How to Obtain High Quality Input Data for Auralization?

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Introduction

Auralization of structure-borne and airborne noise problems contributes to understanding of sound transmission in a significant way. One expects the auralization result to be as close to the real world result as possible. In hybrid models where measurement and simulation data are both used to generate the final audible result, the measurement data obviously has to be very precise. Even highly accurate numeric simulations require measure data, e.g. structure-borne impedances of materials or couplings. Therefore, the measurement of input data is of high interest regarding the simulation quality.

Measuring structure borne impedances up to high frequencies for ongoing coupling calculations often lacks of insufficient signal to noise ratio and phase errors. Mostly measured force and acceleration signals are directly used for impedance calculations without exploiting the advantage of impulse response.

This contribution aims to clarify the influences of special post-processing of the measurement data on the obtained impedance by means of deconvolution, bandpass filtering and time-windowing. Based on a measurement setup to study the prediction of the sound radiation of a small structure-borne sound source, the signal processing for the measured impedances will be presented and discussed. Nevertheless, this method can be applied for airborne impulse responses and other transfer paths as well.

Motivation

Since every simulation and auralization of a virtual scenario requires measured data from the real world this data should be as representative for this material as possible [1]. All ongoing simulations depend on such a database and therefore spending more effort in the measurements will pay off in future. Measurements have to be carried out for boundaries, e.g., in terms of precise impedances.

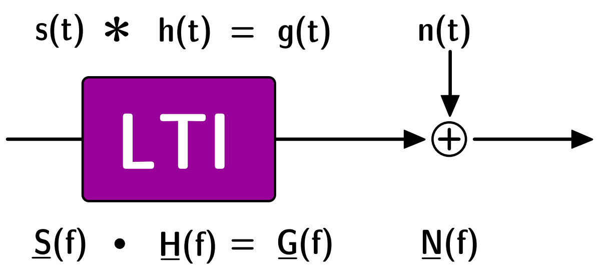

Figure 1: Block diagram describing the behavior of an LTI system and noise in time and frequency domain.

A special class of measurements will significantly benefit from the proposed method, i.e. if the result depends on very small differences of two or more measurements. As an example, consider a complex impedance $Z_{\mathrm{C}}(\omega)$ obtained from two measurements of a structure's impedance at the same point $Z_{\mathrm{B}}(\omega)$ and $Z_{\mathrm{B}}(\omega)$, but on one measurement another structure is coupled to it. The impedance of interest is calculated as

$$Z_{\mathrm{C}}(\omega) = Z_{\mathrm{A}}(\omega) - Z_{\mathrm{B}}(\omega). \tag{1}$$

In case the magnitude of Z_{A} and Z_{B} is much greater than Z_{C} it is obvious, that the measurements have to be very precise to obtain a reliable result. Furthermore, impedances are complex functions over frequency with the phase being a crucial part for ongoing calculations of the correct coupling of different components in a virtual scenario. Hence, understanding the influences on the obtained phase in the measurement and calculation process is required.

Measurement of LTI Systems

Basics and Contraints

Most acoustical systems are mostly assumed to be *linear time-invariant (LTI)* systems. Considering only these type of systems they can be completely described by their *impulse response (IR)* $h(t)$ or their complex transfer function $\underline{\mathrm{H}}(\omega)$ being the Fourier transform of $h(t)$. The response of this system $g(t)$ to any arbitrary input signal $s(t)$ can be calculated by considering the convolution theorem [2].

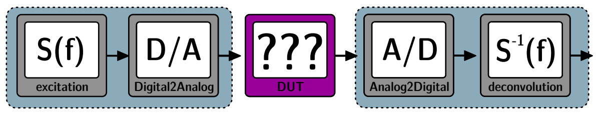

Figure 2: General measurement chain to obtain the IR of a device under test (DUT) with an excitation signal and by using deconvolution.

Every real world system is only LTI within very certain limits in terms of input amplitude and observation time. One has to always consider this constraints for simulations and measurements. The well known calculus from signal theory is only applicable for such LTI systems.

