ISSN 0005–1144
ATKAAF 47(3–4), 149–154 (2006)

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Frequency Response Measurements of Hearing Aids Based on Composed Audio Test Signal

UDK 534.77:621.317.61
616.28-77
IFAC 4.2.1;5.9.2

Original scientific paper

This paper describes the implementation of composed audio test signal (CATS) based test system for evaluation of electroacoustic performance of hearing aids. The system applies a frequency-shaped CATS sequence to a hearing aid as an acoustic stimulus and measures the acoustic response of the aid in IEC standard occluded-ear stimulator. The frequency response can then be found by taking the discrete Fourier Transform of the impulse response. This method has several advantages over traditional noise-based methods regard measurements in the absence or presence of the external interference noise.

Key words: Composed audio test signal – CATS, hearing aid, acoustical measurements

1 INTRODUCTION

Many modern hearing-aid use automatic signal processing (ASP) schemes in an attempt to deal with problems associated with hearing loss such as excess spread of masking or increased sensitivity to background noise. ASP hearing aids adjust their gain of frequency response as function of the input signal they receive. These hearing instruments can be classified into two broad categories: fixed-frequency response (that is, Automatic Gain Control, AGC) and level-dependant frequency response. It should be noted that some authors refer to ASP as Automatic Frequency Response (AFR) [1].

In many instances the hearing-aid (HA) testing procedures in common use (ANSI pure tone tests [2], IEC pure tone tests [3, 4]) cannot easily characterize how these hearing aids will respond to typical environmental stimuli. Also, the information provided to those prescribing and dispensing HA-devices does not facilitate accurate adjustment: typically, descriptions of the signal processing procedures used in a HA-device are not provided or are insufficiently detailed. Because a pure tone test presents only one frequency at one instant of time, the frequency response measured in this way is not indicative of the frequency response when the hearing aid is exposed to complex sounds. An approximation of the characteristics of the hearing aid in actual use can be obtained by using a signal that is acoustically similar to »real-world« input signals. Normally, the input signal of the interest is speech, possibly in the presence of an interfering noise background. For this reason there has been a recent trend toward using broadband »speech-like« signals to test hearing aids.

A new ANSI standard, S3.42 [5], uses a speech-shaped noise test signal and fast Fourier transform (FFT)-based analysis for frequency response measurement that is better approximation of a hearing aid's response to a »real-world« stimulus. It has also been suggested to the ANSI standards committee that maximum-length-sequence (MLS) based testing should be considered as a candidate for hearing aid measurement [6, 7]. Binary MLS are periodic, two-level (normalized values 1 or –1) pseudorandom sequences. Their probability density function (PDF) is, therefore, everywhere zero except at values +1 and –1 where it is an impulse of area 0.5. Their Crest-factor (CF) is 0 dB. Autocorrelation function of a maximum-length-sequence is a perfect impulse, therefore, it's power spectrum is flat [8]. To compute system impulse response (IR), system's output and input signals are cross-correlated. The Frequency response of a system can be found by using FFT of IR.

This paper describes the implementation of composed-audio-test-signal (CATS) based test system for evaluation of hearing aids. CATS is »speech-like« test signal. It was developed for measurement of electroacoustical (EA) components and systems [9, 10]. Previous publications have described applications of CATS in amplifier and loudspeaker testing [10, 11], and frequency response measurement [12] and distortions measurements in EA systems [13]. This tests use CATS to excite a linear time-invariant (LTI) system and an impulse response is extracted from the output signal by cross-correlating it with the input signal (CATS). Then, the frequency response of a system can be found by applying discrete Fourier transform on the system's impulse response.

2 BACKGROUND

The input signal to and output signal from the hearing aid are synchronously collected and periodically cross-correlated. This results in the periodic impulse response (PIR) of the hearing aid. We will denote PIR with $h'(n)$. Since CATS is periodical signal with period **L**, where **L** is defined by equation

$$L = T_s \cdot N \;\; (T_s = \frac{1}{f_s}, \, f_s - \text{sampling frequency})$$

without DC component, its approximated auto-correlation is periodical (denoted with apostrophe) unit sample sequence [8], defined with:

$$\phi_{ss}(n) = s'(n)\Phi s'(n) = \frac{1}{L}\sum_{k=0}^{L-1} s'(n)s'(k+n), \qquad (1)$$

where Φ denotes circular correlation, and $s'(n)$ CATS.

