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Measurement, Estimation, and Modeling of Wind Instruments Using DSP Techniques

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SUMMARY

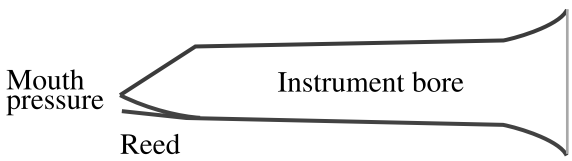
Physical modeling is a modern approach to musical acoustics and model-based sound synthesis of musical instruments. In this paper we apply digital signal processing (DSP) and physical modeling techniques to study the behavior of wind instruments. A DSP-based measurement system is used to obtain impulse responses for wave propagation, reflection, and radiation in the instrument bore, its terminations, and discontinuities. Inverse filtering techniques are then applied to estimate the pressure and volume velocity signals inside the instrument when played normally. The results can be used to estimate the parameters of a synthesis model.

SIGNAL PROCESSING MODELS OF WIND INSTRUMENTS

Digital waveguide modeling is a powerful synthesis technique for musical instruments that is based on a digital filter representation of wave propagation [1]. Related signal processing methods are also well suited to model-based analysis. For linear subsystems both time and frequency-domain processing are available and for nonlinear elements numeric time-domain processing is often the most useful approach. Figure 1 shows a generic wind instrument and its waveguide model (applicable to woodwinds and brass instruments). Signal propagation is assumed to be linear everywhere except in the reed portion.

A digital waveguide model implements a cylindrical bore section as a pair of lossless delay lines, one for each direction, so that the losses and dispersion are commuted to the termination filters (e.g., $R(z)$). This makes the model computationally very efficient [1]. Conical bores may also be modeled effectively [2], [3], and a more complex profile can be approximated as a cascade of cylindrical and conical sections. Finger holes are side branches in the waveguide model and the bell or any remarkable opening forms a combination of a reflection filter $R(z)$ and a radiation filter $T(z)$.

There is an inherent implementation problem in digital models of wave propagation due to the fixed unit delay length, which is determined by the sampling rate. The length of the tube and the positions of finger holes do not in general match a multiple of the discrete unit delay. Fractional delay (FD) approximation using interpolation solves this problem [4], [5]. We have shown various techniques to use FD filters in instrument models, including *fractional delay waveguide filters* [5], [6] that are able to simulate scattering junctions in arbitrary points between cylindrical and conical tube sections.



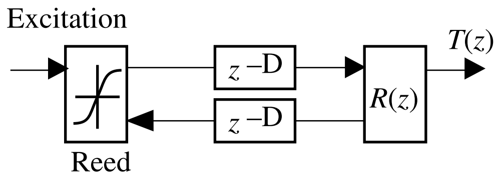


Fig. 1. *A generic wind instrument and its signal processing model.*

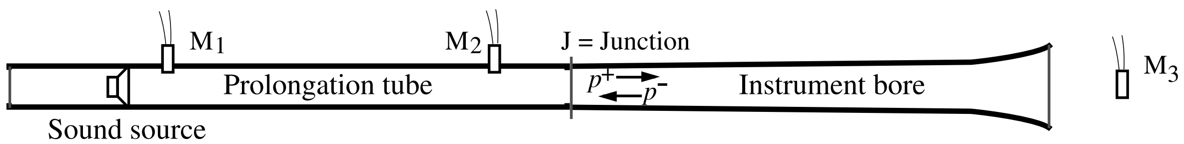


Fig. 2. *A system for the measurement of wave transmission, reflection, and radiation in a wind instrument where the mouthpiece has been replaced with a prolongation tube.*

The complexity of wind instrument modeling is related primarily to the reed portion that behaves highly nonlinearly. Since other parts of the instrument fairly well meet the condition of linearity and time-invariance, the excitation signal from the reed may be analyzed by inverse filtering the radiated signal by an estimated transfer function from the reed to the radiation field.

DSP-BASED MEASUREMENT TECHNIQUES

We have measured transfer functions and impulse responses due to wave propagation in a tube, as well as reflection and radiation at the termination, using DSP techniques in the following way (partly similar to [7]). Let us consider pressure signals in a linear and time-invariant system (see Fig. 2). The mouthpiece or reed portion of the instrument is replaced with a prolongation tube having the same diameter so that the tubes are acoustically matched. An approximation of a unit impulse is generated by a miniature transducer (a Walkman headphone driver) and its response is registered by miniature electret pressure microphones (Sennheiser K211-8) in positions M_1, M_2, and M_3. Averaging, e.g., for 100 repetitions, is used to improve the dynamic range of the results.

