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Input Impedance Measurements of Conical Acoustic Systems using the Two- Microphone Technique

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The two-microphone technique with a broadband excitation has been used for the measurement of various objects including straight conical waveguides and alto saxophones. A procedure has been developed whereby the noise excitation signal is pre-filtered by the inverse frequency response of the system under consideration. This approach helps reduce distortion and improves the SNR of the measurement. Results with and without pre-filtering are compared for different fingerings of an alto saxophone. The input impedance of a straight conical waveguide is compared with theory and discrepancies are analyzed. The input impedance of saxophones are evaluated with the transmission matrix approach and compared with measurements. A software environment for e fficient comparison and analysis of measurement data with theoretical calculations is presented. We also make a call for the sharing of raw measurement data among researchers to allow comparison of results obtained with different apparatus on similar objects to better quantify their accuracy.

1 Introduction

An accurate system for the measurement of acoustic impedance is essential for a musical acoustics laboratory interested in the study of wind instruments. Since 2005, we have been performing measurements using the two-microphone transfer function (TMTF) technique for the validation of input impedance calculations made with the transmission matrix (TM) method [1, 2], which is based on the discretization of instrument air columns into cylindrical and conical sections, with and without toneholes. The TMTF technique allows an input impedance to be determined from a single measurement using a broadband source signal. However, depending on the measurement probe and object characteristics, signal-to-noise ratio (SNR) problems can sometimes occur. The SNR may be improved by using a longer source signal but this increases the overall measurement time and risks the introduction of temperature variations over the duration of the measurement. This paper investigates alternative solutions to the SNR problem.

We begin with a presentation of our measurement system and data analysis procedure, emphasizing the solutions we developed to avoid distortion and improve the SNR of the measurement. Results of impedance measurements with TM calculations are compared for a straight conical waveguide, an alto saxophone without toneholes, and a complete alto saxophone. Finally, we introduce a software environment that we are developing for musical acoustic calculations and and make a call for the sharing of measurement data among researchers in musical acoustics.

2 Measurement system

The two-microphone transfer function technique, introduced by Seybert and Ross [4], makes use of a broadband signal and Fourier analysis to evaluate the input impedance of an object with a single measurement. It has also been described by Chung and Blaser [5] and extended by Chu [6] to include attenuation. Three pairs of microphones are necessary to cover frequencies from around 100 Hz to 4500 Hz. Precise measurements require the calibration of the gain and phase of each microphone relative to the first, a constant and known temperature inside the apparatus, and the knowledge of each microphone's exact location. Our apparatus consists of a compression driver (JBL 2426H) connected through a flexible hose to an aluminum tube of one-half inch internal diameter. The tube is terminated by a flange to which various adapters can be connected to hold the objects under study. The four microphones (Sennheiser KE4-211) are inserted in the side wall of the measurement probe in such a way that they are flush with the inside of the tube. They are respectively located at 3, 6, 18 and 39 cm from the flange. A more detailed description is available in [7].

2.1 Measurement procedure

A noise sequence is sent to the driver while the signals from the four microphones are recorded and then analysed to obtain the input impedance from the microphone transfer functions. Strong resonances in the object to be measured produce minima in these transfer functions that have a low signal-to-noise ratio, a situation that can be exacerbated if the power emitted by the source is low due to its own mechanical impedance coupled with the apparatus. As a possible solution to this problem, a procedure was developed whereby the noise excitation signal is pre-filtered by the inverse frequency response of the system under consideration.The idea is to estimate the power of the emitted signal as the average of the power spectral densities (PSD) from the four microphones obtained in a first measurement. The white noise is then passed through a filter with a frequency response inversely proportional to the emitted power to obtain a new source signal. Finally, the impedance data is acquired from a second measurement using the synthesized signal.

To illustrate the results obtained by that procedure, the signals recorded while measuring an open cone are analyzed. Figure 1 compares the average of the PSD of the recorded signals using the original noise and the pre-filtered noise; the large variations in the PSD almost completely disappear when using the filtered noise. Because there is less power in the filtered signal, the second measurement should be executed with a higher gain setting on the driver's amplifier.