If we excite LTI system with CATS $s'(n)$ and if the periodic-impulse response of this system is $h'(n)$, then the system's output signal $y'(n)$ is expressed with periodical convolution:

$$y'(n) = s'(n) \otimes h'(n) = \sum_{k=0}^{L-1} s'(n)h'(n-k). \qquad (2)$$

To recover the PIR, the output signal of the system $y'(n)$ is cross-correlated with input CATS $s'(n)$ [8]. Mathematically, this can be expressed:

$$\phi_{xy} = s'(n)\Phi y'(n), \qquad (3)$$

$$\phi_{xy} = s'(n)\Phi\left[s'(n) \otimes h'(n)\right] = \\ = \left[s'(n)\Phi s'(n)\right] \otimes h'(n) = \phi_{ss} \otimes h'(n). \qquad (4)$$

This means that periodical cross-correlation of the output signal with the input signal is equal to the convolution of the auto-correlation sequence ϕ_{ss} with systems PIR. For large **L** we have

$$\phi_{xy}(n) = h'(n). \qquad (5)$$

Power spectrum (PS) of the CATS is flat, which means that it has value 0 at DC and constant value at all other frequencies. The CATS's PS is obtained by using DFT of it's auto-correlation:

$$PS\{s'(n)\} = DFT\{\phi_{ss}(n)\}. \qquad (6)$$

The phase spectrum of CATS varies randomly with frequency, and PDF is uniform over range $+\pi$ to $-\pi$. PIR is normal impulse response when the CATS measurement has been properly made $(h'(n)\langle L)$, because there is no time aliasing. To obtain the frequency response of the measured system, the discrete Fourier transform (DFT) of the initial portion of PIR is computed.

3 TEST SIGNAL CONSIDERATION

The most information in speech and music is to be found in transients. The transients are peak-asymmetrical signals, with fast changeable short-time spectrum without DC component. PDF of time distribution of speech and music amplitudes is bell-like. The normalized rate of change of speech or music is not greater than 75 mV/µs/V_p, with peak factor between 4 and 5. According to these parameters composed audio signal is synthesized.

CATS is formed by mixing three mutually independent composed saw signals (CSS) [9, 10]. CSS signal is generated by current generator using D/A converter that charge and discharge the integrator's capacitor with constant current. D/A converter is controlled by pseudo random sequence (MLS) that is generated by shift register composed of m bistabiles. Shift-register is put by clock pulses in $N = 2^n - 1$ different states before repeating, generating on any output pseudo-random sequences of the length N. To every state of the shift-register corresponds N different pseudo-random values of current that estimates the slew-rate and through it the time-duration of a »tooth« of a CSS [14]. After every N clock pulses »teeth« of CSS are repeated. CATS has peak and waveform asymmetry without DC component. The spectrum of CATS is steady and repeatable. CATS with length of L samples and sampling rate f_s has all frequency components from f_s/L to $(1-1/L) \cdot f_s/2$, with frequency resolution of

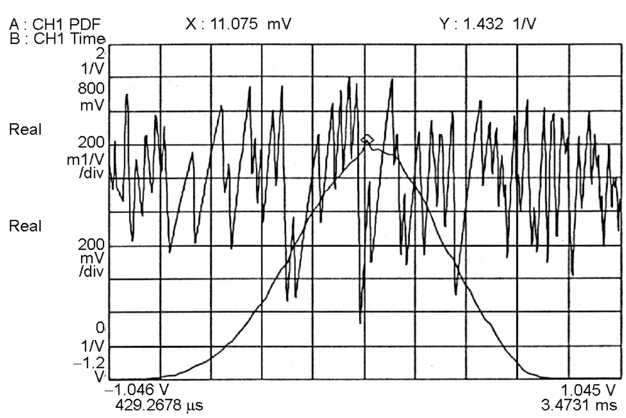

Fig. 1 Composed audio test signal (CATS) and PDF function of time distribution

f_s/L. CATS's probability density function (PDF) is bell-like, and it's CF is somewhere between 4 and 5. Power spectrum of CATS is flat (Figure 1).

