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Analysis of transients for brass instruments under playing conditions using multiple microphones

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This work investigates the development of a novel multiple microphone technique for analysis of the acoustical behaviour of brass instruments under playing conditions. In the current work, multiple microphones are deployed within a cylindrical section of the bore of a brass instrument. The technique allows for measurement of the instrument during playing, or while excited by a loudspeaker signal, and permits for a robust analysis of the transfer of acoustic energy between microphones. The long term goal of this study is to allow wave separation under playing conditions in order to better understand the nature of the lip reed and the strong coupling between player and instrument. Preliminary measurements of transient behaviour in brass instruments are presented and discussed.

1 Introduction

In brass instrument playing transients are highly important for a number of reasons. Players aim to achieve a clean sound when starting notes by 'hitting' the correct note quickly and easily; the ease with which a note can be started is used by the musician as an indicator as to the quality of the instrument being played. The starting transient is also known to be of great importance when determining the character of an instrument [1]. It is therefore important to understand fully the mechanics of the player's lips and the propagation of the pressure waves within the instrument during such transients.

Optical techniques have been used to study the motion of brass player's lips from as early as the 1940s by Martin [2]. This work led to a number of studies investigating the player's lips in more recent times [3, 4, 5, 6]. Optical techniques provide a very good platform for studying the motion of the player's lips but cannot give detailed information about exactly what is happening inside a brass instrument's tubing.

In this work the study of transients is furthered by using multiple microphones to keep track of overlapping forward and backward going waves within an instrument while it is being played. In using this technique no prior knowledge is required of the distances between microphones, speed of sound or acoustic propagation coefficients providing these remain constant throughout the experimental procedure; this is a result of the calibration being solely based on measurements of the time domain transfer function between the microphones. The technique relies on the time domain transfer function being measured for every combination of two microphones, which allows the forward and backward going waves to be determined at each microphone position in the time domain. Recent work includes the application of the technique to impedance measurement and bore reconstruction [7].

The present study involves using loudspeaker excitation for two calibrations: one for measuring the forward going wave and one for measuring the backward going wave in a cylindrical section of pipe. The cylindrical tube is then used as part of the bore of an orchestral horn and recordings are made when the horn is played by a human player. Recordings are then made with a human player. It is inevitable that playing the horn will gradually alter the speed of sound and ultimately invalidate the calibration. For this reason the player was restricted to playing a few short notes (of length 0.2 seconds) so that the conditions between the microphones remained unchanged during the recordings. In the longer term, adaptive filters may be used to adapt the calibration data to playing conditions in a more robust way as suggested by van Walstijn and de Sanctis [9, 10].

2 Experimental Set Up

The instrument used for this study was a Jiracek natural horn with an approximately conical Bb alto crook.

A straight piece of brass tubing of approximately 0.9m is inserted between the crook and the corpus; this extra section of tubing extends the bore to give a horn with a nominal pitch of F. The resulting bore closely approximates that obtained when the Jiracek F crook is used with the corpus as it is also designed using a conical section followed by a cylindrical section [11]. Three holes of diameter 1mm were drilled in the measurement duct, approximately 15cm apart, to allow microphones

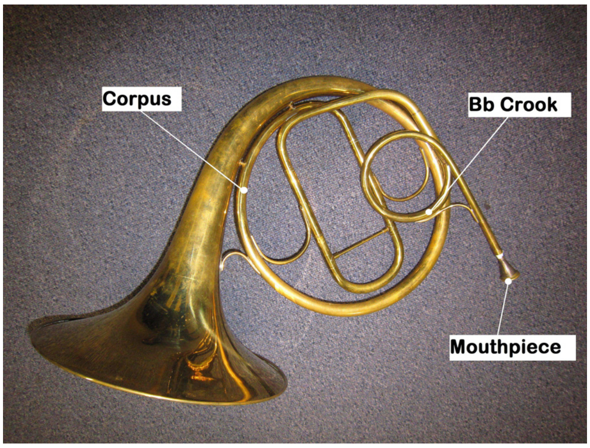

Figure 1: Horn with Bb crook

to detect the wave pressure at these points in the tube. High pressure 106B PCB Piezotronics microphones were placed at each hole. Figure 2 shows the horn with the measurement duct inserted, and the microphones secured to the duct at the positions of the holes by brass coupling blocks.

