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SYNTHETIC SIGNALS FOR VERIFYING NOISE REDUCTION SYSTEMS IN DIGITAL HEARING INSTRUMENTS

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ABSTRACT

Hearing instrument development involves the acoustic verification of system functionality. The used test signals should allow efficient repetition of tests. Synthetic stimuli are therefore proposed as test signals, because they can be tailored to the test objectives. For the purpose of frequency response measurements in digital hearing instruments, we present a synthesis procedure targeted at test signals for modulation-based noise reduction systems. The resulting stimuli are assessed during measurement of the noise reduction performance in an exemplary hearing instrument.

1. INTRODUCTION

System *verification* tests if the system under test meets the system specifications. This process has to be distinguished from system *validation*, which assesses a system's capability of meeting customer needs "in the most realistic environment achievable" [1].

In the following, we present a measurement-based test procedure targeted at modulation-based noise reduction systems in digital hearing instruments. Our focus is on verification and not on validation of the system under test. One important aspect of verification is *regression testing*, i.e. the process of repeating a selected fraction of tests [1] to verify if system performance has remained correct after implementation changes. This can lead to numerous repetitions of one test. In this case, being able to reproduce test results efficiently can be more important than creating a realistic environment. Prior work deals with the validation of noise reduction systems in hearing instruments (e.g. [2], where, for example, the sound of a drilling machine represents a realistic environment), but to our knowledge does not provide a verification procedure optimized for frequent test repetition.

We believe that stimulus-based acoustic verification of hearing instruments should avoid using real-world stimuli, because these lack the following properties desirable in test signal design:

- A structure that forces the system under test to walk through its parameter space systematically.
- Freedom in changing the signal's temporal characteristics, selecting its power-spectrum [3], and making its spectral components sufficiently constant over the frequency range of interest [4].

We therefore propose synthetic test signals, because these can be synthesized with regard to the temporal characteristics of interest, the systematic estimation of system parameters, and accurate measurement results.

First, known synthetic signals are presented. Then, we show how to modify these signals in order to allow noise reduction testing. At the end, the resulting test signals are evaluated while measuring the noise reduction performance in a commercially available digital hearing instrument.

2. SYNTHETIC SIGNALS

Synthetic signals are available from prior published work, e.g. as perturbation signals for system identification. We have chosen to base our test stimuli on the periodic signals among these, because periodicity brings the following advantages:

- Periodic stimuli avoid leakage errors [5] in processing based on Discrete Fourier Transforms (DFT).
- Only one period of the desired stimulus needs to be computed, which limits synthesis time.

A disadvantage of a periodic signal is its discrete power-spectrum: Measuring frequency responses with periodic stimuli will only cover discrete frequencies.

Known synthetic test signals have different synthesis procedures, differing in amplitude distribution of the resulting stimulus and the possibility to select its power-spectrum. The following two sections summarize the kinds of periodic signals that were considered here.

2.1 Multi-sine signals

A periodic signal with a predefined discrete power-spectrum can be obtained by generating and adding up sine waves of different frequencies, those being the ones where non-zero signal power is desired. We call the result of this synthesis procedure a *multi-sine signal*. The desired spectrum determines the amplitude of the sine signals to be summed; phases however can be varied to optimize further properties of the resulting signal. A quite commonly encountered optimization goal is minimizing the peak factor [6] [7] [8] [9].

Since multi-sine signals result from the summing of sine signals with different amplitudes and phases, computation is most efficient when using Fast Fourier Transform methods [3] [5]. The Frequency Domain Identification Toolbox (FDIDENT) for MATLAB [10] offers readily accessible functions for synthesizing multi-sine signals [11].

2.2 Binary signals

A signal whose amplitude has only two discrete values is called a *binary signal*. Known examples for this kind of stimulus are *pseudorandom binary sequences* (e.g. [12]) and *discrete-interval binary sequences* [13] [14]. Approximating a given power-spectrum is in general only possible with the latter kind of signals, because the power of pseudorandom binary sequences has a fixed distribution over frequency.

3. APPLICATION TO NOISE REDUCTION

3.1 Scope

The following relates to noise reduction systems in digital hearing instruments. The scope is limited to modulation-based processing [15], which is one common noise reduction method in hearing instruments [16]. Modulation-based noise reduction systems reduce gain in subbands carrying unmodulated signals, while preserving gain in those bands in which modulation is present. Here, *modulation* means fluctuation of the subband signal's envelope over time.