4 TEST SYSTEM DESIGN

CATS-based frequency response measurements apply CATS to a hearing aid with a set of different input sound pressure levels (SPLs). By testing the hearing aid with different SPLs we can observe the effect of the input SPL on the hearing aid frequency response and determine the degree of distortion for each input SPL. Hearing aid measurement was done at the reference test position (RTP), where RTP was set by measuring the output SPL in the full on gain (FOG) condition for 90 dB SPL input, then reducing the input to 60 dB SPL and adjusting the gain of the hearing aid so that the output SPL drops by 17 dB. Reference frequency response denoted with 50-RR and 90-RR, was measured hearing aid frequency response for 50 dB and 90 dB of input SPL. While using 50-RR, the input SPL to the hearing aid was increased from 50 to 90 dB with step of 10 dB. The input SPL to the hearing aid was decreased from 90 dB to 50 dB, while using 90-RR.

The measurement system components are shown in Figure 2.

The digital signal processing board (DSP) and digital playback board (QDA) both reside in personal computer (PC) that is used for data collection and measurement acquired computations. The DSP samples both input channels, that is reference (REF) and coupler (CPL) channel simultaneously and records obtained data directly to disc. The QDA plays stimuli that have been uploaded to on board memory. To ensure that record and playback operations are tightly synchronized, the DSP uses hardware triggering to start the QDA playback. Both devices share the same sample clock. A programmable output attenuator (PA_OUT) is used to adjust the SPL in the portable anechoic chamber. Two programmable input attenuators (PA_REF and PA_CPL) and a dual-channel programmable preamplifier/high-pass filter are used to scale the input level into range for the DSP analog-to-digital converters. The high-pass filters ($f_c = 80$ Hz) remove low-frequency room (ventilation) noise. A Bruel& Kjaer (B&K) type 4134 pressure microphone is used on each input channel. The reference microphone and the hearing aid (that is, the device under test) are located opposite each other inside the chamber (B&K type 4212, equidistant from the center of the chamber) in symmetric sound field. The acoustic output of the hearing aid is connected to an IEC standard occluded-ear simulator (B&K 4157) to approximate the 'acoustic environment' of an ear. The coupler microphone (B&K 4134) is inside the occluded-ear simulator. Frequency response errors from reference (REF), coupler (CPL), and output channels are corrected by flattening the REF channel frequency response and removing the difference in frequency response between the two input channels. The average for each input channel is cross-correlated. The resulting impulse response for each input channel (REF and CPL) is truncated from 16383 to 2048 samples, windowed using a Hanning window and transformed with FFT. This yields the frequency response for each channel. Flattening frequency response $H_{REF}(f)$ of REF channel and subtracting the frequency response difference between channels $H_{diff}(f)$ (which was determined during test sys-

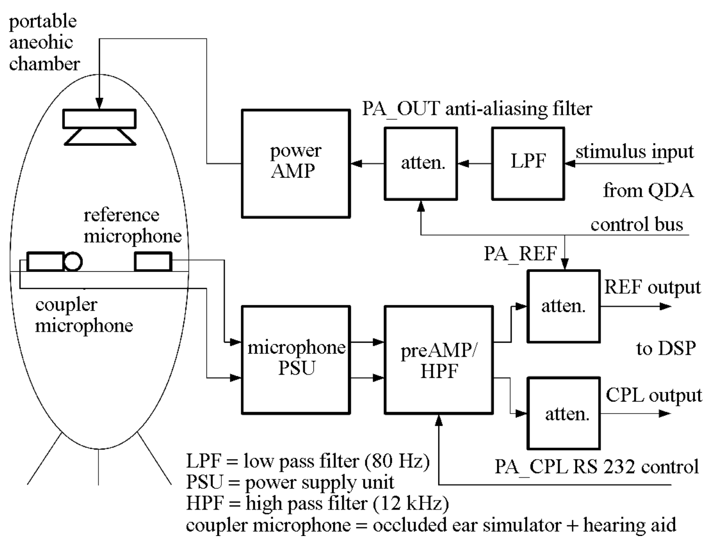

Fig. 2 Block diagram of CATS test system

tem calibration), flattens both input channels and delivers the desired acoustic input signal to the hearing aid. The magnitude of frequency response of a hearing aid expressed in decibels $H_{aid}(f)$ is then:

$$H_{aid}(f) = H_{CPL}(f) - H_{REF}(f) - H_{diff}(f), \text{ dB} \quad (7)$$

where $H_{CPL}(f)$ denotes frequency response of coupler channel.

5 MEASUREMENTS

Measured hearing aid (Digital One RTI) is a two-channel hearing aid with continuous reduction of the low and high tone AGC threshold input SPL, crossover frequency (CF) of 2 kHz, and continuous reduction of the low-tone gain. Hearing aid was measured with 3 different input signals. Test signals SW1 and SW2 used in measurements were fast pure tone sweep test signals from high to low frequency with time duration somewhere around 2 sec. Hearing aid were measured (tested) with two modes of operation. First mode (denoted with number one in the name of input test signal) is two channel compression with CF 2 kHz, compression ratio (CR) 2, and compression threshold (CT) 50 dB SPL. Second mode (denoted with number two in the name of input test signal) is increased AGC threshold input SPL and reduced low-tone frequency gain of measured hearing aid. When hearing aid was measured with second pair of signals SW1B and SW2B, input signals to hearing aid were slow step-sweep test signals (1/24 octave step), and hearing aid itself was in first and second mode of operation, as explained above. CATS1 and CATS2 are CATS test signals where 1 and 2 denote mode one and mode two of hearing aid operation.

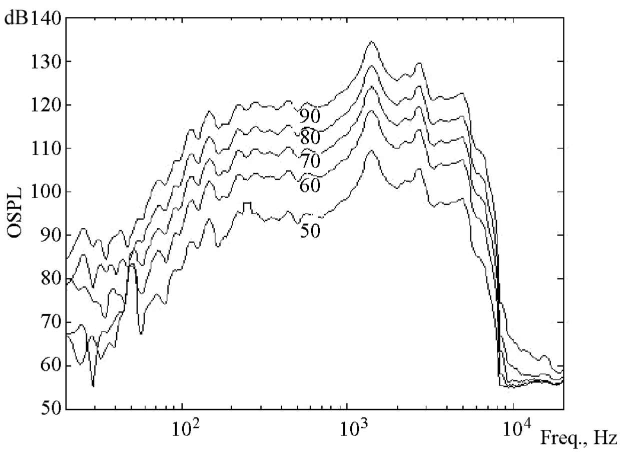

Fig. 3 Hearing aid CATS1 frequency response OSPL dB for 90 to 50 dB SPL input level

In Figure 3 is depicted frequency response (FR) of the hearing aid (HA) while using 90-RR as reference frequency response. The input SPL was decreased from 90 to 50 dB with step of 10 dB. The HA was measured with CATS1. Acoustic gains in both channels are frequency dependent, while differences of FRs are frequency independent, meaning FRs are equidistant. Compression ratio (CR) is reduced with decrease input CATS SPLs. Figure 4. presents hearing aid FR measured with CATS2 using 50-RR as reference frequency response. The input SPLs were increased from 50 to 90 dB. Test signal was CATS2. Acoustic gain in both channels is frequency dependant. Low tone frequency gain is reduced. CR and AGC threshold input SPLs are frequency dependant in both channels. CR increase with CATS input SPL. Peak clipping occurs at 80 dB input SPL. Maximum OSPL is frequency dependant, from 120 dB to 140 dB SPL for channel one, to 135 dB SPL for channel two.