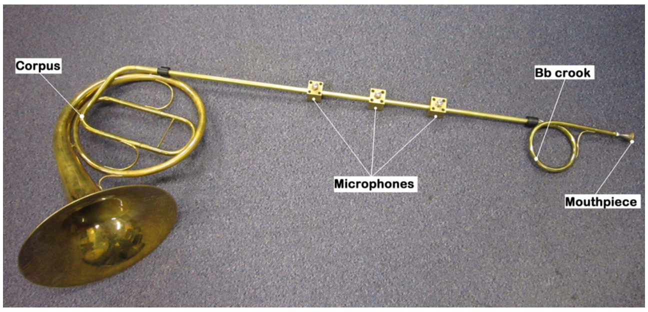

Figure 2: Horn with the measurement duct inserted and microphones in place

The signals from the microphones were directed through a Focusrite Saffire Pro 10 audio interface. For calibration an excitation signal was fed through a Denon hi-fi amplifier into a JBL compression driver loudspeaker. All analysis was done using post-processing in MATLAB.

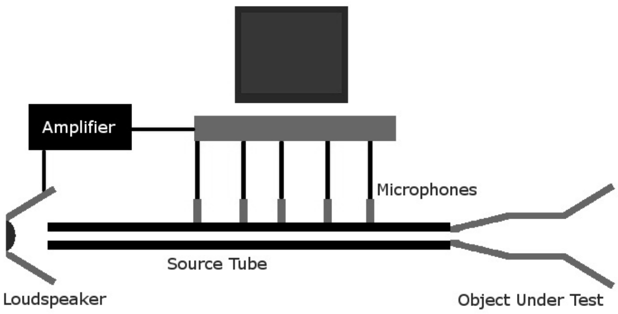

Figure 3: Multiple microphone wave separation apparatus, in the present study only three microphones were used

2.1 Experimental Procedure

The calibration procedure involved playing a logarithmic sine sweep with the loudspeaker sealed to the end of the cylindrical measurement duct and then repeating the procedure with the loudspeaker placed at the other end. Frequency domain division was used, along with time domain windowing, to obtain the time domain transfer functions for each combination of two microphones for both forward going and backward going waves [7].

After calibration the cylindrical duct was placed between the crook and corpus as shown in figure 2 and a skilled horn player was asked to play a number of short notes. The microphones recorded the pressure signals at the three positions along the measurement duct. These signals are analysed so that the forward and backward going waves can be determined.

3 Analysis

The three microphone signals recorded during the start of a note played are shown in Figure 4. It can be easily seen from this figure that there is a slight time delay between the initial transient in each signal corresponding to time taken for sound to travel between the microphones.

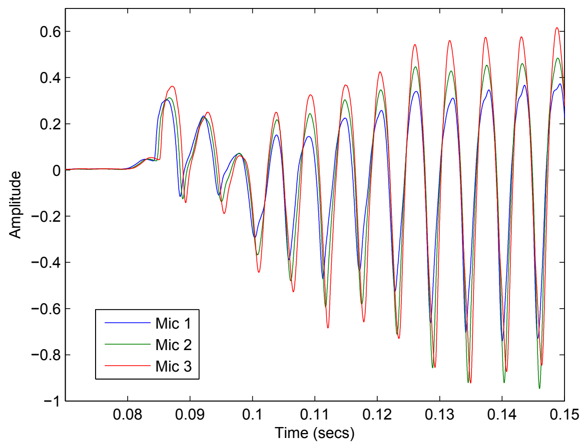

Figure 4: Three microphone pressure signals under playing conditions

The transfer functions that represent the transfer between microphones are expected, in the time domain, to have the form of an impulse delayed by a number of samples and low pass filtered due to losses in the tube and bandwidth of the experimental apparatus[7]. The time domain transfer functions measured for the current apparatus can be seen in Figure 5; this figure shows the transfer functions for the forward going wave. A set of time domain transfer functions for the backward going wave are also calculated but not plotted here as they look very similar, although they are not exactly the same due to the effects of different microphone gain ratios and non-simultaneous sampling in the soundcard.