3.2 Test objective

A noise reduction system should attenuate noise. Therefore, its attenuation is the system parameter of major interest in testing. When the system under test operates in multiple subbands, a systematic test procedure should measure the attenuation in each of them separately. We therefore define as a test objective: stimulate each subband of the noise reduction individually and measure the resulting impact.

3.3 Test signal synthesis

A test signal supporting the above objective has to meet two demands: it should not only perform well in frequency response measurements, but also stimulate each subband of the system under test separately in a predefined way. For simplicity, let us assume that maximum possible attenuation should be achieved in one noise reduction subband, while all other subbands should remain unchanged.

The corresponding stimulus for modulation-based noise reduction thus should have different modulation in different frequency ranges. We now synthesize a band-limited signal for each subband separately, even if adjacent subbands need identical modulation. This gives us signals of well-known properties (e.g. with a defined peak factor) per subband.

Let the number of subbands be M. We assign an index $i \in \{ 1, 2, \dots M \}$ to each of them. Let the lower and upper crossover frequency of subband number i be $f_{c,\,i}$ and $f_{c,\,i+1}$ respectively. The subband's bandwidth B_i is given below:

$$B_i = f_{c,i+1} - f_{c,i}\,. \tag{1}$$

Let n_i be an unmodulated signal for stimulating subband number i. Then it is possible to construct a corresponding modulated signal s_i with a modulation frequency $f_{m,i} < 0.5\, B_i$:

$$s_i(t) = A_i \cdot n_i(t) \cdot \left[1 + \cos\left(2\pi\, f_{m,i}\, t\right)\right]. \tag{2}$$

The idea is to set A_i such that s_i has the same power as n_i. For $f_{m,i} > 0$, signal s_i has approximately the same power as n_i if $A_i = \sqrt{2/3}$. This approximation was sufficient in our case.

Note that modulation in the hearing instrument domain refers to a property of speech and is therefore usually something less formal than the cosine term in equation (2). However, using the given formal definition allows us to impose a precisely defined modulation frequency on the test signal, giving us full control over that test parameter.

Let $\mathbf{\Phi}_i$ be the desired two-sided power-spectral density of signal n_i. In order to have the power of s_i distributed in the frequency range from $f_{c,\,i}$ to $f_{c,\,i+1}$, we specify $\mathbf{\Phi}_i$ by the following equation, where r is the RMS of n_i:

$$\Phi_i(f) = \begin{cases} \frac{r^2}{2B_i - 4 f_{m,i}} & ;\; f_{c,i} + f_{m,i} < |f| < f_{c,i+1} - f_{m,i} \\ 0 & ;\; else \end{cases} . \tag{3}$$

The idea is to choose n_i from the multi-sine signals or discrete-interval binary sequences introduced earlier (pseudorandom sequences are in general no candidates for n_i because of their fixed power-spectrum). This means that n_i is periodic. Thus, n_i can be represented by means of its complex Fourier coefficients $\underline{c}_{k,i}$ via complex Fourier Series, where T is the period of n_i, and j is the imaginary unit:

$$n_i(t) = \sum_{k=-\infty}^{\infty} \underline{c}_{k,i} \cdot e^{j 2\pi \frac{kt}{T}}\,. \tag{4}$$

Being periodic, n_i has its signal power concentrated at discrete frequencies, the multiples of T^{-1}. Therefore it is impossible to synthesize a signal whose power-spectral density is <u>exactly</u> $\mathbf{\Phi}_i$ from equation (3). However, the signal power will be distributed <u>approximately</u> as defined by $\mathbf{\Phi}_i$, if the coefficients $\underline{c}_{k,\,i}$ are prescribed as follows for synthesis of n_i:

$$\left|\underline{c}_{k,i}\right| = \sqrt{\frac{\Phi_i\left(\frac{k}{T}\right)}{T}}\,. \tag{5}$$

For discrete-time processing, the range of k should be limited to $-0.5\, T f_s < k < 0.5\, T f_s$; where f_s is the sampling rate.

