Measurement Technique for High Quality Input Data for Auralization

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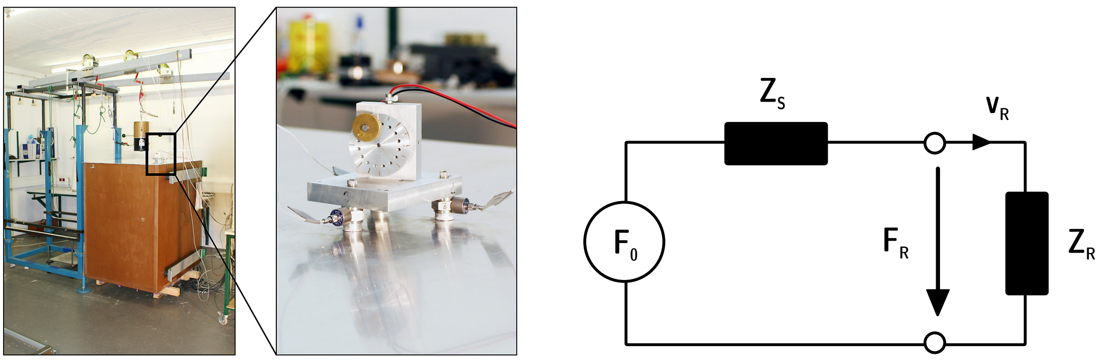
Abstract. *Auralization of structure-borne and airborne noise problems contributes to understanding of sound transmission in a significant way. One expects the auralization result to be as close to the real world result as possible. In hybrid models where measurement and simulation data are both used to generate the final audible result, the measurement data obviously has to be very precise. Even highly accurate numeric simulations require measured 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 fails due to 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 responses
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.

Keywords: *measurement technique, signal processing, impulse response, time windows, filters*

1. MOTIVATION

Simulations and auralization of virtual scenarios require measured data from the real world, such as coefficients of absorption or impedances. For precise results in simulation or auralization, the measured data should be as representative for a material or a subsystem as possible [1]. It is then useful to build a database for this data and spending more effort on the measurements undertaken to obtain the data will pay off in the future.



(a) (b)

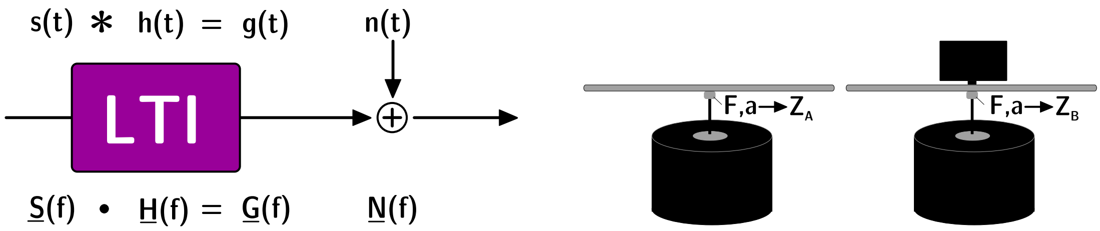
Figure 1. a) Measurement setup consisting of a small machine standing on a floor that radiates in the room below. The small motor with its three feet is shown in a zoomed view. b) Coupling situation of a real source and a receiver using impedances.

The motivation of this investigation can be explained by the measurement scenario as depicted in Figure 1(a). It shows a scaled model of a simplified machine standing on a floor radiating in a room below. In order to simulate and auralize this situation based on independent measurements of the important subsystems – separate measurements of the source (small motor) and the receiver (floor with room) – precise impedances of source and receiver at the coupling points are required. Figure 1(b) shows the relation between the force F_{R} when the source with an impedance $\underline{Z}_{\mathrm{S}}$ is connected to a particular structure with the impedance $\underline{Z}_{\mathrm{R}}$ and the blocked force of the source F_0. The blocked force is a source parameter and is measured by connecting the source to a very high, ideally infinite impedance. In the present study, only one degree of freedom and only one foot of the machine is considered. Of course, this could be modeled by using free velocity of the source as well. At this point, the measurement of the two impedances fully describes the coupling situation.

