Transfer function measurements of the Bassoon

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Introduction

The objective of this work is to determine a linear and time-invariant filter to synthesize the sound of a woodwind instrument in the room from the pressure inside its mouthpiece, by use of convolution.

For two dedicated observation points in a linear, time-invariant system a single-input-single-output transfer function can be obtained. For a brass instrument, this concept has been explored by Farina et al. [1], who have used the sound pressure recorded in anechoic conditions near the bell as input and determined a transfer function into the room. This single-input-single-output concept was applied to a woodwind instrument with more than one radiating opening in a more recent work [2]: Assuming linear time-invariant behaviour in the air column the input observation point was shifted towards the sound generation location, i.e. into the air column, way upstream of the first open tone hole near the mouthpiece inlet. In the absence of reflections, the measured mouthpiece pressure is proportional to the excitation volume velocity q, which can be obtained from a calibration measurement. With the instrument placed in a room, two pressure responses p_s and p_r are measured at a source and at a receiver position, respectively[1]. With the two transfer functions $h_s = p_s/q$ and $h_s = p_r/q$, the source-to-receiver transfer function is $h_{s,r} = h_s^{-1} * h_r$. With source location inside the reed mouthpiece and receiver location in the room (e.g. a dummy head microphone), $h_{s,r}$ has been called *reed to room* transfer function [2].

For the measurements, an impulsive sound source with bandwidth of approx. 2 kHz bandwidth was used. Up to a sampling frequency of 8 kHz an aurally convincing agreement between resynthesis and measurement was reported.

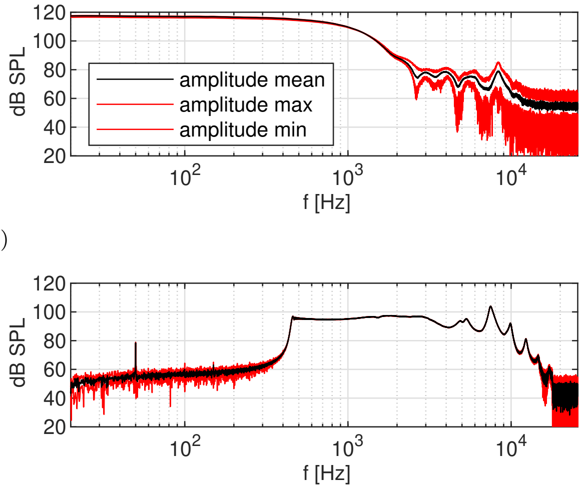
To extend these earlier results, the present work-in-progress paper presents an experiment to increase the bandwidth of the synthesis by combining the low frequency high-pressure-pulse excitation with an swept sine excitation from an electrodynamic driver.

Materials and Methods

The two driver mechanisms used to excite the room through the wind instrument air column are

- a solenoid valve connected to a pressure tank [2], and
- a horn driver (4550, BMS-Elektronik-GmbH, Hannover, Germany).

with a short pulse, and an exponential swept sine as digital input signals. For the calibration measurement a "semi-infinite tube" of d = 4 mm inner diameter was attached to provide a reflection free load.



(a)

(b)

Figure 1: Frequency response of two excitation mechanisms
a) pulse-like excitation provided by a solenoid valve,
b) sweep excitation provided by a horndriver

Excitation Bandwidth

The pressure responses of both excitation mechanisms are depicted in Fig. 1. To illustrate the repeatability, max and min values are shown for 24 (Fig. 1 a) and 4 (Fig. 1 b) repetitions, respectively. Assuming free-field conditions, the excitation volume flow q is linked to the measured pressure response by the characteristic impedance $\rho c/S$, where $S = \pi/4d^2$ is the cross sectional area of the tube.

Non-linearities

The convolution approach used here implies that the measured acoustical system is linear and time-invariant. Indeed, the electroacoustical driver used to excite the instrument near the mouthpiece may present highly non-linear behavior, and in a real playing situation, non-linear phenomena occur at the reed and especially in interaction with the resonator. However, we assume here (similarly to Farina's assumption [1]) that the non-linear part of the system is totally contained in its driver element. Hence, since the position of the so-called mouthpiece measurement is behind the driver (in the rear part of the reed at the connection to the resonator), the strongest non-linearities of the instrument may be excluded from the relationship between the mouthpiece sensor and the room sensor.