Figure 2 displays the SNR of the signal from the first microphone, estimated from the magnitude squared coherence[1] as SNR $= \gamma_{xy}^2/(1-\gamma_{xy}^2)$, for both the white noise and the pre-filtered noise. The minima in the SNR occur at the same frequencies as the minima of the PSD of the same microphone. Improvement of the SNR can be better evaluated from Figure 3 showing the ratio of the two previous curves. Results are particularly satis-

[1] The magnitude squared coherence is calculated from the power spectral densities as $\gamma_{xy}^2 = |P_{xy}(f)|^2/(P_{xx}(f)P_{yy}(f))$, where x refers to the source signal and y to the recorded signal.

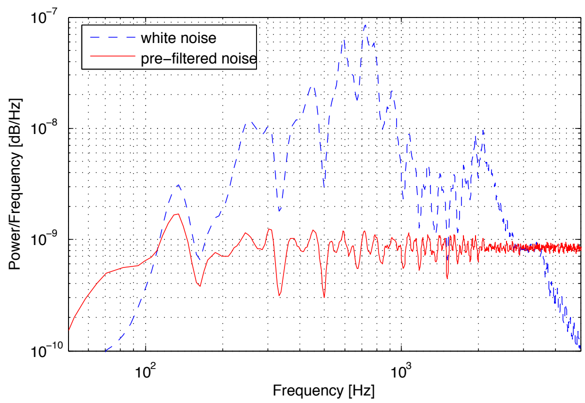

Figure 1: Comparison of the average PSD of the four microphones for the measurement of the open cone, with and without pre-filtering.

factory for the low frequencies (< 100 Hz) and the high frequencies (> 1000 Hz). However, improvement of the SNR at the minima, which determine the maxima of the impedance and where the error is maximum, is only limited. Figure 4 shows an instance where the pre-filtering procedure improves the evaluation of the transfer function. It is difficult to optimaly set the gain of the driver because no easy way of detecting the presence of distorsion in the recording has been found; manual inspection of the transfer functions show that smooth oscillations appears when the gain is too high.

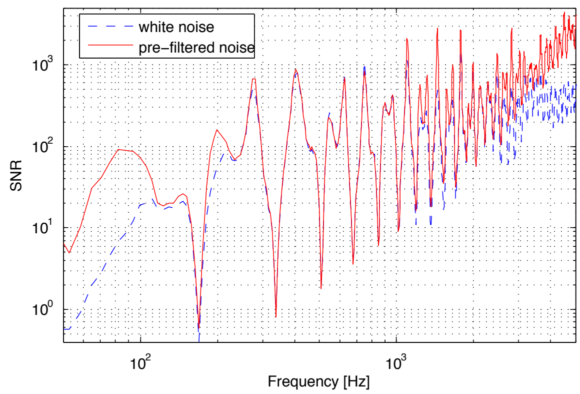

Figure 2: Comparison of the SNR of the signal from the first microphone for the measurement of the open cone, with and without pre-filtering.

2.2 Calibration

The gains and phases of each microphone must be calibrated relative to one another. Our calibration system consists of a pipe of 30 cm length and 2.54 cm diameter to which the JBL compression driver is attached. At the far end, the microphones are flush mounted through a metallic plate. The recorded sound field being identical for one-dimensional (1D) wave propagation, the transfer functions between the microphones are estimates of the calibration functions. The curves are smoothed with a low-pass filter. Verification of the accuracy of the calibration using an alternative approach is in progress. Errors in the calibration could be responsible for some

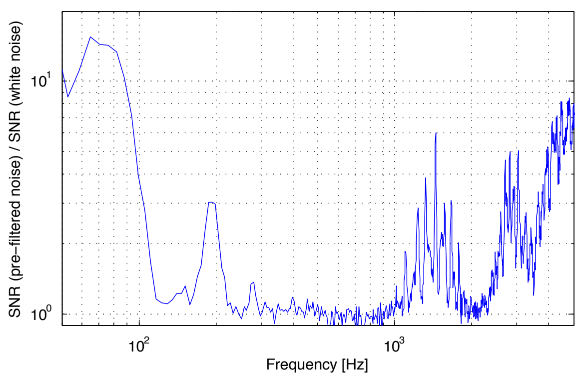

Figure 3: Ratio of the SNR of the signal from the first microphone for the measurement of the open cone (pre-filtered noise over white noise).