A computer program driving 16-bit stereo A/D and D/A converters controls the measurement process. A sampling rate of 22 kHz is used which yields a frequency band up to 10 kHz. Impulse responses are computed by deconvolution in the frequency domain:

$$h(n) = \mathbf{F}^{-1}\{\mathbf{F}[y(n)] \,/\, \mathbf{F}[x(n)]\} \tag{1}$$

where $x(n)$ is the reference signal (e.g., at the position M_1) and $y(n)$ is the response signal (e.g., at M_2), $\mathbf{F}$ is the discrete Fourier transform operator (implemented in practice using the Fast Fourier Transform) and $\mathbf{F}^{-1}$ is the inverse discrete Fourier transform operator. Care should be taken to avoid problems due to the circularity of discrete-time operations. Reference signal $x(n)$ should always have a relatively flat spectrum with no zeros.

The microphone pair M_1, M_2 is used to measure the wave transmission transfer function per unit length in the prolongation tube. A single microphone M_2 registers both the excitation traveling to and the signal returning from the instrument bore. When the path from M_2 to junction J is compensated off twice (by deconvolution), we obtain the reflection impulse response $h_R(n)$ at junction J. Similarly, transmission impulse response $h_T(n)$ from J to M_2 is computed from the pair M_2, M_3 by single compensation for the path from M_2 to J. Further analysis may be based on the two responses $h_R(n)$ and $h_T(n)$ with Fourier transforms $H_R(\omega)$ and $H_T(\omega)$, respectively.

A major issue with this procedure is to isolate the reference signal $x(n)$ from the response $y(n)$ (see Eq. (1)) in each case since they have been superimposed in the measured signals and may overlap in time. This separation must be done by applying careful time-domain windowing to proper signal portions before the deconvolution. In our measurements the main problem was the spreading of the excitation due to ringing of the transducer at its fundamental mechanical resonance frequency around 200 Hz. This spreading can be counteracted by using a long enough prolongation tube (for better temporal isolation) or by preprocessing the excitation to the transducer in order to shorten its output, e.g., by using the reciprocal $\mathbf{F}^{-1}\{1/\mathbf{F}[h(n)]\}$ of its impulse response $h(n)$.

As an example, Fig. 3 depicts the reflection impulse response $h_R(n)$ for a clarinet (note E_1, all holes closed) and the corresponding magnitude response $|H_R(\omega)|$. Notice in $h_R(n)$ the series of closed finger hole reflections as well as the main reflection from the open termination.

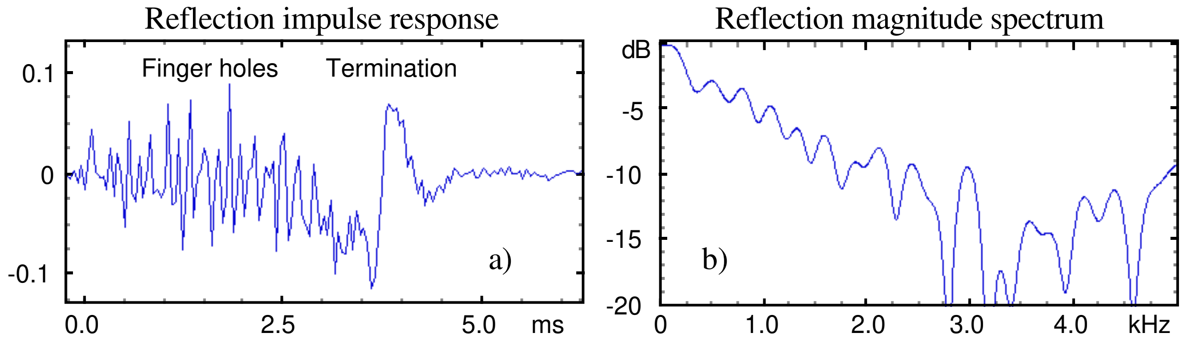


Fig. 3. *a) Reflection function of a clarinet as measured with the system of Fig. 2. b) The corresponding magnitude spectrum.*

The measured reflection response $h_R(n)$ may be utilized in estimating and designing the loop delays and the termination reflection filters of a waveguide synthesis model. The main reflection is used to find the total loop delay. In an efficient real-time synthesis system, every finger hole is not modeled but only the first open one. In such a case the closed finger hole reflections cannot be implemented. Only the overall loop transfer function is approximated in the reflection and radiation filters. The transmission response $h_T(n)$ may be utilized to design a digital filter for modeling the sound radiation.