Additionally, every measured output of a system has an inherent noise term $n(t)$ as can be seen in Figure 1. The spectral composition and temporal structure of the noise can vary drastically from white gaussian noise to sinusoidal buzz noise.

Excitation Signal

In order to measure the IR of such a system, the system has to be externally excited by a signal $s(t)$ going out of the measurement software to the analog domain, through the LTI system and back to digital domain into the software. As can be seen in Figure 2 the IR is obtained by deconvolution, meaning by a spectral multiplication of the output signal with the inverse of the excitation signal $S_{\mathrm{comp}}(\omega) = \frac{1}{S(\omega)}$.

With deconvolution arises the problem of inherent noise in every measurement. Since the interest mostly lies in a certain frequency range, the excitation signal does not need to have a white amplitude spectrum and could therefore be seen as in Figure 3. The unwanted effect of amplifying noise outside the frequency range of interest can be handled by using appropriate bandpass filters. In general, two types of filters can be distinguished, one has causal IRs but influence on the phase response, the other one has no influence on the phase but a non-causal IR [3].

There exist various excitation signals as already shown in the extensive work by Müller and Massarani [4]. As shown in this work, the advantages of sweep outweigh and hence only sweeps are considered for such measurements. Additionally, the advantages of long excitation signals instead of averaging with shorter signals resulting in the same measurement time become obvious under the consideration of the proposed method involving time windows.

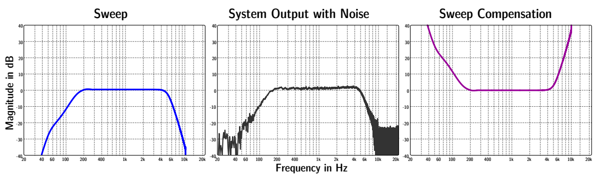

Figure 3: A band-limited excitation signal results in amplification of noise after deconvolution and therefore requires appropriate filtering.

Possible Scenarios

Transfer function measurements could be divided into four different types as illustrated in Figure 4. The *device under test (DUT)* can be a loudspeaker, a sensor or microphone, or any other acoustic system. The blue areas show the components completely known in terms of their IRs. In the last row, the typical case for structure-borne measurements with a special loudspeaker, a mechanical shaker, is shown. Usually, this component is not fully known and therefore not compensated. In structure-borne measurements two or more physical quantities are measured at the same time and the result of interest is always a ratio of two of the measured quantities.

Analysis and Optimization

By using an exponential sweep as excitation signal the measured IR of a system after deconvolution may look similar to Figure 5. The system has obviously non-linear behavior indicated by the purple impulses in the end. These are the IRs of certain harmonic distortions in the system, which can actually be measured by using exponential sweeps as excitation signals. Linear sweeps can also separate the linear and non-linear responses in

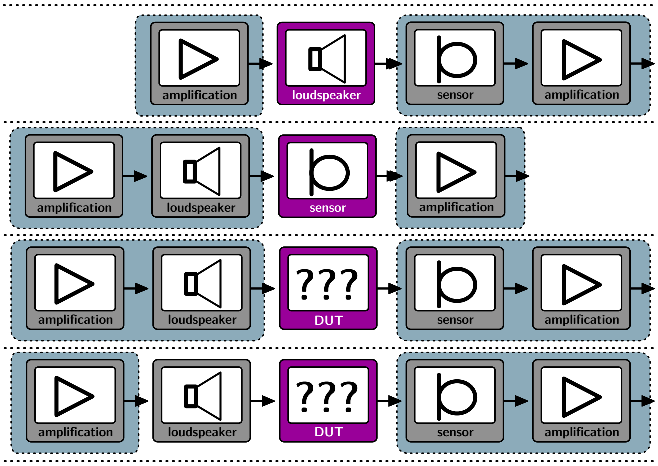

Figure 4: Four abstract measurement scenarios for transfer function or IR measurements. The purple blocks represent the device under test, and the blue areas show the subsystems of the measurement setup which can be fully compensated.

the IR, but one cannot directly identify the harmonic IRs. Noise and MLS are not capable of this separation, with these signals the noise floor will just increase. This indication of the amount of non-linearities is a measured quantity useful to analyze the measurement or the setup itself. In this case a loose and broken mechanical component in the measurement chain could be identified. Normally, the distortion is much lower than in this extreme example.