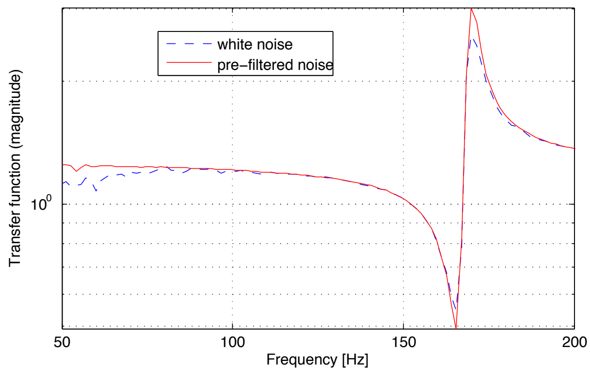

Figure 4: Comparison of the transfer functions between the first two microphones for the measurement of the open cone, with and without pre-filtering.

of the discrepancies that are reported later in this paper. To further refine the precision of the apparatus, the exact locations of the microphones are adjusted using a least square minimization algorithm that matches the theoretical and measured transfer functions for a rigid termination. The resulting microphone locations for the actual measurement session are 2.98 cm, 6.00 cm, 14.97 cm and 38.94 cm. Figure 5 displays a comparison between the theoretical transfer functions ($\mathrm{TF}_{1n} = \cosh(\Gamma x_n)/\cosh(\Gamma x_1), \Gamma = (1+i)\alpha + i\omega/c$) and the measured data (calibrated, with microphone positions adjusted). At a frequency of about 2875 Hz, the TF_{12} and TF_{13} measured transfer functions deviate slightly from theory. Note that the behavior of the TF_{13} and TF_{14} curves near this frequency is due to the simultaneous presence of a minima at both microphones.

2.3 Data analysis

In the context of musical acoustics, we are primarily interested in the magnitudes and frequencies of the maxima and minima of strongly resonant bodies. The low-frequency resonances tend to be narrow (Q around 25) and thus must be estimated with a fine frequency resolution. The resonances gradually become broad with increasing frequency because of greater thermoviscous and radiative losses; a coarser frequency analysis cap-

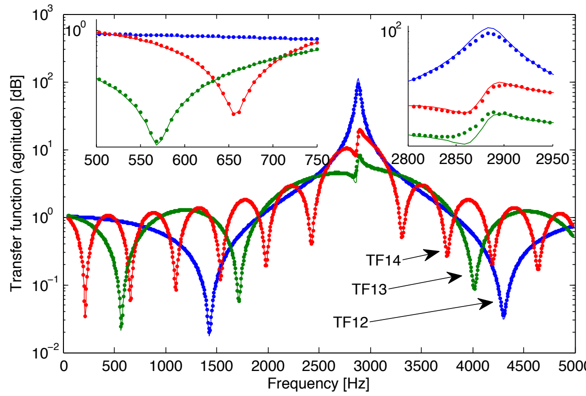

Figure 5: Transfer functions between microphones for a rigid termination: (-) theoretical (·) calibrated measured data.

tures these broad peaks correctly. A compromise must be found that provides a sufficiently fine resolution but enough averaging to reduce the variance in the results. Therefore, we implemented a procedure whereby the results are computed with successively narrower FFT sizes from low to high frequencies. Table 1 presents the FFT sizes associated with each frequency range and microphone pair. A single impedance vector is created by concatenating results across all the ranges. The readers are referred to the literature for the mathematical equations involved with the TMTF technique [4, 5, 6, 7].