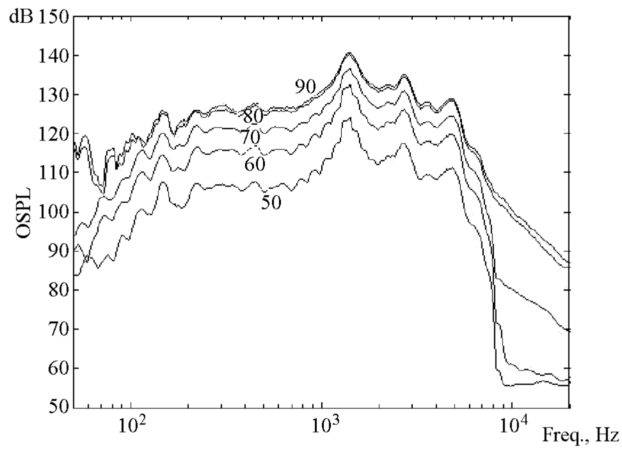

Fig. 4 Hearing aid CATS2 frequency response OSPL dB for 50 to 90 dB SPL input level

In Figure 5 is presented hearing aid steep-sweep FR for 50 to 90 dB input SPL, using 50-RR as reference, and HA in mode 2 of operation. CR and AGC thresholds (input SPL) are frequency dependant in both channels. CR increases with sweep input SPL. Peak clipping occurs at 80 dB input SPL. Acoustic gains in both channels are frequency dependant. Low tone frequency gain is reduced. Maximal OSPL is frequency from 120 dB to 135 dB SPL for channel 1, and to 130 dB SPL for channel 2. In Figure 6 is presented HA steep-sweep FR for 50 to 90 input SPL, 50-RR as reference, mode 1 of operation. Acoustic gain in both channels is frequency dependant. Measured frequency responses are equidistant.

Figure 7 depicts fast tone sweep test FR for 50 to 90 dB input SPL, using 50-RR as reference and

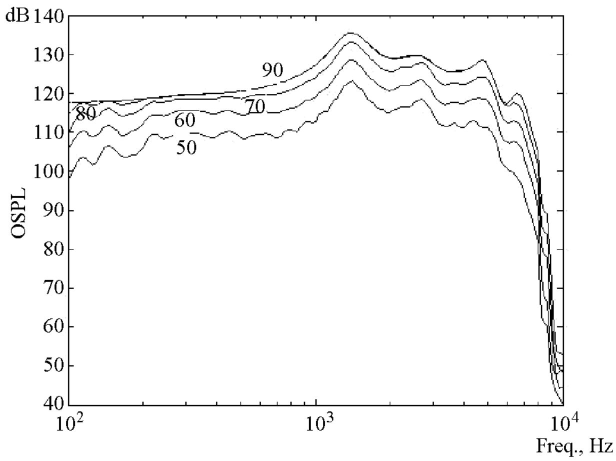

Fig. 5 Hearing aid SW2B (step sweep) frequency response OSPL dB for 50 to 90 dB SPL input level

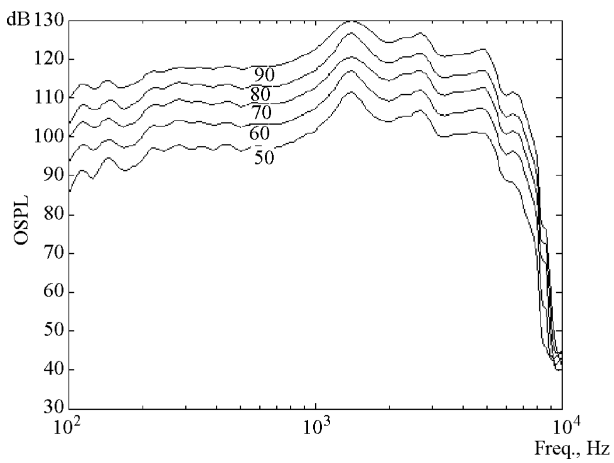

Fig. 6 Hearing aid SW1B (step sweep) frequency response OSPL dB for 50 to 90 dB SPL input level