To consider the extent of the variation of source impedance as a function of structural coupling, the impedance of a source has to be measured in its installed condition (in-situ) as well. This measurement is crucial and it becomes obvious in the following discussion why such high quality measurement data is required. Normal measurements of impedances can of course also benefit from this approach. Figure 2(b) shows a schematic diagram of the setup that is used to measure the source impedance in its coupled condition. The impedance of the uncoupled receiving plate $\underline{Z}_{\mathrm{A}}$ and the impedance of the source and the plate together, the coupled impedance, $\underline{Z}_{\mathrm{B}}$ are measured separately. By subtraction, the in-situ source impedance can be calculated as follows,

$$\underline{Z}_{\mathrm{S,in-situ}}(\omega) = \underline{Z}_{\mathrm{B}}(\omega) - \underline{Z}_{\mathrm{A}}(\omega). \tag{1}$$

The result significantly benefits from the proposed method as it depends on very small differences of two measurements. In case that the magnitude of $\underline{Z}_{\mathrm{A}}$ and $\underline{Z}_{\mathrm{B}}$ is in the same range is evident that the measurements have to be very precise and with low noise 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 phase in the measurement and calculation process is required. In this context it has to be considered that structure-borne point impedances of passive systems have a phase in the range from -90 degree to $+90$ degree only.



(a) (b)

Figure 2. a) Block diagram describing the behavior of an LTI system and noise in time and frequency domain. b) Schematic diagram of the measurement setup to investigate the source impedance.

2. MEASUREMENT PROCEDURE

2.1 Basics

Usually acoustical systems can be assumed to be *linear time-invariant (LTI)*. Considering only this type of systems they can be completely described by their *impulse response (IR)* $h(t)$ or their complex transfer function $\underline{\mathrm{H}}(\omega)$ as 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.

Even assuming real systems to be LTI this applies only within certain limits in terms of input amplitude, observation time and the maximum allowable error. These constraints must therefore be considered in measurements because the well known calculus is only valid for LTI systems. Additionally, every measured output of a system has an inherent noise term $n(t)$ as can be seen in Figure 2(a). The spectral composition and temporal structure of the noise can vary drastically from white gaussian noise to buzz noise.

2.2 Excitation Signal

In order to measure the IR of 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 3 the IR is obtained by deconvolution, this is by spectral multiplication of the output signal with the inverse of the excitation signal, giving the compensation function $S_{\mathrm{comp}}(\omega) = \frac{1}{S(\omega)}$.

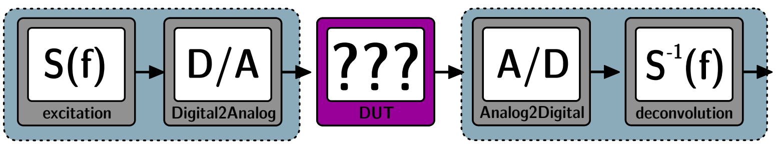


Figure 3. Signal flow graph to measure the impulse response of an LTI device under test (DUT) involving deconvolution technique.

With deconvolution the problem of inherent noise in every measurement arrises. 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 given in terms of a sweep as shown in Figure 4. 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 [2]. Excitation signals with a white spectrum do not directly require this filtering. But everytime filters are involved in the measurement analysis the filter types have to be considered.

There exist various excitation signals as already shown in the extensive work by MÜLLER and MASSARANI [3]. As shown in this work, the advantages of sweep outweigh and hence only sweeps are considered for such measurements for the present work. It has to be mentioned that Maximum Length Sequences have been used in the past for transfer functions measurements due to shorter calculation times as shown by VORLÄNDER [4]. As personal computers now are capable of deconvolving multichannel measurement signals within seconds this advantage of calculation time is negligible. Nevertheless, MLS are often just used to substitute white noise, without using the possibility to calculate impulse responses. Additionally, the advantages of long excitation signals instead of averaging with shorter signals resulting in the same overall measurement duration become apparent under the consideration of the proposed method involving time windows. A long sweep measurement is more suitable if the system is slightly varying over time during a measurement [3, 5]. Also this measurement results in an impulse response of the same size as the excitation signal, but the measurement noise is spread over time. Both methods, averaging and long measurement, will capture the same noise energy as the overall measurement time is the same. But the amplitude of the noise in the impulse response with a long sweep is lower, which is of great advantage as the information lies only in a particular part of the measured impulse response.