Good quality low frequency impedance results are more difficult to obtain for a number of reasons: (1) the number of possible is smaller because the FFT sizes are larger; (2) the resonances being stronger, the minima are deeper and the SNR lower; and (3) less energy is radiated from the driver. Other types of source signals, such as swept sines, might help mitigate these problems.

Microphones	Spacing [cm]	Frequencies [Hz]	FFT size
1-4	36	50-250	32768
1-3	12	250-1000	16484
1-2	3	1000-4500	8192

Table 1: Frequency range and FFT size for each pair of microphone.

3 Comparison of TM calculations with measured data

In this section, results of theoretical calculations of the input impedance with measured data are compared for three acoustic systems of increasing complexity: a truncated cone, an alto saxophone body without toneholes, and an alto saxophone. From these cases we evaluate: (1) the accuracy of the truncated cone model, (2) the validity of the calculations for a complex system including a flaring bell, and (3) the validity of the calculations for the previous system with toneholes added. The concordance between theory and experiment is evaluated by comparing the frequency and magnitude of the first few impedance maxima. In the following tables, subscripts 1 and 2 refer respectively to the theoretical results and measurement data. A positive "interval" implies that the measured frequency is greater and a positive "ratio" means that the measured resonance has a greater magnitude.

The truncated cone, made of carbon fiber and epoxy resin with a 12.5 mm input diameter, a 63.1 mm output diameter, and a 965.2 mm length (angle is 3.00°), was measured for two boundary conditions: unflanged and flanged[2]. The theoretical input impedances are calculated with Kulik's [8] truncated cone model coupled to Dalmont and Nederveen [9] radiation impedance approximations. Table 2 provides a comparison of the frequencies and magnitudes of the first five impedance maxima for each of the two boundary conditions; the theoretical calculations are in good agreement with measurement data. The slight discrepancies in the resonance frequencies might easily be a consequence of small fabrication errors in the cone. However, the magnitudes of the impedance peaks are significantly lower than predicted, which is probably due either to measurement errors or inaccurate loss models. Simulations with greater losses improve the agreement of the higher (less pronounced) resonances but leave significant errors in the first few peaks.

Given that the pre-filtering procedure does not significantly improve the SNR of the transfer function minima, we suspect that our low-frequency magnitude results may be negatively impacted by noise contamination. A solution to this problem might be to use a different measurement approach. Nevertheless, the general agreement of measured data with theory appears to be sufficiently accurate for various musical acoustics purposes.

The alto saxophone body without toneholes, obtained directly from the Selmer factory in Indiana (USA), is not perfectly conical because of its two bends and its flared bell. Figure 6 displays the theoretical and measured input impedance curves of this instrument while Table 3 shows the comparison of the frequency and magnitude of the first six maxima. The calculations are in good agreement with the theory for frequencies below 2 kHz, although the theoretical curve inevitably contains errors because the instrument's geometry was difficult to measure and because the bends are not taken into account by the theory. The first few peaks, which are the most important in determining the playing frequency of the instrument, are very well estimated. In the high frequency range (above 2 kHz), the theoretical curve shows stronger and more defined resonances than the measured data, confirming that the TM calculation method is most accurate at low frequencies, as previously reported [2]. Experiments on precisely constructed prototypes of flaring instruments would help to determine the precision of various methods in evaluating the low frequency impedance and whether or not the unflanged pipe radiation model can be used with non-cylindrical instruments.

[2]The flange dimensions are 122 cm by 107 cm.

Harmonic	Interval [cents] $1200 \log_2 (f_2/f_1)$	Ratio [dB] $10 \log_{10} (\bar{Z}_2/\bar{Z}_1)$
Unflanged		
1	+6.1	−0.66
2	+0.3	−2.06
3	+1.9	−2.24
4	+1.0	−1.89
5	+1.8	−1.44
Flanged		
1	+4.5	−0.26
2	−1.7	−1.84
3	−1.6	−1.51
4	+3.2	−0.97
5	+1.6	−1.58

Table 2: Comparison of impedance maxima between measurement and calculation for the straight cone.

