ICSV14
Cairns • Australia
9-12 July, 2007

14th International Congress on Sound & Vibration

ANALYSIS AND PERFORMANCE MONITORING OF ELECTRO ACOUSTIC SYSTEMS

Frantisek Kadlec

Faculty of Electrical Engineering, Czech Technical University in Prague
Technicka 2, 166 27 Prague, Czech Republic
kadlec@fel.cvut.cz

Abstract

The analysis of acoustic and electro-acoustic systems includes measurement of their transfer functions, determination of their behavior from the non-linear point of view and impact of external noises on their performance. The analysis and performance monitoring may be performed by exciting the system under test by either harmonic signals, random signals or by impulse methods. For analysis of electro-acoustic systems we concentrated on a method using swept harmonic signals. This method uses driving the system under test by logarithmically swept signal and recording the system response to the test signal. A so-called "inversion filter" is derived from the exciting signal. By convolution of the system response with this inversion filter we obtain the impulse response of the system under test. During a large signal excitation or even overload of electro-acoustic systems, a non-linear distortion may take place. This phenomenon can be analyzed from responses obtained by this method. By appropriate selection of driving signal parameters we can determine the individual distorting components. The method proved especially suitable for monitoring of multichannel systems, where it is possible to observe performance of individual channels as well as their mutual interaction, crosstalk etc.

1. INTRODUCTION

To determine properties of electro-acoustic (*EAC*) and audio systems, i.e. to find their transfer functions including analysis of non-linear distortion, signal to noise ratio etc., it is possible to use harmonic, impulse and noise signals. We can also consider the effect of digital signal processing on the system analysis. *DSP* of audio signals can cause certain specific distortions, such as aliasing and distortion due to quantizing of low level signals. Various methods can be used for analysis of audio, eletromechanical and acoustic systems. Currently, the *MLS* signal method is widely used, however it fails for analysis of systems with non-linearities [1], [2], [3]. Another method used is so-called *MISO* (Multiple Input Single Output) which is suitable for non-linear systems. The idea of *MISO* method is that a general nonlinear model can be replaced by an equivalent multiple input single output linear model, where inputs are nonlinear contributions of the original input signal [4].

Our choice for systems analysis was the method using swept harmonic signals. For design of measuring method, testing and program equipment we used information contained in publications [5], [6], and [7]. In our project we analyzed both the individual components of the electro-acoustic chain, starting with signal pickup, recording and playback, and the total system as a whole. Compression systems, such as *ATRAC, DTS* and others, were tested. It should be noted here that by testing of codecs we mean monitoring of their performance, but not analysis of these systems from the psychoacoustic point of view.

2. DESCRIPTION OF MEASURING METHOD

The swept signal method is based on relation between an input of a linear system signal $x(t)$ and its response on the system output $y(t)$. Signal of the system output can be described as

$$y(t) = h(t) \otimes x(t) \tag{1}$$

where $h(t)$ is the impulse response of the system and $\otimes$ is the symbol for convolution. The description of log-swept signals can be found e.g. in [6], [8]

$$x(t) = \sin\left\{ \frac{2\pi f_1 T}{\ln \frac{f_2}{f_1}} \left[\left(\frac{f_2}{f_1} \right)^{\frac{t}{T}} - 1 \right] \right\}, \quad 0 \le t \le T \tag{2}$$

where T is a sweeping time of the signal in frequency range $f_1 \div f_2$.

A so-called “inversion filter” $f_i(t)$ can be derived from the driving signal. Definition of $f_i(t)$ is obvious from the following expression

$$x(t) \otimes f_i(t) = \delta(t) \tag{3}$$

Using relations (1) and (3), we can get the impulse response $h(t)$ of the system under test

$$h(t) = f_i(t) \otimes y(t) \tag{4}$$

From Equation (4) it can be seen that the impulse response of the tested system $h(t)$ may be obtained as a convolution of the inversion filter $f_i(t)$ with the response of a system under test $y(t)$.

If we further consider an *EAC* system, which under large signal excitation will produce non-linear distortion, we can describe it as a non-linear memoryless system. Such a system can be described by N^{th} order Volterra kernels [6], [9]. By an appropriate selection of test method parameters we can make this method suitable for analysis of *EAC* systems with non-linearities as well. In that case, the distorting components can be separated into partial impulse responses of the system, from which the frequency responses can be obtained by Fourier transform (*FT*).