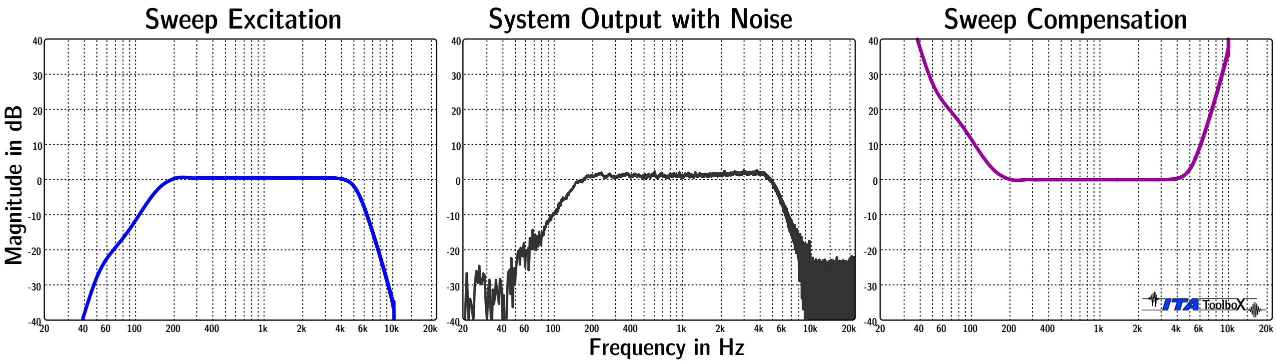


Figure 4. Band-limited excitation sweep (blue), system response to sweep including measurement noise (gray), compensation signal used to calculate the impulse response by deconvolution (purple).

2.3 Possible Scenarios

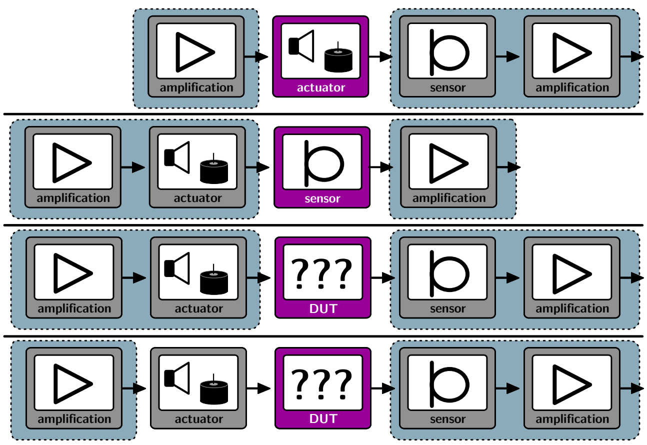


Figure 5. Four typical measurement scenarios for transfer function or impulse response measurements of LTI systems. The purple blocks represent the device under test (DUT), and the blue areas show the subsystems of the measurement setup which can be fully compensated in terms of their own transfer functions or impulse responses.

Transfer function measurements could be divided into four different types as illustrated in Figure 5. 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. Due to this division, the transfer function of the actuator is always inherently compensated.

3. ANALYSIS

By using an exponential sweep as excitation signal the measured IR of a system after deconvolution may look similar to Figure 6. 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 as already shown by FARINA [6]. Linear sweeps can also separate the linear and non-linear responses in the IR, but one cannot directly identify the harmonic IRs. Other known excitation signals such as Noise and MLS are not capable of this separation, with these signals the noise floor will just increase.

The amount of non-linearities, as measured, can be used to analyze the measurement or the setup itself, e.g. a loose or broken mechanical component in the measurement chain could be identified. Normally, the distortion is much lower than in the extreme example given by Figure 6.

Another important aspect is the signal-to-noise-ratio (SNR) of the measurement, which can be approximated by looking at the peak of the system response compared to the noise floor, being approximately $95\,\mathrm{dB}$ in this case. The definition of SNR given here differs from literature since signal energy should be compared to noise energy. This could be done by integrating the IR in the time limits where the system response is assumed and the noise energy by integration of the remaining part of the IR. When 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 $16\,\mathrm{bit}$ to $32\,\mathrm{bit}$ the ongoing calculations can significantly benefit from using high precision floating point representation to avoid further significant quantization or discretization noise introduced by the signal processing itself.