The third system, an alto saxophone model Selmer Super Action Series II with serial number 438024, was measured for the high register D fingering, a note based on the second resonance of the instrument. The open tonehole matrices are calculated by the formulation presented by Dalmont and Nederveen [9] with the influence of the pad taken into account by Nederveen's formulas [11]. Figure 7 displays the theoretical and measured input impedance curves for this fingering. Table 4 confirms that the calculations poorly estimate the measured input impedance of that particular system. The transmission matrices for the toneholes are very difficult to evaluate due to the complex nature of the geometry that includes a leather pad hanging at a small distance above the hole. Furthermore, as reported previously [9], the published formulas may not be valid for the short large-diameter toneholes of the saxophone because "the radiation field and the inner field are coupled" when the length of the hole is less than the radius.

4 Concluding remarks

Although the impedance measurement results are good, they are still not satisfactory in evaluating the maxima of the resonances. Other solutions than the pre-filtering approach may be explored to find an accurate and quick measurement method. A solution might be to use a swept-sine signal with amplitude adjusted to avoid distortion and with a frequency rate proportional to the SNR so that more time is spent where necessary.

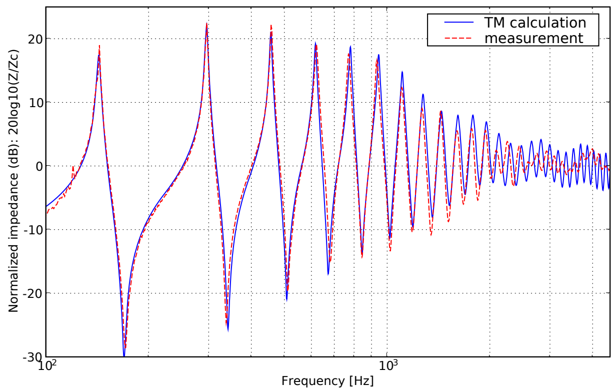

Figure 6: Theoretical and measured input impedance of an alto saxophone without toneholes.

Harmonic	Interval [cents]	Ratio [dB]
1	+6.6	+1.52
2	+0.3	+0.31
3	+2.2	+1.23
4	+10.1	-0.08
5	-21.1	-1.13
6	-21.3	-0.96

Table 3: Comparison of impedance maxima between measurement and calculation for the saxophone without toneholes.

Our study also suggests that more work is necessary for the TM calculation method to be useful at the design stage of woodwind instruments, particularly to take into account the behavior of the toneholes. Also, the use of a multimodal model to calculate the impedance of the flaring part of the instrument can possibly improve results in the high frequency range. Further identification of the weaknesses of actual simulation models require measuring the acoustic impedance of many instruments for all their fingerings including multiphonics and extended register.

The precise characterization of large-diameter toneholes with short chimney heights, typical of the saxophone, as well as register holes, including the influence of a key with pad, is an important aspect demanding future research.

5 The Wind Instrument Acoustic Toolkit

The theoretical results presented in this paper were obtained from a software environment developed by the first author[3]. It consists of a package of computer programs useful for researchers in wind instrument musical

[3] `http://www.music.mcgill.ca/musictech/caml/wiat/`

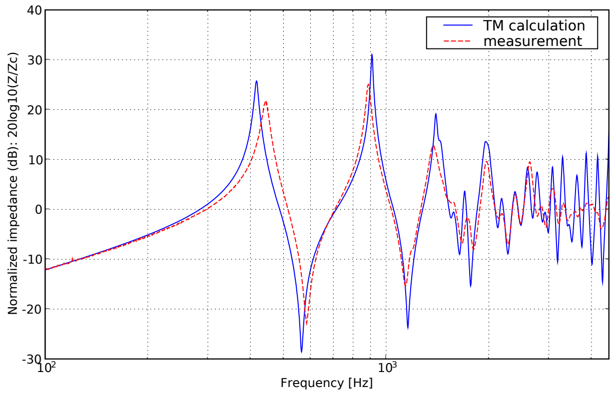

Figure 7: Theoretical and measured input impedance of an alto saxophone for the high D fingering.

