFT points. Table 1 shows results of distortion calculations for different number of FFT points.

Table 1 Distortion level calculations for different number of FFT points (CATS signal compared to IMD-SMPTE test signal).

Number of FFT points	Distortion level (CATS)	IMD-SMPTE
4096	0,63%	1,57%
8192	1,14%	1,10%
16384	1,27%	1,19%
65536	0,97%	1,1%

As one can see, number of FFT points is directly connected with frequency resolution, and has big influence on result of distortion level calculations. If the number of FFT points is to low, we can loose accuracy at low frequencies, where intermodulation components occur (Figure 4a).

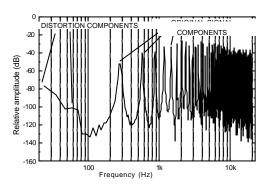


Figure 4a Averaged distorted spectrum of CATS (FFT points: 8192).

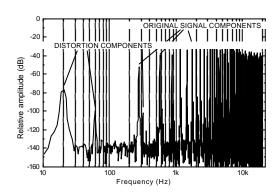


Figure 4b Averaged distorted spectrum of CATS (FFT points: 65536).

CONCLUSION

Composed audio test signal has characteristics of natural signals. The signal is composed from three sawtooth signals, which are randomly generated and synchronized. Its probability density function is bell-like, and similar to the Gaussian distribution. Normalized rate of change is not grater then 75 mV/µs/V_P. Composed audio test signal can be used for distortion measurements, if certain rules are applied. Spectrum frequencies cannot be in harmonic relationship, so spectrum shifting must be applied. To obtain better frequency resolution large number of FFT points should be used when DFFT and IFFT are applied. Better frequency resolution gives more accurate calculations of distortion levels, but demands longer calculation times.

Disadvantage of this method, which uses peak search algorithm to find distortion harmonics is that this algorithm also finds noise signals which are not generated as a result of distortion. This is especially important when measuring loudspeaker distortions in noisy environment.

To obtain better evaluation and analysis of distortion measurement results, extensive subjective test will be carried out. Results of subjective testing compared with distortion level measurements with CATS will give better understanding of distortion measurements. Only after that distortion borders can be determined, which will give better evaluation of audio systems and audio equipment.

REFERENCES:

- [1] E. Czerwinski, A. Voishvillo, S. Alexandrov, and A. Terekhov, Multitone testing of Sound System Components Some Results and Conclusions, Part 1: History and Theory, Journal of The Audio Engineering Society, Vol. 49, Nr. 11, November 2001, 1011-1048
- [2] E. Czerwinski, A. Voishvillo, S. Alexandrov, and A. Terekhov, Multitone testing of Sound System Components Some Results and Conclusions, Part 2: Modeling and Application, Journal of The Audio Engineering Society, Vol. 49, Nr. 12, December 2001, 1181-1192
- [3] MacWilliams, F.J., Sloane, N., Pseudo-Random Sequences and Arrays, Proc. of the IEEE, Vol. 64., Pages 1715-1728, No. 12, December 1976
- [4] M. Maletic, H. Domitrovic, I. Djurek, Frequency Response Measurement with Composed Audio Test Signal, Proc. of 137th Meeting of the ASA and the 2nd Convention of the EAA: Forum Acusticum Integrating the 25th German Acoustics Daga Conference, Berlin, 5PAA-2, 1999