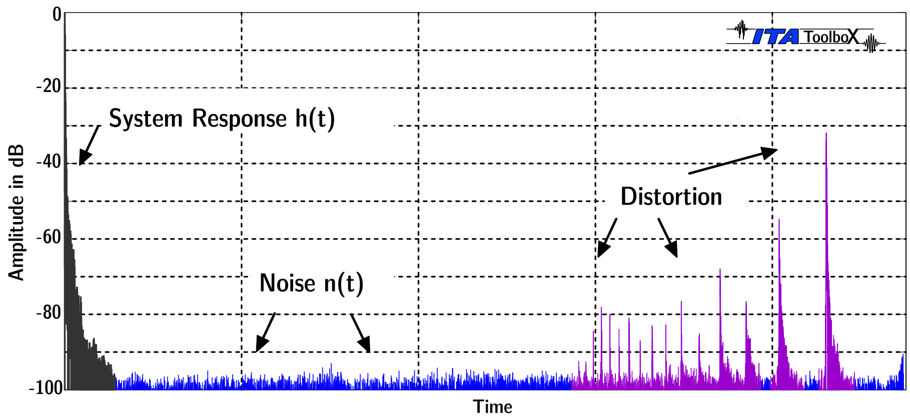


Figure 6. IR of a system with fairly good SNR and a certain amount of non-linearities (impulse responses in the end of the IR) – measured with an exponentially swept sine.

Figure 7 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 $100\,\mathrm{Hz}$ and the non-linearities have strong impact at higher frequencies above approx. $900\,\mathrm{Hz}$.

To investigate the characteristic of the measurement result in more detail, the IR can be analyzed in separate frequency bands, e.g. octave or third-octave bands as depicted in Figure 8. Here, zerophase bandpass filters should be used to obtain non-shifted band-filtered IRs. The acausal part of the IR due to this filtering is not shown in this plot. In general, lower frequency bands (blue) show a slower decay than higher frequencies (green). The same effect is observed by investigating the band-filtered impulse responses. 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 the analysis of the SNR in the frequency bands and to adapt the measurement setup and does not have an influence on the result.

In order to optimize the measurement preliminary measurements are indicated. In this preliminary measurements several measured IRs for all channels of interest are to be obtained and odd behavior of components can be investigated by means of the obtained IRs and solved. According to the preliminary measurements 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 no longer 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 low SNR and shorter

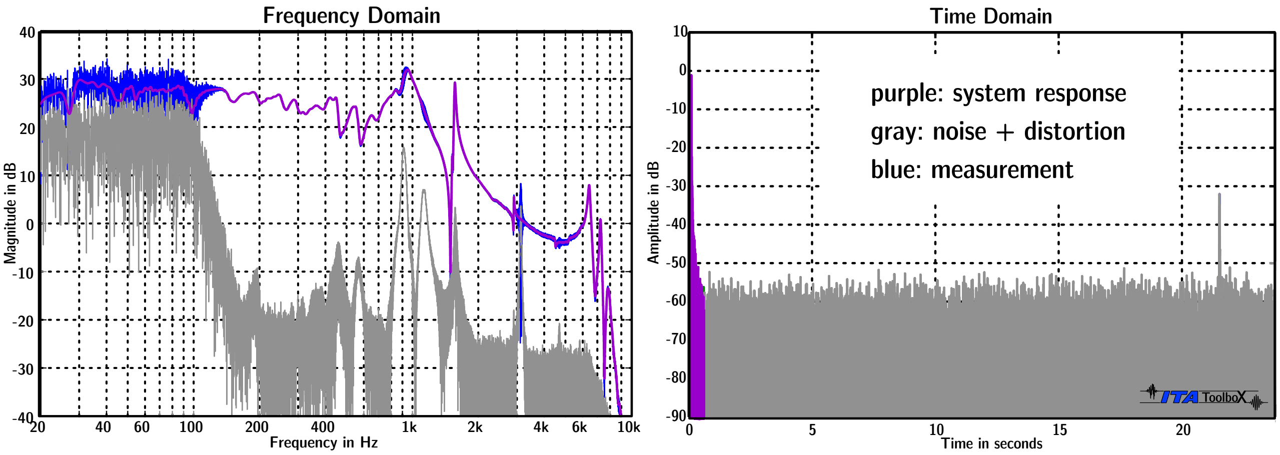


Figure 7. Frequency response of different parts of a measured IR of a real system (excitation: exponentially swept sine). The useful information on the linear behavior of the system lies in the beginning of the impulse response.