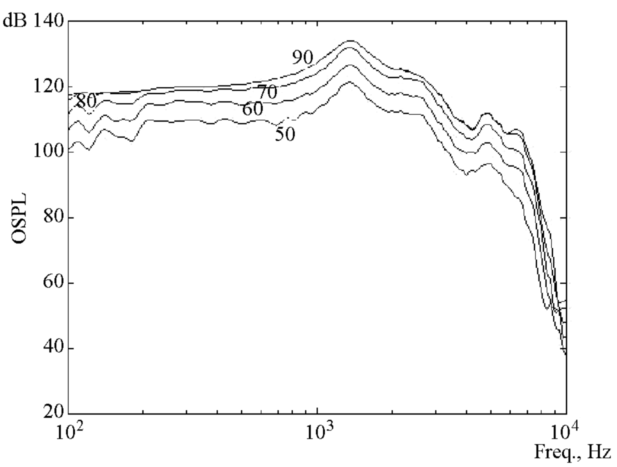

Fig. 7 Hearing aid SW2 (fast sweep) frequency response OSPL dB for 50 to 90 dB SPL input level

with HA in mode 2 of operation. Figure 8 depicts fast tone sweep test FR, input levels increased from 50 to 90 dB SPL, using 50-RR as reference, with HA in mode 1 of operation. Fast sweep test shows reduced high frequency gain versus step (slow) sweep test.

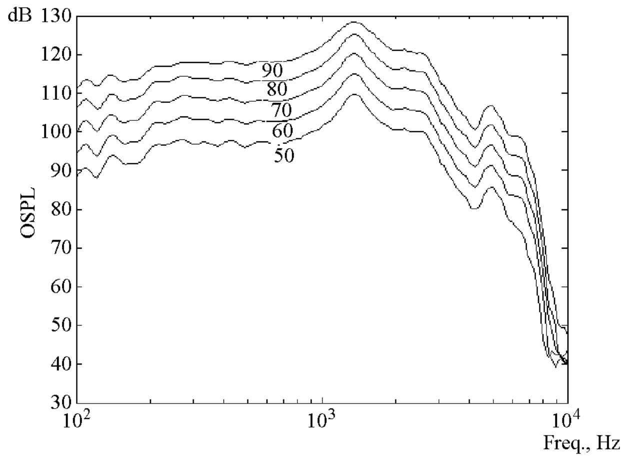

Fig. 8 Hearing aid SW1 (fast sweep) frequency response OSPL dB for 50 to 90 dB SPL input level

6 CONCLUSION

In conventional hearing aid test procedures, pure tones sweep test are used to measure the frequency response. When applied to hearing aids that incorporate nonlinear or frequency-selective processing, these tests can sometimes give misleading results. A new ANSI standard addresses this problem by using a speech-shaped noise stimulus with FFT-based analysis. We propose frequency response measurement with CATS that can predict how a hearing aid will respond to environmental stimuli, and characterize modern ASP hearing aids that use input signal parameters to control hearing aid frequency response. When measured with CATS2 frequency response OSPL of the hearing aid is higher, than the OSPL obtained using traditional sweep-tone measurement. Change of the input SPL to the hearing aid for 40 dB results in 20 dB change of the OSPL when measuring with CATS2, and 10 dB change of the OSPL when using sweep-tone test. Compression threshold is increased and compression ratio is reduced for CATS2 measurement. We concur with others who propose that CATS-based methods should be considered as candidates for this measurement application because of their increased speed and their flexibility.

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Mjerenje frekvencijskog odziva slušnog pomagala pomoću složenog audio test signala. U članku je opisana primjena složenog audio test signala (CATS) pri određivanju elektroakustičkih svojstava slušnih pomagala. Test sustav primjenjuje frekvencijski ugođenu CATS sekvenciju kao akustičku pobudu za mjerenje akustičkog odziva slušnog pomagala u stimulatoru po IEC standardu. Frekvencijski odziv se dobiva iz impulsnog odziva pomoću diskretne Fourierove transformacije. Ta metoda (CATS) ima znatne prednosti spram uobičajene mjerne metode sa šumnim signalom, pri mjerenju u prisutnosti ili odsutnosti vanjskog interferirajućeg šuma.

Ključne riječi: složeni audio test signal, slušno pomagalo, akustička mjerenja

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Received: 2006-11-08