The above analysis results in there being six time domain transfer functions as three microphones were used. These time domain transfer functions are then used to

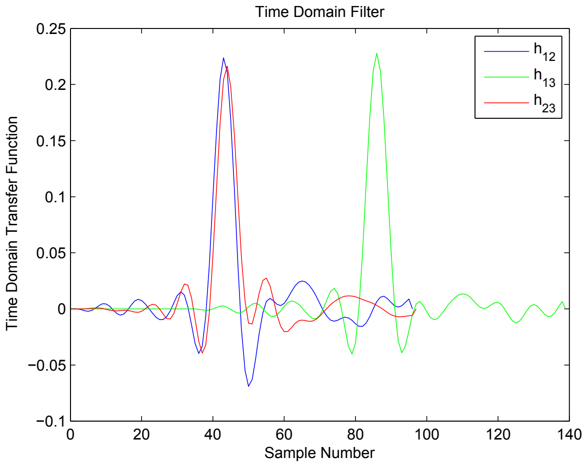

Figure 5: Time domain transfer functions for three microphones using recorded forward going calibration sinesweep

separate the forward and backward going waves. The pseudo MATLAB code for the N microphone wave separation algorithm is shown below. Here h_{ab} is the time domain transfer function (of length N_f samples) from microphone a to microphone b. $p_a^+(n)$ and $p_a^-(n)$ are the forward and backward going waves at the nth time sample for microphone a. g^+ is the guess matrix for the forward going waves, where $g^+(a,b)$ is the forward going wave at microphone b at the current time step predicted using the data from microphone number a (where a counts all but the current microphone).

$$
\begin{array}{l}
for \quad n = N_f : N_f + N_s - 1 \\
\quad for \;\; a = 1 : N-1 \\
\qquad for \;\; b = a+1 : N \\
\qquad\quad g^+(a,b) = p_a^+(n : -1 : n - N_f + 1) * h_{ab}; \\
\qquad end \\
\quad end \\
\quad for \;\; a = N : -1 : 2 \\
\qquad for \;\; b = a-1 : -1 : 1 \\
\qquad\quad g^+(a-1,b) = \\
\qquad\qquad p_b(n) - \left(p_a^-(n : -1 : n - N_f + 1) * h_{ab}\right); \\
\qquad end \\
\quad end \\
\quad for \;\; a = 1 : N \\
\qquad p_a^+(n) = median(g^+(:,a)); \\
\qquad p_a^-(n) = p_a(n) - p_a^+(n); \\
\quad end \\
end
\end{array} \tag{1}
$$

4 Results

The wave separation for the 4th resonant mode of the horn can be seen in Figure 6. This figure shows the signals at the position of the third microphone which is situated at the corpus end of the measurement duct. The pitch of the played note was calculated as being F3 plus 3 cents with an error of ± 15 cents. The top section of the figure shows the third microphone signal with the forward and backward going waves superimposed on top of this. Since the guess matrix entry $g^+(a,b)$ gives the forward going wave at microphone b as predicted at the current time step by projection of the incoming wave from microphone a, we can extract an estimate of the error in the wave separation at the 3rd microphone by taking the standard deviation of the column $N = 3$, which we label $g^+_{*,N}$. The lower section of the figure shows this error term displayed as a function of time.