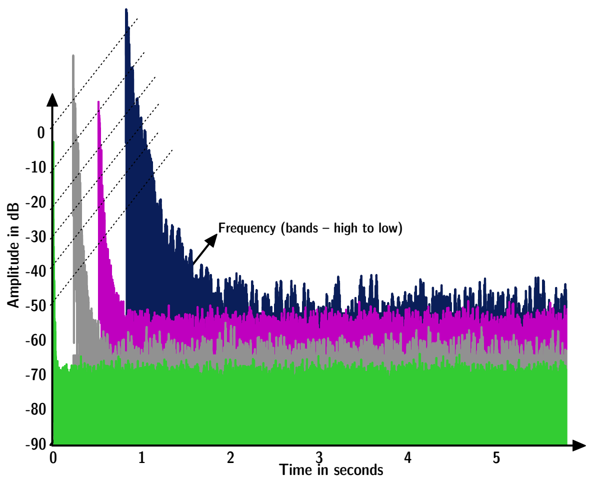


Figure 8. Band-filtered IR to investigate the SNR in each frequency band separately. Low frequency bands (blue) show in general slower decay as higher frequency bands (green).

measurement time in regions of high SNR. As this method is certainly an iterative approach, measurement software should be capable of optimizing a measurement itself. This is currently studied in terms of measurement optimization algorithms at the Institute of Technical Acoustics in Aachen.

4. STRUCTURE-BORNE IMPEDANCE

The impedance measurements shown in the following belong to current research work by LIEVENS and DIETRICH [7] where the described technique has been successfully applied. The mechanical impedance is defined as the ratio of the force $F(\omega)$ and the velocity $v(\omega)$ at one position of interest,

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

Often, these measurements are carried out using *impact hammer* measurements, lacking precision and sufficient SNR. The authors are aware that impact hammer measurements are important in industry, when quick and cost optimized results are required. For auralization purposes and in order to obtain material information for ongoing simulations these measurements are not sufficiently precise. When mechanical shakers are used, normally noise or MLS are used as excitation signal without applying the deconvolution technique in order to obtain the IRs. Instead, the measured signals are transformed to frequency domain and divided by each other, often involving averaging of several frequency functions obtained by short-time Fourier transform. The applied averaging circumvents obtaining IRs as the detailed phase information required for the transformation is lost.

Figure 9 shows impedance results for a given measurement setup. As can be seen, raw values are used to directly calculate the impedance (without applying the aforementioned averaging) in a). Curve b) shows, for demonstration purpose only, the optimum result obtained by very long measurements signals. Usually much higher precision is required, but such low noise measurements do not illustrate the discussed effects. Therefore measurements of lower quality are used to show the positive effects of the proposed method. The optimum measurement result proves, that the applied technique improves the measurement result towards the optimum result without applying a simple smoothing in the frequency domain. 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 does not give any precise information on the impedance for frequencies over $2\,\mathrm{kHz}$.

Nevertheless such information is important for auditory perception of auralized data, both for perception of timbre and source localization. Correct localization of sound sources in auralized situations for example depends on correct phase information at lower frequencies and precise magnitudes at frequencies higher than 2 kHz [1]. The proposed method significantly improves the result for the measurements without departing from the correct result, providing therefore a mean to enhance the quality of auralized data.