Another important aspect is the SNR of the measurement, which can be approximated by looking at the peak of the system response compared to the noise floor, being approximately 95 dB in this case. As such high quality is required, it has to be considered that fixed-point calculations and representations do not allow such precision. Although the measured input data for deconvolution usually is represented with 16 bit to 32 bit the ongoing calculations can significantly benefit from using high precision floating point representation.

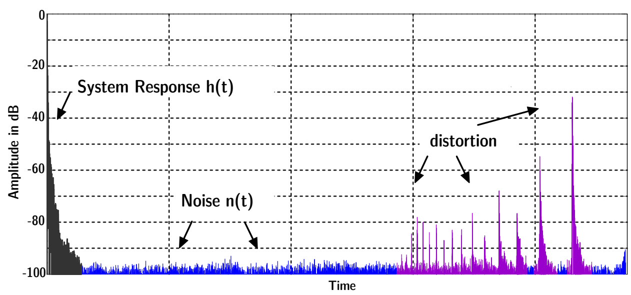

Figure 5: IR of a system with fairly good SNR and certain amount of distortion (smaller impulses in the end of the IR) – measured with an exponential sweep.

Figure 6 shows the spectral composition of different parts of a measured IR. As can be seen from the spectral information on the noise tail including the distortion part there is a noise problem below 100 Hz and the non-linearities have strong impact at higher frequencies.

To investigate the characteristic of the measurement

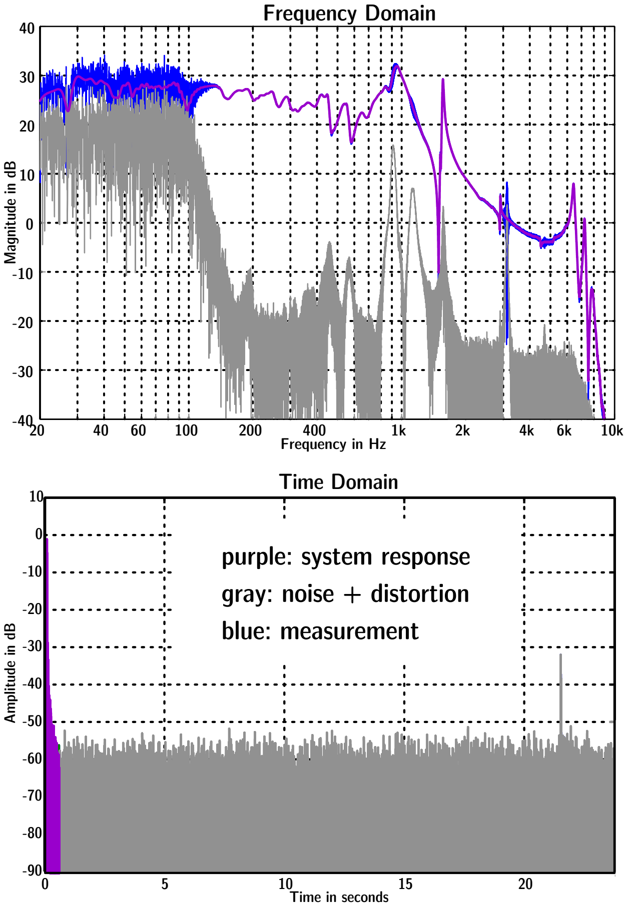

Figure 6: Relationship between the different parts of a measured IR of a real system (excitation: exp. sweep).

result in more detail, the IR can be analyzed in separate frequency bands, e.g. octave or third-octave bands as depicted in Figure 7. Here, a zerophase bandpass filter was used to obtain non-shifted IRs in the bands. The acausal part of the IR due to this filtering is not shown in this plot. In general, lower frequency bands (blue) show more decay than higher frequencies (green). In each band, the SNR and the transition of the system response into the noise floor can be analyzed. The aim is to achieve good SNR in all bands of interest. This band filtering is only used for analysis and does not have any filtering influence of the result.