2.1 Measurement Method Accuracy Test

Before a proper systems measurement, we evaluated the algorithm accuracy. This test was based on generation of test signals with defined distortion, calculation of their impulse responses $h(t)$ using relation (4), their transformation to frequency domain by means of $FT\{h(t)\}$ and analysis of the calculation from numerical accuracy point of view. Artificially distorted test signal $d(t)$ is created as a sum of the fundamental signal $x(t)$ and distorting components $x_d(t)$. Distorting components can be obtained from a formula for $\sin^n(\alpha)$ [10]

$$\sin^{2n-1}(\alpha) = 2^{2-2n}\sum_{k=0}^{n-1}(-1)^{n-k-1}\binom{2n-1}{k}\sin[(2n-2k-1)\alpha] \tag{5}$$

In our case we consider the *7th* harmonic of the fundamental frequency the highest distorting component. Because of our particular application we have limited the calculation to odd harmonic frequencies only. For *3rd*, *5th* and *7th* harmonic component of the signal $\sin(\alpha)$ we can write

$$\sin(3\alpha) = 3\sin(\alpha) - 4\sin^3(\alpha) \tag{6}$$

$$\sin(5\alpha) = -10\,\sin(\alpha) + 5\sin(3\alpha) + 16\sin^5(\alpha) \tag{7}$$

$$\sin(7\alpha) = 35\sin(\alpha) - 21\sin(3\alpha) + 7\sin(5\alpha) - 64\sin^7(\alpha) \tag{8}$$

Using relations (6), (7) and (8) we can define distorted signal, which can be used for testing of measurement method accuracy

$$d(\alpha) = \sin(\alpha) + \sum_i k_i \sin(i\alpha), \qquad i = 3, 5, 7 \tag{9}$$

where k_i are selected distorting coefficients.

Another way is to calculate the highest harmonic component using relation (5). For illustration we used a formula $\sin^5(\alpha)$, which can be modified as

$$\frac{8}{5}\sin^5(\alpha) = \sin(\alpha) - \frac{1}{2}\sin(3\alpha) + \frac{1}{10}\sin(5\alpha) \tag{10}$$

We obtain a normalized fundamental component $\sin(\alpha)$ together with other distorting components. Distorting signal $d(t)$, using Equations (2) and (10), can be written as

$$d(t) = \frac{8}{5}\sin^5\left\{\frac{2\pi f_1 T}{\ln\frac{f_2}{f_1}}\left[\left(\frac{f_2}{f_1}\right)^{\frac{t}{T}} - 1\right]\right\}, \qquad 0 \le t \le T \tag{11}$$

For illustration we used swept signal in the $20\,Hz \div 2\,kHz$ range with sweeping time of $T = 20\sec$ and sampling frequency $f_s = 44{,}1\,kHz$. Figure 1 shows the result of total impulse response calculation

$$h_d(t) = d(t) \otimes f_i(t) \tag{12}$$

which can be divided into following partial components

$$h_d(t) = h_1(t) + h_3(t) + h_5(t) \tag{13}$$

By Fourier transformation $FT\{h_1(t)\}$, $FT\{h_3(t)\}$ and $FT\{h_5(t)\}$ we obtain the frequency spectra $H_1(\omega)$, $H_3(\omega)$ and $H_5(\omega)$, that are depicted in Figure 2.

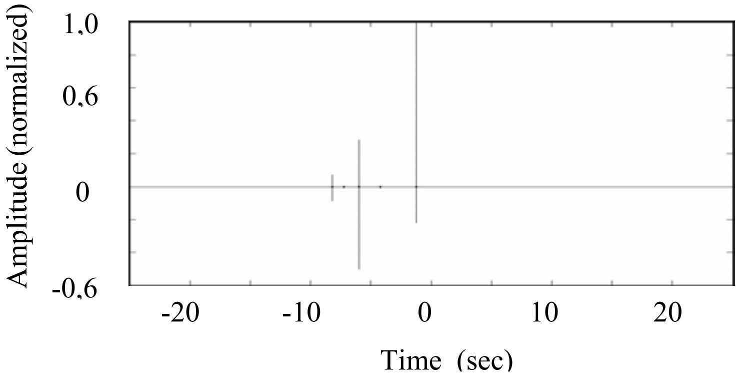

Figure 1. Impulse responses $h_1(t)$ (right), $h_3(t)$ (middle) and $h_5(t)$ (left) obtained by distorted test signal $d(t)$.