To synthesize a test signal θ_b that evokes attenuation of a noise reduction system in subband number b only, we add an unmodulated signal having most of its power in that subband to modulated signals corresponding to the other subbands: We set $f_{m,\,b} = 0$. Additionally, we set $f_{m,\,i} > 0$ to a modulation frequency characteristic for the noise reduction system under test for all $i \in (\{ 1, 2, \dots M \}\backslash\{b\})$. Then θ_b can be constructed according to the following equation:

$$\theta_b(t) = n_b(t) + \sum_{i \in (\{1,2 \dots M\} \backslash \{b\})} s_i(t)\,. \tag{6}$$

The above test signal θ_b is an exemplary stimulus for noise reduction testing. The following section describes an example of its application.

4. MEASUREMENTS

The presented synthetic stimuli were used to measure noise reduction performance in a commercially available hearing instrument. Measurements were performed with stimuli based on a multi-sine signal on the one hand, and based on a discrete-interval binary sequence on the other hand.

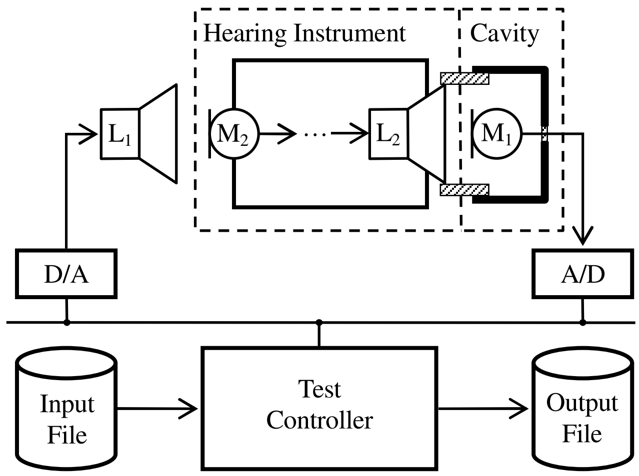

Figure 1 – Measurement setup

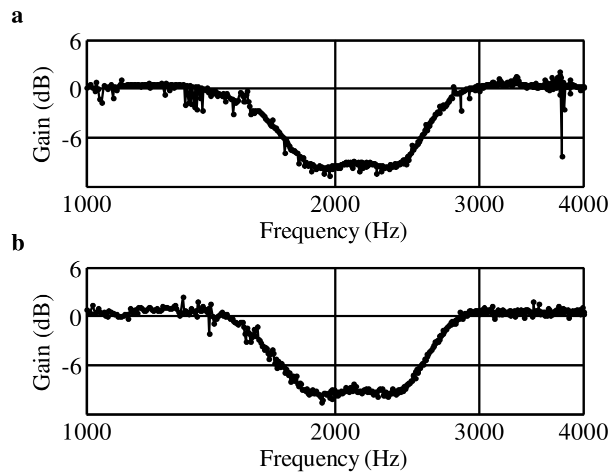

Figure 2 – Frequency response of a noise reduction system that is stimulated for attenuating only one subband and measured with (**a**) multi-sines; (**b**) discrete-interval binary sequences

4.1 Setup

A test system was set up for making measurements with synthetic test signals. Figure 1 illustrates the setup: The hearing instrument under test is located in an off-the-shelf acoustic measurement box with a loudspeaker (L_1) for presenting test stimuli to be picked up by the hearing instrument's input transducer (M_2). The hearing instrument's output transducer (L_2) is coupled with a measurement microphone (M_1) so tightly that environment sounds can be neglected in comparison to the hearing instrument's output. The coupler is a cavity simulating the human ear canal.

A digital playback and recording system can play stimuli from a MATLAB-created file (Input File) via a digital-to-analogue converter (D/A) and the loudspeaker of the measurement box (L_1), while recording the hearing instrument's output via the measurement microphone and an analogue-to-digital converter (A/D). The recorded digital data is stored in a MATLAB-readable file (Output File). The sampling rate for both playing and recording signals is 22050 Hz. A test controller ensures synchronous playback and recording.

4.2 Procedure

DFT-based processing can approximate the frequency response function of a system whose input stimulus is a periodic digital test signal: The system's output is digitized with the clock of the input signal. Then, the DFT is applied to both the input signal and the digitized output. The frequency response is calculated at each DFT frequency by dividing the absolute value of the output-related DFT bin by the corresponding input-related value [4] [17].

Leakage errors can be avoided if the stimulus is periodic, and if the DFT window contains an integer number of its periods [5]. For equation (2), this means: an integer number of periods of both $\cos(2\pi f_{m,i}\, t)$ and n_i should fit into one window. We therefore set the window length as the period T of n_i and ensured that $T \cdot f_{m,i}$ was integer. A window of 4096 samples allowed us both the use of the Fast Fourier Transform (FFT) and the choice of about 5.4 Hz modulation frequency. This frequency is typical for speech, whose modulation spectrum is significant in the range from 1 to 12 Hz [15].