Harmonic	Interval [cents]	Ratio [dB]
1	+106.3	-4.01
2	-43.5	-6.04
3	-29.3	-6.31

Table 4: Comparison of impedance maxima between measurement and calculation for the high D fingering.

acoustics and eventually, the wind instrument designer as well. It includes both simulation and measurement capabilities. Researchers from musical acoustics laboratories around the world are invited to collaborate in its development to keep it up to date with the latest theoretical results. The capabilities of the software include: (1) the calculation of linear input impedance from a geometry definition using either a one-dimensional transmission-matrix method or multimodal method; (2) optimization routines for the design of wind instrument prototypes; (3) processing of measurement data for comparison with simulation results; and (4) automatic generation of reports for easy incorporation into publications and/or web sites. Other suggestions are welcome.

6 Sharing measurement data

Experimental results obtained by different laboratories using various measurement techniques should be compared to confirm the validity of the results. Attempts to zoom in on published plots and to translate data into the computer is generally laborious and error prone. While it is always possible to contact authors to obtain their data, it is not easy to search for what is available. We propose the creation of a database for sharing acoustic measurement data. Databases of experimental data are common in some fields of research such as nucleic acids research and atomic physic. Looking at the NIST[4] web site reveals hundreds of databases for various properties of matter. It generally includes references to authors of the experiments and theoretical explanations where necessary. Unfortunately, it seems there is no such database in the field of acoustics yet. While it is obviously an important task demanding coordination and planning, it would benefit all the laboratories undertaking experimental and theoretical research in the field of acoustics. Researchers interested in the project are invited to contact the authors.

[4] `http://physics.nist.gov/PhysRefData/contents.html`

References

[1] G. R. Plitnik, W. J. Strong, "Numerical method for calculating input impedances of the oboe", *J. Acoust. Soc. Am. 65*, 816-825 (1979)

[2] R. Caussé, J. Kergomard, X. Lurton, "Input impedance of brass musical instruments - comparison between experimental and numerical models", *J. Acoust. Soc. Am. 75*, 241-254 (1984)

[3] M. V. Walstijn, J. M. Campbell, D. Sharp, "Wideband Measurement of the Acoustic Impedance of Tubular Objects", *Acust. Acta Acust. 91*, 590-604 (2005)

[4] A. F. Seybert, D. F. Ross, "Experimental determination of acoustic properties using a two-microphone random-excitation technique", *J. Acoust. Soc. Am. 61*, 1362-1370 (1977)

[5] J. Y. Chung, D. A. Blaser,"Transfer function method of measuring in-duct acoustic properties. I. Theory", *J. Acoust. Soc. Am. 68*, 907-913 (1980)

[6] W. T. Chu, "Extension of the two-microphone transfer function method for impedance tube measurements", *J. Acoust. Soc. Am.* 80, 347-348 (1986)

[7] A. Lefebvre, G. P. Scavone, J. Abel, A. Buckiewicz-Smith, "A Comparison of Impedance Measurements Using One and Two Microphones", Proceeding of the ISMA, Barcelona, Spain. (2007). This is available online at `http://www.music.mcgill.ca/~gary/papers.html`.

[8] Y. Kulik, "Transfer matrix of conical waveguides with any geometric parameters for increased precision in computer modeling", *J. Acoust. Soc. Am. 122*, EL179-EL184 (2007)

[9] J. Dalmont, C. J. Nederveen, V. Dubos, S. Ollivier, V. Meserette, E. te Sligte, "Experimental determination of the equivalent circuit of an open side hole: linear and non linear behaviour", *Acust. Acta Acust. 88*, 567-575 (2002)

[10] J. Dalmont, C. J. Nederveen, "Radiation impedance of tubes with different flanges: numerical and experimental investigations", *J. Sound Vib. 244*, 505-534 (2001)

[11] C. J. Nederveen, *Acoustical Aspects of Woodwind Instruments*. Revised edition. Northern Illinois University Press. (1998). (Chap. 3, pp. 64, eq. 38.6)