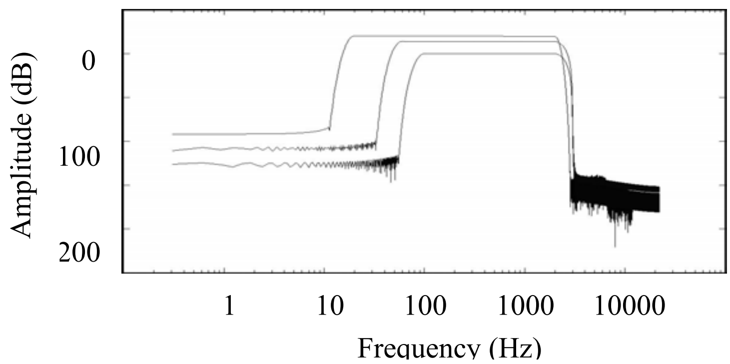

Figure 2. Transfer functions $H_1(f)$ (top), $H_3(f)$ (middle) and $H_5(f)$ (bottom) obtained by distorted test signal $d(t)$.

Observing Figure 1 and Figure 2, we can get an idea about the accuracy of the measuring method and signal to noise ratio. This noise is created by signal quantizing and signal processing. By appropriate selection of a test signal $x(t)$ parameters, it is possible to achieve results suitable for testing of *EAC* systems. The most important of these parameters are the sweeping range $f_1 \div f_2$, sweeping time T, quantizing level and sampling frequency f_s. Following these recommendations leads to optimum results.

3. ANALYSIS OF MAGNETIC RECORDING

The swept harmonic signal measuring method was used for analysis of the entire electro-acoustic chain for analogue magnetic recording with bias. This form of signal recording, which is characterized by having distorting odd harmonic components, enables us to practically verify of this test method. Sweeping test signal $x(t)$, digitally generated by a computer, is the first input to an external sound card, then via line amplifier to a tape recorder input. The signal is recorded onto a magnetic tape and subsequently reproduced and via sound card brought back to the computer. Details of this method of measurement and analysis concerning magnetic recording can be found in [11].

For measurement we used swept signal in $20\,Hz \div 20\ kHz$ range with sweeping time $T = 20$ sec and amplitude $0\ dB(1V)$, $-6\ dB$ and $-12dB$. We used the sampling frequency $f_s = 44{,}1\ kHz$ or $88{,}2\ kHz$. It is necessary to use higher sampling frequency when non-linear harmonic components exceed $f_s/2$ and cause aliasing. Figure 3 shows response of the entire record/playback chain to a test signal $x(t)$. We used a two-channel system, where only one channel was driven by signal $x(t)$, while a crosstalk $c_t(t)$ was measured in the other channel. During recording and playback process, there is a noise component $n(t)$ arising. The calculation of the total impulse response $h(t)$ of the channel driven by $x(t)$ can be written as

$$h(t) = f_i(t) \otimes [y(t) + n(t)] \tag{14}$$

which can be divided into partial components

$$h(t) = \sum_k h_k(t), \quad k = 1, 3, 5, ... \tag{15}$$

The response of crosstalk we can write as

$$h_c(t) = f_i(t) \otimes c_t(t) \tag{16}$$

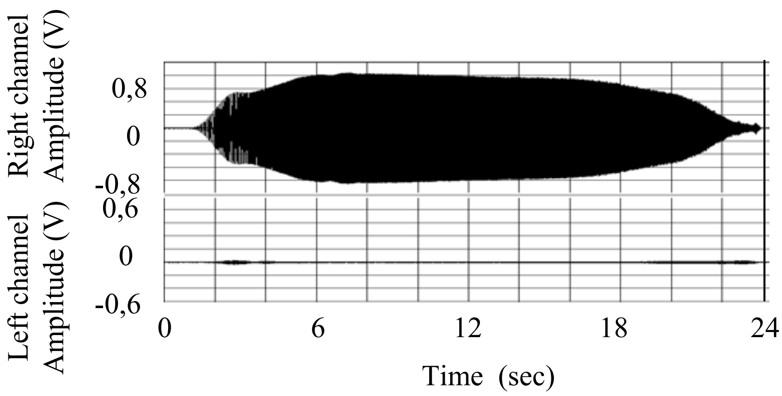

Figure 3. Time response of the recording system to the test signal (top) and crosstalk $c_t(t)$ between channels (bottom).