Equations (1) to (6) were used for synthesizing test signals, where n_i was varied for two subsequent test runs:

- For the first test run, n_i was a multi-sine signal generated by the FDIDENT function "msinclip", which uses a synthesis algorithm by Van der Ouderaa, Schoukens and Renneboog [7].
- For the second test run, n_i was a discrete-interval binary sequence, generated by the FDIDENT function "dibs", which is based on a synthesis algorithm by Van den Bos and Krol [13].

A commercially available hearing instrument was the device under test, and its gain was set 20 dB below the maximum offered value to reduce non-linearities. All adaptive features of the hearing instrument, apart from noise reduction, were turned off for all test runs. Measurements were performed with different θ_b according to equation (6), where the synthesis parameter r in equation (3) was adjusted to yield a 70 dB SPL level in each subband. Different θ_b were used, corresponding to different band indices b. Two measurements were performed per signal θ_b: first with the noise reduction system of the hearing instrument switched off, and second while having it switched on. For each measurement, the test stimulus was presented during 15 seconds in order to allow the system under test to reach steady-state.

Bin-by-bin division of FFT absolute values from the second measurement by corresponding values of the first measurement delivered the frequency response of the noise reduction system. Five FFT windows were averaged for spectral smoothing [18]. These windows were taken from the last five seconds of the test run in order to observe the steady-state condition.

4.3 Results

Figure 2 and 3 show exemplary results of the measured frequency responses. The upper diagrams of these figures show results obtained with a multi-sine-based stimulus, while the lower ones relate to discrete-interval binary sequences.

It is evident from figure 2 that both kinds of stimuli could be used as a test signal for this condition, although

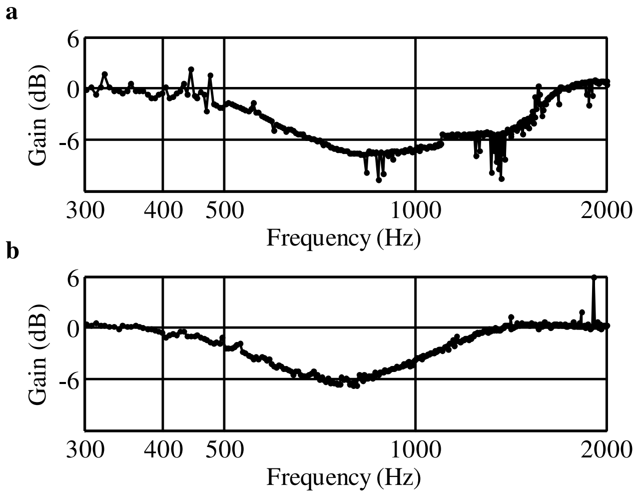

Figure 3 – Results similar to figure 2, but with a stimulus targeted at a different subband; again measured with (**a**) multi-sines; (**b**) discrete-interval binary sequences

the multi-sine-based measurement has higher measurement errors than the one based on discrete-interval binary sequences. Figure 3 shows a case in which the multi-sine-based stimulus failed its goal of evoking attenuation in only one noise reduction subband: While the results based on discrete-interval binary sequences (figure 3b) are close to the expected frequency response function with an attenuating subband around 800 Hz, the multi-sine based measurement obviously triggers additional attenuation in a subband around 1300 Hz (figure 3a).

5. CONCLUSION

Synthetic test signals have been proposed for verification in the domain of digital hearing instruments. Multi-sine signals and discrete-interval binary sequences have been used to synthesize stimuli targeted at systematic verification of a modulation-based noise reduction system.

Measurements with a commercially available hearing instrument showed that the synthetic signals succeeded in both stimulating attenuation of unmodulated signals in the noise reduction system under test and measuring the system's frequency response function.

By comparing the measurement results, we conclude that discrete-interval binary sequences are preferred for stimulus design in the given test case, due to their accuracy in stimulating the desired system behaviour and measuring frequency response functions. Since we need about 15 seconds per measurement, we can systematically walk through all subbands of a typical noise reduction system in less than half an hour. Therefore we regard the presented test procedure as a suitable basis for affordable regression testing.

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