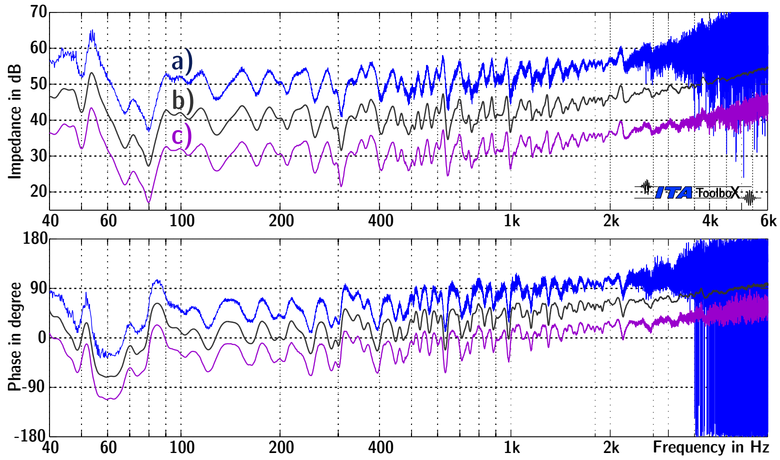


Figure 9. Calculated impedance – a) raw input data, b) optimum result, very low noise, c) proposed method using right-sided windowed IRs (results shifted for demonstration)

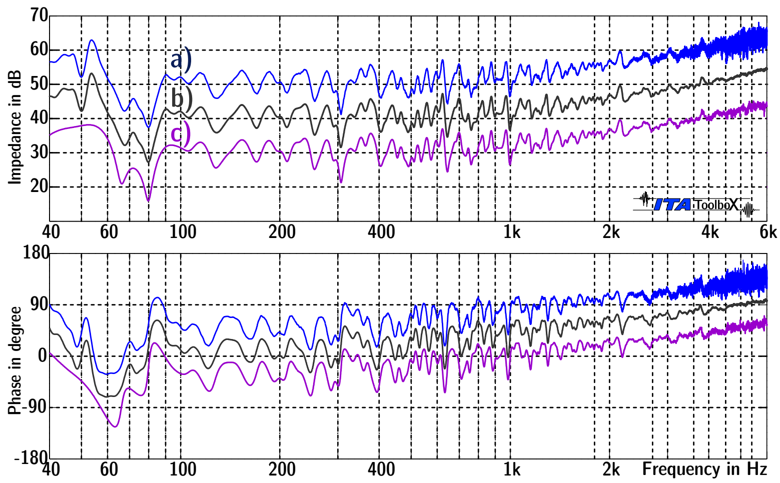


Figure 10. Calculated impedance – a) proposed method, b) optimum result, very low noise, c) drastically windowed impedance IR (results shifted for demonstration)

The time window used on the IR is a right-sided time window that starts at $0.3\,\mathrm{s}$ and ends at $0.5\,\mathrm{s}$. 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 floor and non-linearities. The actual window function used for the fading part is a HANN window. 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, loosing frequency information) are much lower for this method. Influences due to the type of the original window function have not been observed by the authors so far. This can be explained by the fact, that a large section – before the window is actually applied – of the important part of the impulse response remains unaffected.

Depending on the particular measurement another advantage of considering results as IRs could arise. Here, both the impedance and the admittance – the inverse of the impedance – are causal and stable IRs (real and passive systems). By looking at the z-plane representation this means that both do not have zeros outside the unit-circle as they would be transformed to poles by taking the inverse, which would then result in unstable systems. Such systems are known as minimum-phase systems (without non-minimumphase zeros) [2] and have short IRs. Therefore, time windows with much lower time constants could be applied.

Figure 10 demonstrates this approach, where a) is the already optimized result from Figure 9 c), b) is again the optimum result and c) the frequency domain representation of the time windowed impedance IRs. Here, the window starts at $0.02\,\mathrm{s}$ and ends at $0.03\,\mathrm{s}$ and is symmetric concerning the y-axis. A symmetric window is required since noise arises for low and high frequencies after division of F by 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. In other words, the band-limitation of the impedance results in an acausal impedance IR, and therefore symmetric time windows have to be used. Nevertheless, the original system response stays causal, just the filters introduce this effect.

The result obtained by application of the symmetric window 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. $70\,\mathrm{Hz}$ deviations from the desired optimum result become evident. 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 to assure optimum low noise input data.

5. CONCLUSION

The concept of examining every measurement with an excitation signal as a measurement of IRs proved to be of great advantage. In particular, measurements for input data for ongoing simulations and auralization can benefit from this technique. If the desired result is obtained by evaluating small differences of other measurement results, the original results have to be even more 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.

6. ACKNOWLEDGMENTS

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