Figure 4 shows the frequency response of the system $H_1(f)$, the 3rd and 5th harmonic distortion component $H_3(f)$ and $H_5(f)$, respectively. Frequency responses $H_1(f)$, $H_3(f)$ and $H_5(f)$ can be obtained by $FT\{h_k(t)\}$.

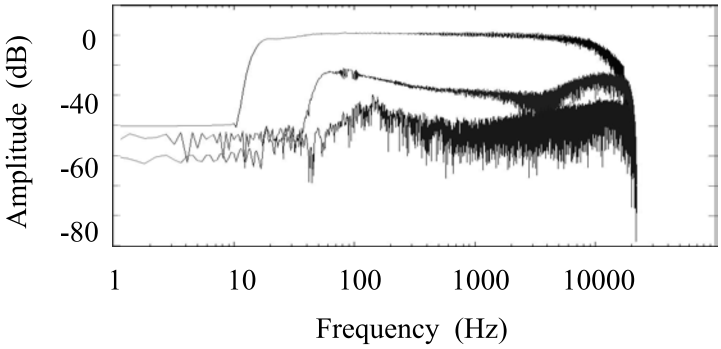

Figure 4. Frequency responses of the recording system $H_1(f)$ (top), $H_3(f)$ (middle) and $H_5(f)$ (bottom).

The response of crosstalk between channels $FT\{h_c(t)\}$ in frequency domain is shown in Figure 5. The impact of noise $n(t)$ manifests by reduced accuracy of measured system responses. This test method was used both on individual parts of system under test and the system as a whole. When analyzing systems, where only one component distorts, such as in magnetic recording, the calculation of responses $h_k(t)$ is relatively simple. However, when overload takes place in additional system components, a deeper analysis of non-linear distortion is required.

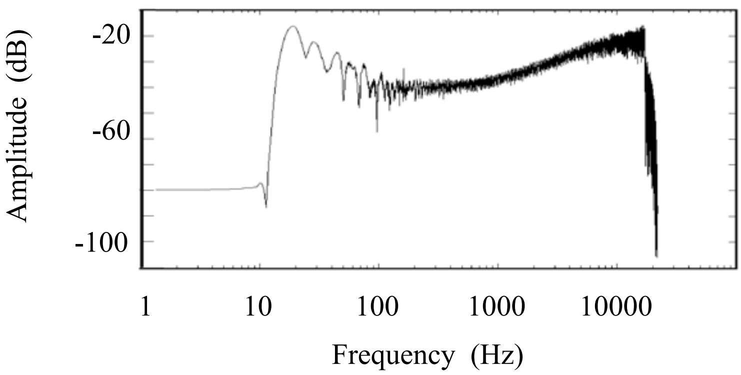

Figure 5. Frequency analysis of the crosstalk $H_c(f)$ between channels.

4. ANALYSIS AND PERFORMANCE MONITORING OF MULTICHANNEL SYSTEMS

Currently, there are multi-channel systems used both in audio technology and for testing purposes. Some of them also make use of signal compression [12]. During their analysis we can observe the individual channels or their mutual interaction. In our particular case, when analyzing systems with a signal compression, we do not analyze them from the psychoacoustic point of view, but we only monitor their functionality.

As an example, we used of measuring method for multi-channel audio system. It is a so-called 5.1 system, where the audio signal contains left, center and right front channels, two side channels and a low-frequency effects channel (*LFE*). In our project we placed test signals into individual channels in such a way that they either appear in all channels simultaneously or only one at a time in each channel. Such a system allows us to analyze either their mutual interaction or the performance of each channel on its own. While analyzing an individual channel, we can write for each of them

$$h_j(t) = y_j(t) \otimes f_j(t), \qquad j = 1,2,...,n \tag{17}$$

where $y_j(t)$ are the responses of individual channels to test signals $x_j(t)$ and $f_j(t)$ are the corresponding inversion filters. The set of test signals $x_j(t)$ was encoded with a *DTS* encoder and the result was analyzed. In Figure 6 is an example of *LFE* channel analysis, showing both the measured transfer function of the channel and the error signal, occurring due to signal encoding.

