IMPROVEMENTS TO BORE PROFILE MEASUREMENT IN ACOUSTIC PULSE REFLECTOMETRY

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1 INTRODUCTION

Acoustic Pulse Reflectometry (APR) is a non-invasive technique that can be used for the measurement of the input impulse response, input impedance and the deduction of the internal profile of a tubular object. In conventional APR a loudspeaker produces an audible click which travels down the air contained in a cylindrical source tube and into the object under test. Reflections occur within the object under test when the bore contracts or expands and the resulting backward going waves travel back down the source tube to be measured by a microphone mounted in the source tube wall. Calibration is performed by running a measurement with the source tube closed by a cap, and the bore profile (as a plot of internal radius versus axial distance) of the object under test can then be deduced using the layer peeling bore reconstruction algorithm [1].

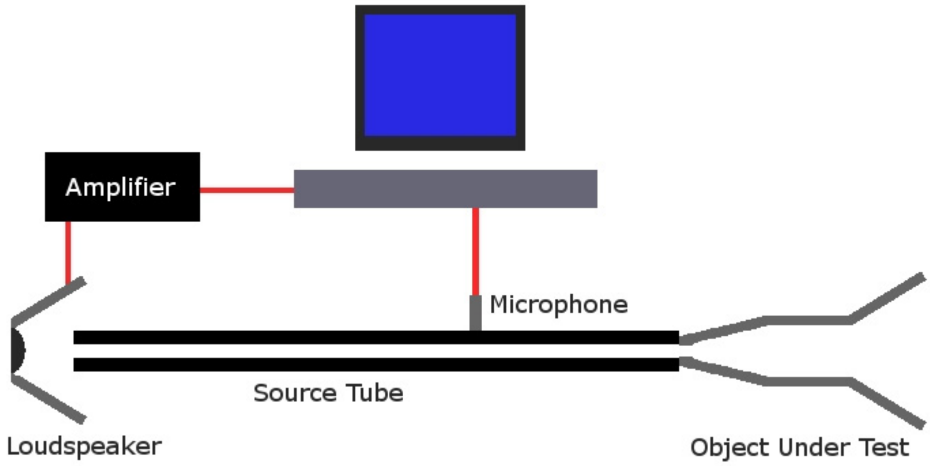

Figure 1. Schematic diagram of the standard apparatus used in acoustic pulse reflectometry (APR).

This paper describes recent developments in the technique at University of Edinburgh. These relate to the method of excitation, the importance of DC and low frequency information in determining slow trends in the resulting reconstruction, and the method of calibration of the apparatus.

2 MAXIMUM LENGTH SEQUENCES

2.1 Theory

While conventional APR uses an impulsive excitation, this method is not efficient at getting energy into the system. For this reason the experiment must be repeated over and over again and results averaged to give a good signal to noise ratio. A more time efficient technique is to use maximum length sequences (or MLS).

An MLS signal consists of an apparently random sequence of 0's and 1's that has a flat frequency spectrum for all frequencies up to the Nyquist frequency with the exception of the DC value. It is also computationally efficient to generate such a sequence and unlike white noise an MLS signal is deterministic and therefore repeatable. In pulse reflectometry the sequence is used as the input to the loudspeaker and the signal is measured at the microphone. Since a MLS is a periodic signal the loudspeaker should produce two repetitions of the MLS with the microphone signal recorded during the second play. The input impulse response of a system can be extracted by cross-correlation of the original MLS with the microphone signal. This method of excitation has been employed frequently in measuring the input impulse response of rooms for reverberation measurement [2] and more recently for APR.

The easiest method of performing the cross-correlation to program is to use frequency domain multiplication.

$$H(\omega) = MLS^{*}(\omega) \times P(\omega)$$

where $H(\omega)$ is the Discrete Fourier Transform of the system impulse response, $P(\omega)$ is the Discrete Fourier Transform of the measured microphone signal and $MLS^{*}(\omega)$ is the complex conjugate of the Discrete Fourier Transform of the original MLS sequence that was fed to the loudspeaker. A computationally much faster technique is to use the Fast Hadamard Transform for cross-correlation but with modern computer power the Discrete Fourier Transform approach is perfectly adequate.