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[1] indeces $(.)_s$ and $(.)_r$ denote *source* and *receiver*

Having excluded the non-linear driving mechanism, it is worth also to discuss the linearity assumption for the rest of the system, i.e. the air column and the room: For the room, this will certainly be a good approximation, however this is not so clear for the instrument's air column. At high acoustic levels, pressure waves can distort as they travel through a waveguide. For example in the trumpet, While playing a smooth crescendo, the sound can suddenly to a "brassy" timbre, which in effect is the result of wave-steepening in the resonator.

In a conical woodwind instrument such as the bassoon, this effect is not expected [3]. We conducted a few confirmative experiments using pulse excitation at various, peak pressures ranging in the typical blowing pressure range between 2 kPa to 12 kPa, and in comparison the the trumpet we did not see a clear effect on the bassoon. However it is known that non-linear effects appear at tone holes [4, 5].

The linear approach use here will not be able to represent these effects.

Inversion

To calculate the transfer functions, inversions are performed on the measured signals using the MATLAB-software.

For pulse-like measurement, we use regularized time domain deconvolution, as suggested by Kirkeby & Nelson method [6]

$$h_c = [H^T H + \beta B^T B]^{-1} \dot{H}^T d, \tag{1}$$

where H is a Toeplitz matrix of the original measured time series of the excitation signal, β is a scalar regularization parameter B is a Toeplitz matrix of the excitation signal bandpassed in the regularization rang, d is the measured response and h_c is the desired impulse response.

For the swept-sine measurement, we use regularized frequency domain spectral inversion, suggested by Farina [7], as implemented in ita-toolbox[2] (function `ita_invert_spk_regularization.m`)

The two band-limited impulse responses where merged into one full-band impulse response using a Linkwitz-Riley filter of order 4, as implemented in the ita-toolbox (function `ita_filter_LiRi.m`). Linkwitz-Riley filters have been chosen for their ability to provide interference-free reconstruction at the crossover (synchronized phase responses).

Results and Discussion

The obtained full band reed-to-room impulse responses are convolved with the mouthpiece pressure to give synthesized room responses. These are compared to the real, measured room responses, for two different excitations:

- the technical excitation with the valve/horn driver,
- a *musical* excitation, with a muscian blowing a reed.

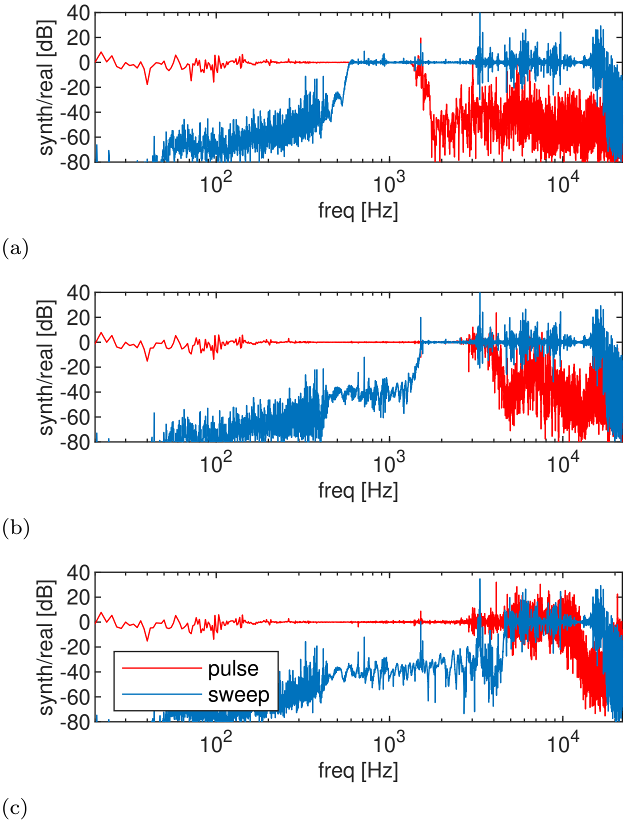

Figure 2: Full-band spectrum ratio of synthesized vs. real response, for technical excitation with a valve ("pulse") and a horndriver ("sweep") merged at three different crossover-frequencies a) $f_x \approx 0.8$ kHz, b) $f_x \approx 2$ kHz, c) $f_x \approx 6$ kHz

Within both experiments, the mouthpiece pressure and room pressure were recorded simultaneously, and between both experiments, the setup of the instrument in the room was left unchanged.