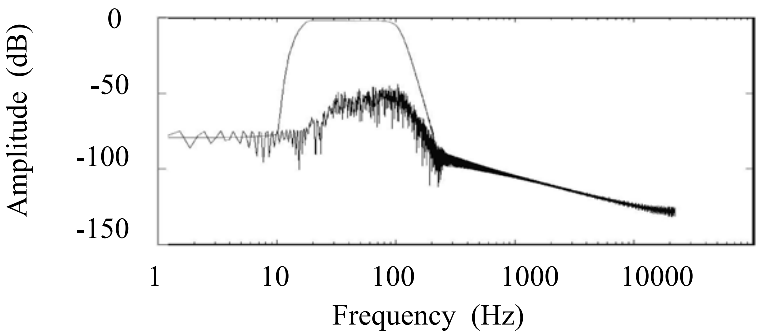

Figure 6. Frequency response $H_1(f)$ (top) and harmonic distortion (bottom) of the low frequency effects channel.

In case of a mutual channel interaction analysis, we can write for a simplified resultant response to individual test signals

$$y_{j,n}(t) = y_j(t) + \sum_{k=1}^{n+1} h_{j,k}(t) \otimes u_k(t), \quad j \neq k \tag{18}$$

where $y_{j,n}(t)$ is a total response of a j-channel during interaction of other n channels, $h_{j,k}(t)$ are impulse responses among individual channels and $u_k(t)$ are the test signals placed in other channels. Analysis of such a system has to be performed in partial steps, so that we gradually determine $h_j(t)$ and $h_{j,k}(t)$.

5. CONCLUSIONS

In our contribution we focused on analysis of transfer functions and performance monitoring of multi-channel *ELA* systems with swept signals. Initially we evaluated the test method from the calculated accuracy point of view, which also enables proper settings of *DSP* signal parameters. We examined both the individual components and the system as a whole. This test method also enables monitoring and diagnosing of faulty components.

When analyzing the whole *ELA* system in which only one component exhibits from non-linearities, the method can be used without problems. Here we achieved good results compared with conventional test methods. In cases where more than one component suffers from non-linearities, a deeper analysis of system behavior is required. In future we plan to develop a test method having the capability to analyze and monitor more complex systems that can exhibit non-linear distortion.

ACKNOWLEDGEMENTS

This work has been supported by the grant of the Czech Science Foundation no. 102/05/2054 “Qualitative Aspects of Audiovisual Information Processing in Multimedia Systems” and research project MSM6840770014 “Research in the Area of Prospective Information and Communication Technologies”.

REFERENCES

[1] D. D. Rife, J. Vanderkooy, “Transfer-function measurement with maximum-length sequences”, *Journal of Audio Engineering Society*, **37**, 419-444 (1989).

[2] C. Dunn, M. O. Hawksford, “Distortion Immunity of MLS-Derived Impulse Response Measurements ”, *Journal of Audio Engineering Society*, **41**, 314-335 (1993).

[3] F. Kadlec, “Measurement of Distributed Acoustic Systems Using Maximum-Length Sequences”, *The 100th Audio Engineering Society Convention*, 11-14 May 1996, Copenhagen, Convention paper 4269.

[4] J. S. BENDAT, *Nonlinear System Techniques and Applications*, John Wiley & Sons, New York, 1998.

[5] A. Farina, “Simultaneous Measurement of Impulse Response and Distortion with a Swept-Sine Technique”, *The 108th Audio Engineering Society Convention*, 19-22 February 2000, Paris, Convention paper 5093.

[6] A. Farina, A. Bellini, E. Armelloni, “Non-linear Convolution: a New Approach for the Auralization of Distorting Systems“, *The 110th Audio Engineering Society Convention*, 12–15 May 2001, Amsterdam, Convention paper 5359.

[7] G. Stan, J. Embrechts, D. Archambeau, “Comparision of Different Impulse Response Measurements Techniques”, *Journal of Audio Engineering Society*, **50**, 249-262 (2002).

[8] F. Kadlec, “Design, Generation and Analysis of Digital Test Signals”, *The 111th Audio Engineering Society Convention*, 30 November – 3 December 2001, New York, Convention paper 3501.

[9] M. Schetzen, *The Volterra and Wiener Theories of Nonlinear Systems*, John Wiley & Sons, 1980.

[10] http://functions.wolfram.com/

[11] F. Kadlec, L. Husník, “Analysis of Multimedia Audio Systems with Nonlinearities”, *Proceedings of the 9th Western Pacific Acoustic Conference*, 26-28 June 2006, Seoul, Korea.

[12] M. Bosi, R. Goldberg, *Introduction to Digital Audio Coding and Standards,* Kluwer Academic Publishers, Norwell, 2003.