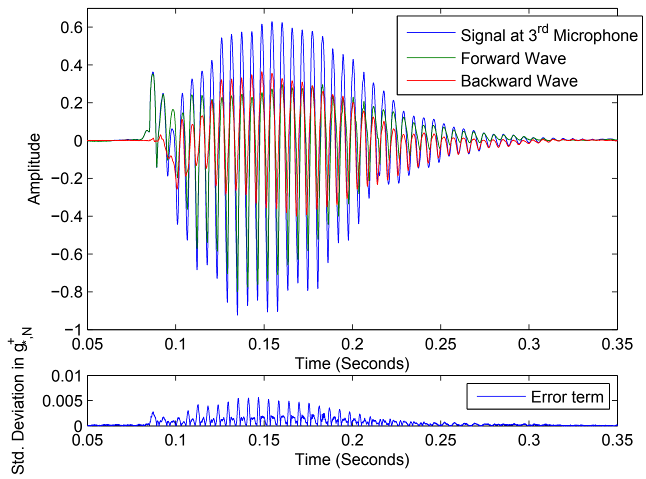

Figure 6: Wave separation for 4th mode on horn

It can be seen that the forward going wave is equal to the measurement for the first two cycles, and then these signals appear to separate and the amplitude of the forward going wave decreases. At the point where the microphone signal and forward going wave signals separate the backward wave shows a significant decrease in amplitude. This negative amplitude is a result of the first cycles of the forward going wave reflecting from the expansion at the bell. It can be seen that by the fourth cycle the forward and backward going wave signals have obtained their steady state phase relationship. The forward and backward going waves are in phase from this point on and it can be seen that the forward going wave has a slightly larger amplitude during the first 0.1 seconds of the note as the player is putting energy into the system. The travel time of the wave from the third microphone to the end of the bell and back is deduced to be approximately 14ms; this can be seen as the time on the figure from the peak of the first cycle, where the forward going wave passes the third microphone, to the trough of the third cycle, where the backward going wave is in phase with the forward going wave for the first time.

Figure 7 shows the results for the sixth mode played on the same instrument and the same set-up; this figure gives a clearer indication of the transient behaviour. The pitch was calculated to be C4 plus 39 cents with an error of ± 15 cents for this mode. Similar trends can be seen for these modes of resonance as for the fourth mode. The forward going wave is initially equal to the microphone signal before the backward going wave shows a negative amplitude reflection from the bell section to set up a steady state phase relationship with the forward going wave. It should be noted that we expect the forward and backward going waves to be in phase at the mouthpiece but not necessarily exactly in phase in the middle of the bore. A well played note results from the backward going wave arriving with the appropriate phase to obtain a steady state relationship. It can be seen that it takes approximately three cycles from the peak of the first cycle to the point where the backward going wave obtains its steady state phase relationship for this higher frequency mode of vibration.

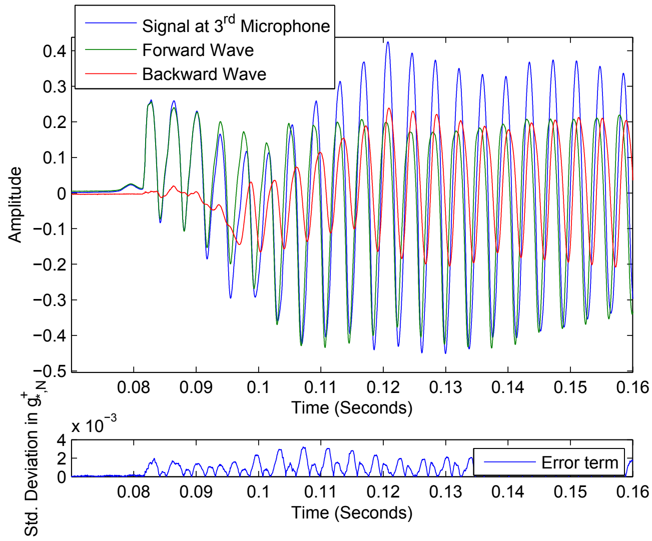

Figure 7: Wave separation for 6th mode on horn

It is particularly interesting to see the transient behaviour for a "split" or "cracked" note as shown in Figure 8. The forward going and backward going waves initially look similar to previous plots. The beating apparent in the microphone signal data is shown to be caused by the forward going wave increasing in pitch. This increase is consistent with the backward going reflections from the bell arriving at the lips out of phase due to the player initiating the note slightly too low in frequency.