As a result we first present the full-band spectrum ratio of synthesized vs. real response, for the technical excitation a three different crossover-frequencies f_x see Fig. 2. In the preceeding inversions, the regularization ranges were adjusted to the intended usable bandwidth before applying the crossover. Note that the excessive reconstruction errors in the technical excitation experiment in Fig. 2 are not necessarily representative for the audio quality of the syntheses with musical excitation, because there we are looking at a sustained musical note with a harmonic structure. We can consider Fig. 2 as a worst-case graphical representation.

Next, we compare for the case of musical excitation the synthesized vs. the real measured room responses. Here, we provide a best-case representation by evaluating the harmonic sound spectrum as given by the averaged level of the partials resulting from short-term spectral analysis of the measured sound (Fig. 3).

Finally, for these harmonic spectra we show the deviation between the fullband syntheses with different crossover and the measured sound (Fig. 4). As a baseline the synthesis result is shown using a full-band impulse response computed only from the pulse measurement, without sweep excitation by the horndriver.

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[2] an open source acoustics toolbox for Matlab from the Institute for Technical Acoustics, RWTH Aachen, developed since 2007 [8], www.ita-toolbox.org

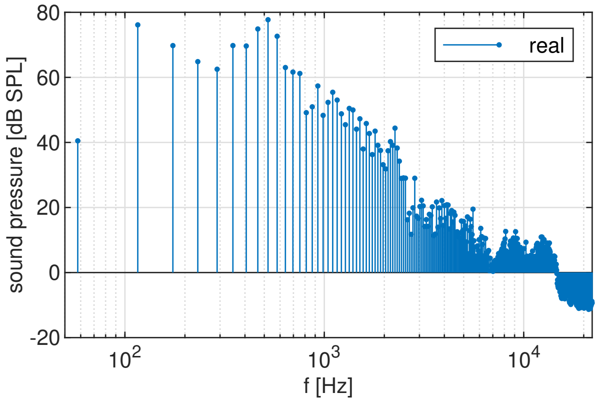

Figure 3: Harmonic spectrum of the room sound pressure during musical excitation (Bassoonist playing B$\flat$1, f_0 =58 Hz)

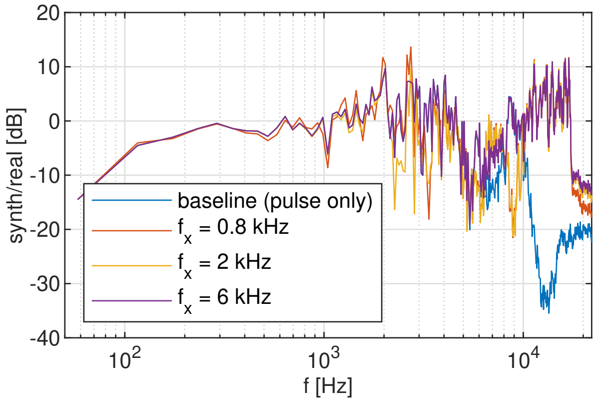

Figure 4: Level difference between measurement and synthesis for a real bassoon room sound (Fig. 3).
Syntheses from mouthpiece pressure convolved with full-band "reed-to-room" impulse responses joined from pulse and sweep excitation at three different crossover frequencies.
For comparison: pulse excitation only (baseline).

This illustrates the enhancement due to the additional high-frequency swept sine excitation over the original low-frequency pulse excitation, which is most clear above 10 kHz. It is interesting that for the lowest crossover between pulse and sweep excitation ($f_x \approx 0.8$ kHz) we do not see a substantial improvement of the synthesis until 10 kHz.

Conclusions

Interestingly, the general sound quality of the syntheses is already convincing, despite the fact that single partials deviate by up to 20 dB from 2 kHz onwards. In direct comparison with the measurement artificially sounding components can be identified in all syntheses to some extent. Using a hybrid approach with pulse and sweep improves the synthesis result significantly above 10 kHz. However, in the mid-high frequency range between 3 and 8 kHz the improvement is only slight. When blind-listening to the auralizations, the differences between the three different crossovers are not obvious. Possibly, it is insufficient SNR which leads to artificial, "whooshing" sound components. It occurs above 5 kHz, for sweep less than for pulse excitation. Temporally varying harshness in the attack is a key feature to the impression of naturalness, possibly due to non-linear effects. Such hypothesized non-linearities are lost in any of the syntheses.
To further improve the audio quality, alternatives to the linear convolution (e.g. [9]) should be considered.

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