2.1.1 Compensating for distortion

Harmonic distortion has a bad effect on the results of performing a measurement using MLS signals through causing spurious peaks in the system impulse response. These peaks are often easy to spot in the time domain system impulse response as they show up even beyond the point in time at which the response could be expected to have died away to zero. They are also repeatable in that the peaks show up at the same places in the system impulse response each time if the same MLS signal is used to excite the same system but they cannot be easily removed by analytical deduction. The main source of harmonic distortion in the current experiments was due to the loudspeaker being driven at high volumes. To combat this effect the volume can be reduced until the spurious peaks were much less significant. The best way of removing the effect of distortion altogether is to use more than two orders of MLS or more than two different MLS signals of the same order (found using different feedback shift registers based on primitive polynomial coefficients) and looking for unexpected disagreements in the results. With an order of 18 there are a very large number of suitable shift registers available. These (deduced from [3]) include the following five:

Order 18 shift registers for generating MLS
a(i+18) = a(i+7) + a(i)
a(i+18) = a(i+11) + a(i)
a(i+18) = a(i+10) + a(i+7) +a(i+5) + a(i)
a(i+18) = a(i+10) + a(i+8) +a(i+5) + a(i)
a(i+18) = a(i+15) + a(i+5) +a(i+2) + a(i)

The involved recursion orders can be listed more conveniently as

Order 18 recursion codes for generating MLS				
[7,18]	[11,18]	[5,7,10,18]	[5,8,10,18]	[2,5,15,18]

Note that the order (in this case 18) is present on the left hand side of the recursion expression and the order 0 is assumed on the right hand side in every case. To construct an order 18 MLS the first 18 samples in the vector *a* are set to a value of 1 and then one of the feedback registers above is applied with the result taken modulo 2. The registers are designed so that the vector repeats itself after 2^{18} - 1 samples (hence the need to play the MLS twice and record the microphone signal during the second playing). An order 18 MLS sequence therefore consists of 2^{18} - 1 samples meaning that the sequence lasts for just under 6 seconds when played back at a 44.1 kHz sample rate. This is also the length of the pressure signal used in the measurement and the length of the resulting system impulse response. An order 18 MLS is therefore long enough for most reverberation measurements and lower orders can be used for measurements in APR where the length of the input impulse response of a wind musical instrument is usually less than 0.1 seconds. The signal is converted from a series of 0s and 1s into a sequence of 1s and -1s before being fed to the loudspeaker in order to prevent DC offset problems.

The recursion codes for lower orders (as deduced from [3]) are tabulated below.

Order 17 recursion codes for generating MLS				
[14,17]	[1,2,3,17]	[3,4,8,17]	[3,5,7,9,11,13,15,17]	[1,2,3,6,12,17]

Order 16 recursion codes for generating MLS				
[4,13,15,16]	[1,3,12,16]	[1,3,6,7,11,12,13,16]	[1,6,7,9,10,12,13,16]	1,2,6,9,10,11,15,16]

Order 15 recursion codes for generating MLS				
[14,15]	[1,5,10,15];	[1,3,12,15]	[1,2,4,5,10,15]	[1,2,6,7,11,15];

Order 14 recursion codes for generating MLS				
[4,8,13,14]	[1,6,10,14]	[1,3,4,6,7,9,10,14]	[4,5,6,7,8,9,12,14]	[1,6,8,14]

Order 13 recursion codes for generating MLS				
[1,3,4,13]	[4,5,7,9,10,13]	[1,4,7,8,11,13]	[1,2,3,6,8,9,10,13]	[9,10,12,13]

Note that an order 13 MLS sequence has 2^{13} - 1 samples and so lasts for 0.19 seconds to 2 d.p. at 44.1kHz playback. This is the most suitable order for measuring most brass musical wind instruments as the signal takes longer to play (at 44.1 kHz) than typical brass instrument input impulse responses. For clarity the suitable shift registers (deduced from [4]) are tabulated below:

Order 13 shift registers for generating MLS
a(i+13) = a(i+4) + a(i+3) + a(i+1) + a(i)
a(i+13) = a(i+10) + a(i+9) + a(i+7) + a(i+5) + a(i+4) + a(i)
a(i+13) = a(i+11) + a(i+8) + a(i+7) + a(i+4) + a(i+1) + a(i)
a(i+13) = a(i+10) + a(i+9) + a(i+8) + a(i+6) + a(i+3) + a(i+2) + a(i+1) + a(i)
a(i+13) = a(i+12) + a(i+10) +a(i+9) + a(i)

In order to subtract the spurious peaks in the system impulse response the following algorithm was used in the matlab programming language:

```
for n=1:N
      pbest(n)=(sum(p(1:5,n))-max(p(1:5,n))-min(p(1:5,n)))/3;
end
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were p(q,n) is the system impulse response at the nth time step measured with the qth MLS. The output (called pbest) then consists of the average of the three best values at that time step (i.e. canceling the maximum and minimum values from the average at that time step as they may correspond to spurious peaks). As the spurious peaks do not coincide for measurements using MLS signals derived from different recursion relations the effect of the distortion is successfully removed from the measurement.

2.1.2 Pre-Filtering

The frequency spectrum of an MLS is flat at all frequencies up to Nyquist (half the sample rate). In this way all frequency ranges can be measured at once. Occasionally it might be desirable to measure only in a specific bandwidth or to measure in several bandwidths, one after the other and combine the results into a large bandwidth measurement. This can be achieved by filtering the MLS with a time domain feedforward filter. This sort of filter is performed by convolution. Normally the system impulse response is the desired impulse response of the source tube and object under test coloured by the various responses of the components within the system. In the time domain the impulse responses must be convolved with one another and this can be expressed in the frequency domain as multiplication:

$$H(\omega) = OBJ(\omega) \times M(\omega) \times I(\omega) \times L(\omega) \times A(\omega) \times O(\omega)$$

where $H(\omega)$ is measured (frequency domain) signal, $OBJ(\omega)$ is the desired response of the source tube and object under test, $M(\omega)$ is the response of the microphone, $I(\omega)$ is the response of the soundcard input, $L(\omega)$ is the response of the loudspeaker, $A(\omega)$ is the response of the amplifier and $O(\omega)$ is the response of the soundcard output. With good quality components most of the colouration on the measurement is due to loss of energy within the source tube. If we filter the MLS by a further feedforward filter then the effect will not cause the auto-correlation property of the MLS to fail, it will simply colour the experimental result by the impulse response of the filter. If the feedforward filter has a frequency response of one over a particular frequency range then the impulse response will be accurately measured for that frequency range (and not at other frequencies outside the bandwidth of the filter). In the current application it was found that high pass filtering the output sharpens up the impulse in the measurement and enables better windowing of the input and output signals coming from the object under test.