Other interesting properties of the instrument can also be analyzed from $h_R(n)$ and $h_T(n)$. As an example, the acoustic impedance of the instrument bore, as seen from point J, may be derived. A pressure unit impulse $p(n)$ with Fourier transform $P(\omega)$, when driven into the bore, reflects back as $P(\omega)H_R(\omega)$. These are summed for the overall pressure signal at J. The acoustic impedance of the prolongation tube is $Z_0 = \rho c/A$, where ρ is the air density, c is the sound velocity, and A is the cross-sectional area of the tube. The excitation $P(\omega)$ will result in a volume velocity $P(\omega)/Z_0$ initially and $-P(\omega)H_R(\omega)/Z_0$ due to the reflected signal. Thus the impedance of the bore at junction J is the ratio of pressure and volume velocity, i.e.,

$$Z_J(\omega) = Z_0[1 + H_R(\omega)] / [1 - H_R(\omega)] \quad (2)$$

INVERSE FILTERING TECHNIQUES

Inverse filtering has long been used in speech research (see, e.g., [8]) and it is also applied in string instrument modeling [9] to compute the source excitation from the output when a reasonably good estimate of the source-to-output transfer function is available. The same technique is applied here to woodwind and brass instruments in order to estimate the traveling wave components inside the bore on the basis of the radiated sound.

Let us assume that we have measured the transfer functions $H_R(\omega)$ and $H_T(\omega)$ accurately. The transmission function $H_T(\omega)$ is useful as such since it may simply be applied to inverse filtering in order to compute the pressure wave at junction J traveling to the right, denoted by $P_J^+(\omega)$, from the measured pressure wave $P_3(\omega)$ at the microphone position M_3.

$$P_J^+(\omega) = P_3(\omega)/H_T(\omega) \quad (3)$$

We may derive one more signal relation for inverse filtering purposes. The pressure signal $P_J^-(\omega)$ at point J traveling to the left can be expressed as

$$P_J^-(\omega) = P_J^+(\omega)H_R(\omega) = P_3(\omega)H_R(\omega)/H_T(\omega) \quad (4)$$

The derivation shows that we may compute both pressure signal components at the entrance of the tube when we know the radiated sound pressure $p_3(n)$ and the two primary transfer functions. The sum of these components gives the total sound pressure. This may be used to check the validity and accuracy of the method whenever the sum pressure can be measured directly.

The above derivation is valid for the case where the bore has been impedance matched to a driving prolongation tube. One may ask whether the inverse filtering method is valid also during normal playing conditions when the highly nonlinear reed portion feeds the instrument bore. If it is assumed that the instrument is linear and time-invariant from point J to the right, then the inverse filtering is theoretically valid for all driving conditions, since the two primary transfer functions are not dependent on what will be connected to the left of point J. (We must also assume that there are no other signal paths from J to M_3. This is not true for, e.g., the flute.) This procedure—like deconvolution in general—is critical to the accuracy of the measurements and estimated transfer functions. There are various sources of inaccuracy that may disturb the final results. In addition to the ones mentioned above, one specific problem is how to guarantee that the transfer functions estimated from the instrument without a mouthpiece remain the same during normal playing conditions. Among the possible variations are the air temperature and related sound velocity, as well as changed humidity. The third factor is the possibility of nonlinearities in the bore due to high pressure levels.

CONCLUSIONS

In this paper we have presented a method to measure and analyze the behavior of wind instruments using the physical modeling approach and digital signal processing techniques. Besides being able to efficiently and precisely measure the transfer functions and impedance curves of a wind instrument, it is possible to apply inverse filtering—deconvolution by division in the frequency domain—to analyze the signal components close to the reed portion during normal playing and thus collect information on the nonlinear behavior of the reed. The results of the measurements and estimation techniques can be used to calibrate a parametric synthesis model.

ACKNOWLEDGMENTS

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