In order to optimize the measurement according to a pre-measurement the following procedure is proposed in this paper. From the several measured IRs, odd behavior of components can be investigated and solved. The excitation signal should be reduced in resonances of measured sub-systems to allow an overall higher excitation. Sensor preamplifiers can then work well for all frequencies and settings are not anymore constrained by strong resonances. Finally, the sweep rate has to be adapted to the SNR in the observed frequency bands, to allow longer measurements in regions of bad SNR and shorter measurement time in regions of good SNR. As this method is obviously an iterative approach measurement software should be capable of optimizing a measurement itself. This is currently investigated in terms of measurement optimization algorithms at the

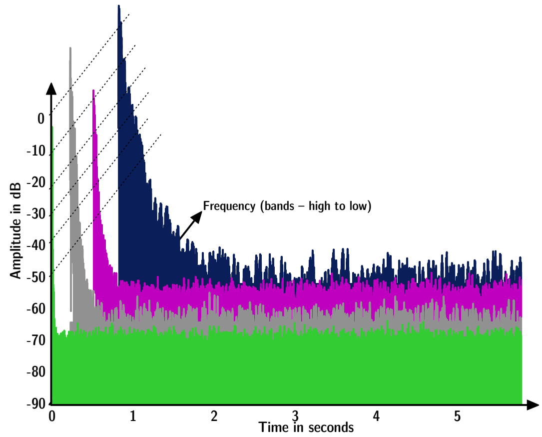

Figure 7: Band-filtered IR to investigate the SNR in each frequency band separately.

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Example – Impedance

The impedance measurements shown in the following belong to current research work by Lievens [5] where the described technique has been successfully applied. The mechanical impedance is defined as ratio of the force $F(\omega)$ and the velocity $v(\omega)$ at one position of interest,

$$Z_{mech}(\omega) = \frac{F(\omega)}{v(\omega)}. \tag{2}$$

Usually, these measurements are carried out using *impact hammer* measurements, lacking of precision and sufficient SNR. In cases mechanical shakers are used mostly noise or MLS are used as excitation signal without applying deconvolution to obtain IRs. Instead, the measured signals are transformed to frequency domain and divided by each other, mostly involving averaging. The applied averaging circumvents obtaining IRs.

Figure 8 shows impedance results for a given measurement setup. As can be seen, raw results are used to directly calculate the impedance, without applying the aforementioned averaging in a). In b) the optimum result is shown for demonstration purpose. Usually much higher precision is required, but such low noise measurements are not illustrating the discussed effects, therefore measurements of lower quality are used to show the effects and the optimum measurement result proves, that the applied techniques yield in refining the measurement result towards the optimum result. In c) the measured IRs are time windowed, transformed to frequency domain and divided by each other. It can be clearly seen, that the raw result shows noise in the curve and is not giving any precise information on the impedance for frequencies over 2 kHz. The proposed method significantly improves the result without departing from the correct result.

The time window used is a right-sided time window beginning at 0.3 s and ending at 0.5 s. The IRs remain the same up to the beginning of the window and are then faded out over the window length. Values after the ending time, are all set to zero, suppressing the noise