3 LOW FREQUENCY ACCURACY

As noted by Li and Sharp [5], the low frequency information is particularly important in determining slow drifts in the reconstruction of the internal profile. Filtering the MLS in order to increase the low frequency content of the input signal may seem useful but this means that the forward and backward going pulses measured in the system impulse response are made wider and this prevents them being windowed. In order to measure low frequencies accurately the two microphone technique described by van Waalstijn et al [6] was used using a theoretical calculation of the losses in the source tube. Sine wave measurements of the impedance were made at 11 Hz and 22 Hz. These values were then used to deduce the impulse response and these values were used to replace the values produced using MLS APR at these frequencies. The DC component was set to the theoretical value -1 meaning that no averaging was needed to subtract this component.

4 CALIBRATION

In order to get a true impulse response of the object placed on the end of the source tube a measurement is made of the system impulse response with the object under test on the end and then a measurement is made of the system impulse response with the end terminated in a rigid cap. The length of source tube between the microphone and object under test are chosen to be long enough to give good separation of the 'forward going' pulse traveling from the loudspeaker and the 'backward going' measurement resulting from its reflection from the object under test. Also the length of the portion of the source tube between the microphone and loudspeaker can be chosen to separate the backward going wave from the object under test from the result of this wave reflecting from the loudspeaker. In the experiments performed in this paper the source cancellation method was used as set out in [7]. In this method the 'cap measurement' is windowed to include both the backward going wave from the object under test and its reflection from the loudspeaker. These two are then deconvolved by frequency domain division to get a transfer function for waves reflecting from the loudspeaker. This transfer function is then convolved with the backward going wave from the object under test (with a fade out applied for windowing) to predict the forward going wave resulting from its reflection in the loudspeaker. The predicted reflections are then subtracted to give the backward going reflection with the source reflections removed.

Once the measurements have been windowed to remove the forward going impulse, which is not of interest, deconvolution is applied by frequency domain division.

$$IIR(\omega) = \frac{OBJ(\omega)}{CAP(\omega)}$$