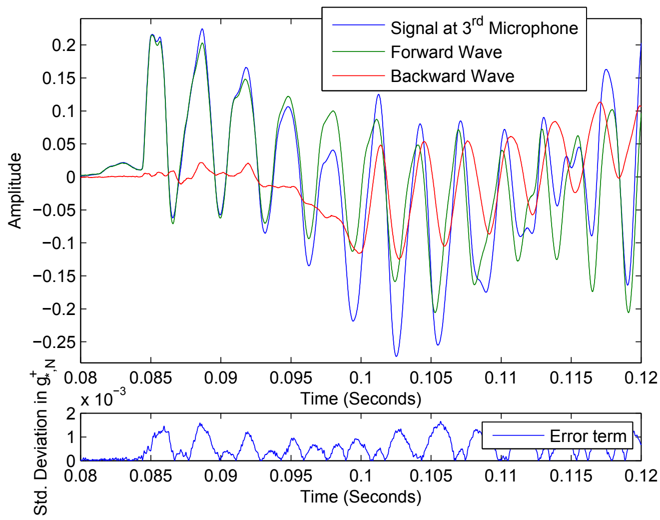

Figure 8: Wave separation for a split note on horn

5 Conclusions and Future Work

This work has used a new technique involving multiple microphones to study starting transients on brass instruments. The results show that this technique works well in separating the forward and backward going waves and showing these relative to the initial measurement at the third microphone. It can be seen clearly that there are a number of cycles where the forward going wave is equal to the measurement. When the backward going wave arrives at the third microphone the forward and backward going waves should have have their final phase relationship necessary for the backward and forward going waves to be in phase at the lips.

With this technique there is the possibility to study a range of interesting aspects of brass instrument playing including lip-slurs, special effects, and super high notes. Future work on this subject could involve combining this technique with measurements at the lips, using optical or microphone based measurements, in order to obtain increased knowledge on the relationship between air pressure and lip motion. Other areas of study may be concerned with analysing transients produced by different players or instruments and deducing the factors which encourage good tone production and instrument design.

References

[1] Luce D., Clark M., "Physical correlates of brass-instrument tones", *J. Acoust. Soc. Am.* 42, 1232-1243 (1967).

[2] Martin D.W., "Lip Vibrations in a Cornet Mouthpiece", *J. Acoust. Soc. Am.* 13, 305-309 (1942).

[3] Copley D.C., Strong W.J., "A Stroboscopic Study of Lip Vibrations in a Trombone", *J. Acoust. Soc. Am.* 99(2), 1219-1226 (1996).

[4] Chick J.P., Bromage S., Campbell D.M., "Transient Behaviour in the motion of the brass player's lips", *Forum Acusticum, Budapest* (2005).

[5] Stevenson S., Chick J., Bromage S., Campbell D.M., "Brass Wind Instruments: Transient Behaviour in the Lip-Reed", *Proceedings of the International Symposium of Musical Acoustics, Barcelona* (2007).

[6] Logie S., Stevenson S., Apostoli A., Chick J., Campbell D.M., "Transient behaviour in the motion of the brass player's lips during a lip-slur." *Proceedings of the NAG-DAGA international conference on Acoustics, Rotterdam* (2009).

[7] Kemp J.A., van Walstijn M., Campbell D.M., Chick J.P., Smith R.A., "Time Domain Wave Separation Using Multiple Microphones", *J. Acoust. Soc. Am. (Under Review)* (2010).

[8] van Walstijn M., Campbell M., Kemp J., Sharp D., "Wideband Measurement of the Acoustic Impedance of Tubular Objects" *Acta Acustica united with Acustica* 91, 590-604 (2005).

[9] van Walstijn M., de Sanctis, G., "Towards physics-based re-synthesis of woodwind tones" *19th International Congress on Acoustics, Madrid, Spain* (2007).

[10] de Sanctis, G., van Walstijn M., "A frequency domain adaptive algorithm for wave separation" *12th Int. Conference on Digital Audio Effects (DAFx-09), Como, Italy* (2009).

[11] Kemp J.A., Campbell D.M., Chick J.P., Hendrie D., "Acoustic pulse reflectometry for the measurement of horn crooks", *Proceedings of Acoustics'08, Paris* (2008).