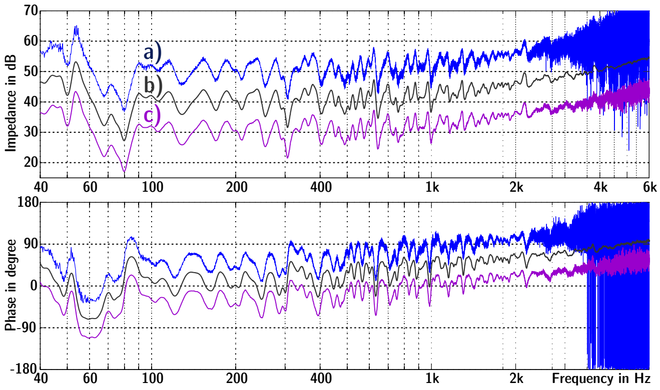

Figure 8: Calculated impedance – a) raw input data, b) optimum result, very low noise, c) proposed method using right-sided windowed IRs (results shifted for demonstration)

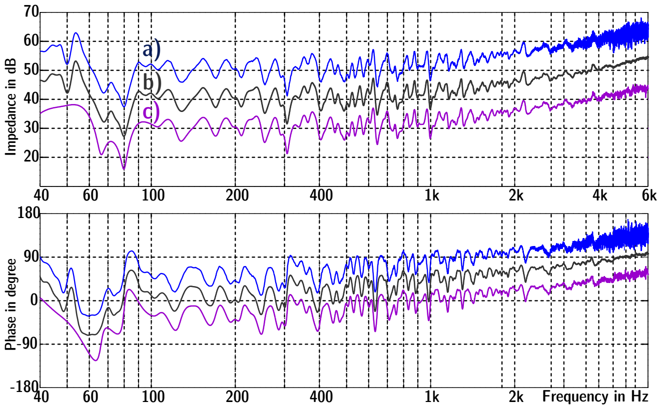

Figure 9: Calculated impedance – a) proposed method, b) optimum result, very low noise, c) drastically windowed impedance IR (results shifted for demonstration)

floor and non-linearities. The actual window function used for the fading part is a HANN window in this case. As the time constants used for this method are mostly high compared to other applications involving window functions the known side effects (e.g. smoothing) are much lower for this method. A major influence of type of the original window function used has not been observed by the authors.

Depending on the particular measurement another advantage of considering results as IRs could arise. In this case both the impedance and the admittance – the inverse of the impedance – are causal and stable IRs (real and passive systems). By looking at z-plane representation this means that both do not have zeros in outside the unity-circle as they would be transformed to poles by taking the inverse, which would than result in unstable systems. Such systems are known as minimum-phase systems (without non-minimumphase zeros) [3] and have short IRs. Therefore, time windows with much lower time constants could be applied.

Figure 9 demonstrates this approach, where a) is the already optimized result from Figure 8 c), b) is again the optimum result and c) the frequency domain representation of the time windowed impedance IRs. Here, the window starts at 0.02 s and ends at 0.03 s and is symmetric concerning the y-axis. A symmetric window is required since noise arises for low and high frequencies after division of F and v, analog to the effect described above for deconvolution. This noise must be filtered before applying a time window. As the phase of the result should remain unchanged, zerophase filters have to be applied, resulting in the aforementioned acausal behavior. Nevertheless, the original system response stays causal, just the filters introduce this effect.

The calculated result clearly shows the advantage for higher frequencies, meaning that the usable frequency range could be extended up to higher frequencies. But for low frequencies below approx. 70 Hz deviations from the desired optimum result become obvious. This effect is already known and the frequency limitation corresponds to the time constants of time windowing. As mentioned before, low frequencies, in general, correspond to longer required observation times. Frequency dependent time-windowing seems to solve this trade-off in future, i.e. longer time constants for lower frequencies and shorter time constants for higher frequencies, depending on the particular measurement. This windowing technique could be used for the initial IRs of force and velocity as well.

Conclusion

The concept of examining every measurement with an excitation signal as a measurement of IRs has proved to be of great advantage. In particular, measurements for input data for ongoing simulations and auralization can benefit from this technique. In cases the desired result is obtained by evaluating small differences of other measurement results, the original results have to be very precise. Using a-priori information on LTI systems by separating the system response from the noise, improves the measurement results in many ways. A mechanical impedance has been used as an example for this technique. Nevertheless, this approach is applicable for all kinds of IR or transfer function measurements.

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