This process removes the colouration due to the microphone, loudspeaker, soundcard and source tube propagation and any pre-filtering to give the impulse response of the object that has been placed on the end of the source tube. The only limitation is that the signal to noise ratio is very poor at frequencies outside the bandwidth of the above elements.

No signal is measured above around 15 kHz for most pulse reflectometry measurements (depending on the length of the reflectometer source tube) due to the heavy loss of energy due to propagation for high frequencies. For similar reasons and due to the laws of diffraction the reflection coefficients of most musical instruments to these high frequencies are minimal. In practice the random noise at high frequencies means that sometimes the cap measurement has a much smaller signal level at a particular frequency than the object reflections measurement and this leads to the deconvolution diverging at these frequencies. These singularities in the measurement have often been compensated for by using a small constraining factor, q, added to the denominator of the frequency domain deconvolution [5,8,9]:

$$IIR(\omega) = \frac{OBJ(\omega)}{CAP(\omega) + q}$$

This has the desired effect of forcing the impulse response to zero outside the bandwidth of the measured signal. The disadvantage of this technique is that it has a small effect on frequencies within the measurement bandwidth. The current experiments use a new version of this processing which can be described as a constraining vector:

$$IIR(\omega) = \frac{OBJ(\omega)}{CAP(\omega)(1 + q(\omega))}$$

with $q(\omega)$ being a vector consisting of zeros for low frequencies and then increasing exponentially at high frequencies to force the impulse response to zero outside the bandwidth. The variables used for setting up the constraining vector must be fine tuned depending largely on the losses determined by the radius and length of the source tube and typically to a lesser degree on the microphone and loudspeaker bandwidth. Another technique for dealing with these singularities (not employed here), developed by Forbes et al [9], involves truncated singular value decomposition (TSVD).

RESULTS

The experiments were performed using MLS (and sine wave) signals fed from a soundcard through a Denon hi-fi amplifier into an Eminence compression driver loudspeaker which was coupled to a copper source tube of internal radius 10mm. Miniature Sennheiser microphone capsules (model KE4-211) were used to pick up the sound which was recorded into the soundcard. In figure 2 we see an example of frequency domain division and the problem of high frequency noise above the bandwidth which for this experiment was around 15 kHz.

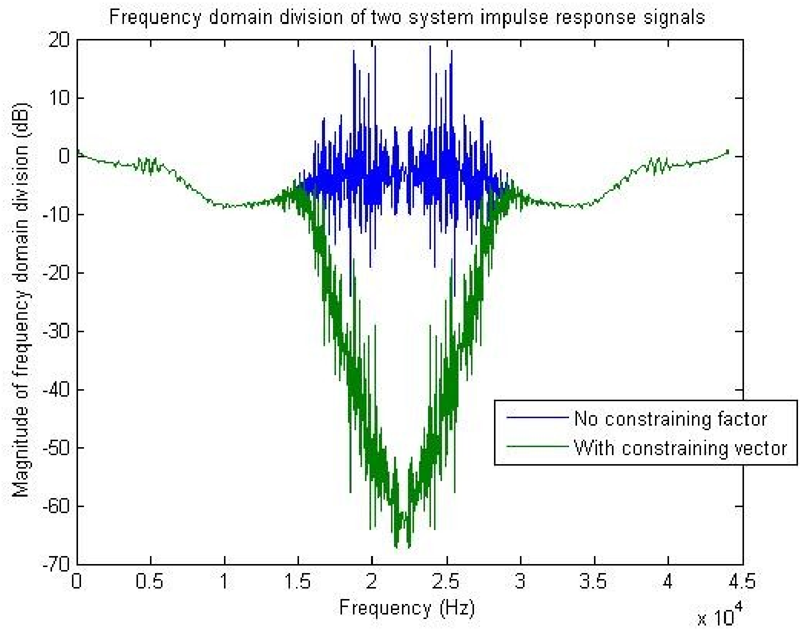

Figure 2. Frequency domain division of the system impulse responses measured by two microphones in a semi-infinite tube. The calculations with no constraining factor and with the use of a constraining vector are shown.

In figure 3 we see a typical system impulse response for a pulse reflectometry system. In this case the negative pulse can be seen as it passes the loudspeaker on the way towards an object under test. By zooming in to a region around 30 milliseconds later as shown in figure 4 we can see details of the reflections from an object under test placed on the end of the source tube. Five different order 13 MLS excitations were used and each has spurious peaks at different times. As mentioned above these spurious peaks are due to the non-linear behaviour of the loudspeaker (i.e. harmonic distortion). As can be seen from the black signal the “average of the best three of five” algorithm has successfully prevented the spurious peaks from effecting the averaging.

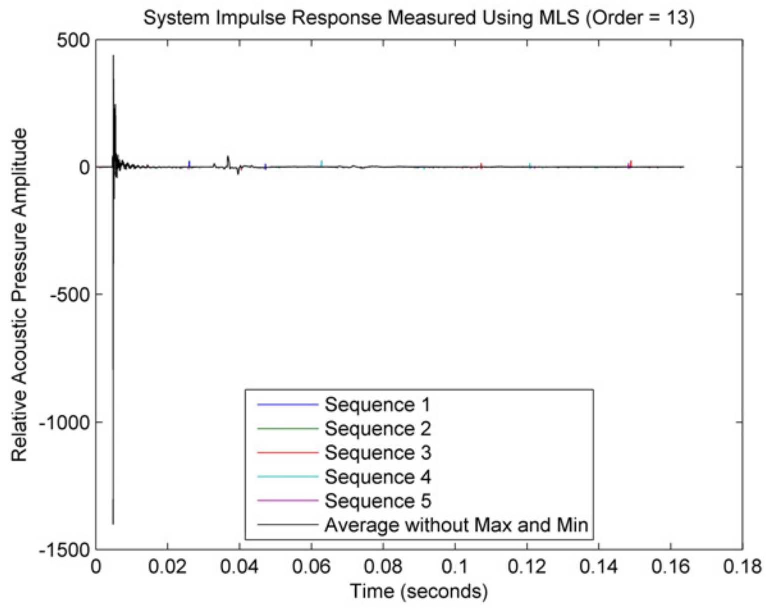

Figure 3. The system impulse response of an acoustic pulse reflectometry system using five different MLS sequences all of order 13. The average without the highest and lowest value at each time step is also shown.

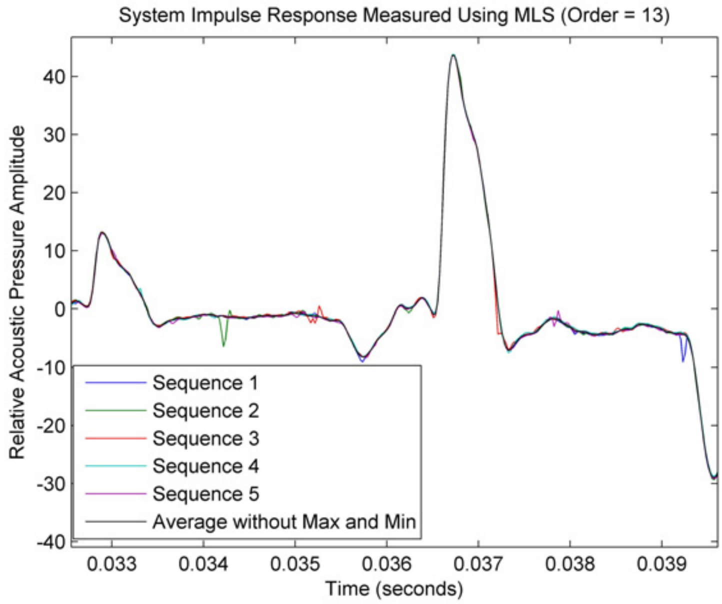

Figure 4. Detail of measurement shown in figure 3. The average without the highest and lowest value at each time step is also shown. The small spurious coloured peaks are due to harmonic distortion and are correctly ignored.

As an example of the application of the improvements to APR, in figure 5 we see the measurement of three 19th century crooks from a terminally crooked orchestral horn made by Gautrot, Paris. The measurements show accurate exit radii for the E flat and D basso crooks while the C basso crook shows an over-predicted radius. This indicates the presence of a previously unnoticed leak 700 mm from the mouthpiece end. Underwater blowing of the crook confirmed the presence of a leak at that location. Pulse reflectometry thus shows how these crooks are composed of sections of roughly conical and cylindrical shape and can locate any leaks. In general, crooks of smoother bore have been found to be preferable for playing [10].

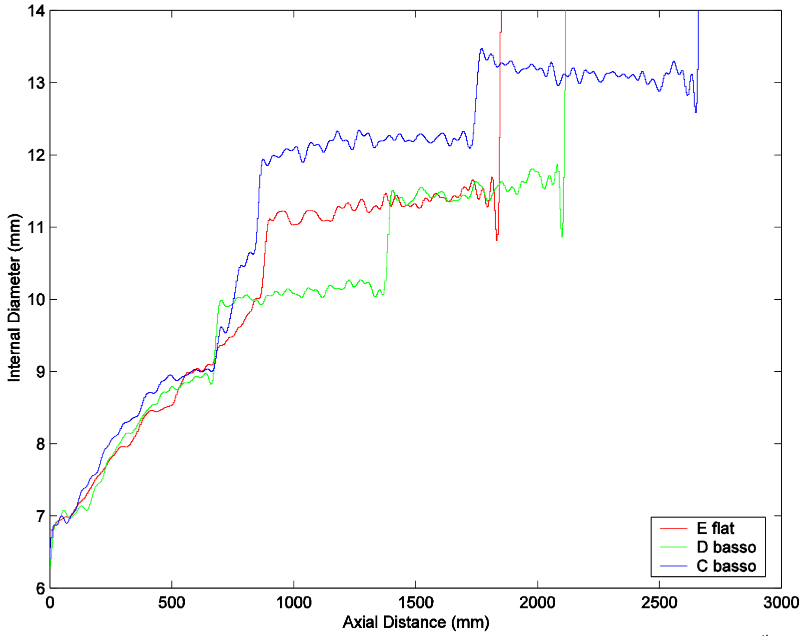

Figure 5. Bore reconstructions of three different Gautrot crooks for use with a 19th century orchestral horn. The construction using sections of roughly conical and cylindrical shape is evident. The C basso crook has a small leak at around 700 mm hence the over-predicted bore profile from this point onwards.

5 CONCLUSIONS

The technique of acoustic pulse reflectometry has been extended to allow the measurement of longer objects such as terminal orchestral horns by the use of sine wave excitation to significantly improve the low frequency accuracy. MLS signals have been found to improve the measurement speed and the problem of spurious peaks introduced by loudspeaker distortion has been removed using multiple MLS signals. This paper also provides a useful reference for producing MLS signals and analyzing experimental results.

6 REFERENCES

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