THE NATURAL TRUMPET AND ITS VIRTUAL SOUND

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Abstract

Many museums and private collections have ancient musical instruments, but very often they are damaged and it isn't possible to listen to how they sound.
Furthermore, these instruments are not easily classified, since it should be necessary to use destructive technique to make measurements on their body.
In this paper the sound characterisation of trumpets is analysed. The linear behaviour of the trumpet was taken into account considering the system included between the bore, just after the mouthpiece, and the near acoustic field, one meter outside the bell. Limiting the maximum amplitude of sound pressure within the instrument, this system behaves linearly. Such an amplitude limitation means that the nuance "fortissimo" cannot be reproduced perfectly. Using very little microphones positioned in 2 different points, few performances in a semi-anechoic chamber have been recorded. In a following step, a "dry" signal inside the trumpet has been obtained, after checking different techniques, getting the inverse impulse response (F.I.R.) and convoluting the FIR with the recording of performance. The "dry" signal has been convoluted with the IR measured outside, and finally compared with the recordings. The sound examples coming from the "virtual performance" are presented; the different techniques are shown and discussed.

INTRODUCTION

Wind instruments, like flutes or trumpets, are quite difficult to model properly. In the case of trumpet, especially natural (baroque) trumpet, it is possible to consider its non linear characteristics due to the interaction of lips and mouthpiece, and to the viscosity and turbulence of air flow near the lips. It is feasible to suppose the bore and the flaring bell as the main source of sound radiation, since they work like a horn loudspeaker.

It is assumed that the recovered dry excitation signal is essentially independent of the instrument's geometry. Just the non-linear interaction between lips and mouthpiece (as they work by self-oscillation) is still part of it, but this interaction is influenced by different geometries of different actual instruments only as a second-order effect, which is assumed to be negligible at little amplitudes.

IMPULSE RESPONSE MEASUREMENT TECHNIQUES

Two different techniques were available for the measurements: the MLS technique and the "stretched pulse" technique.

In the first case, a PC equipped with a sound card was generating the MLS (Maximum Length Sequence) signal, which was driving a small loudspeaker that has been fixed on the mouthpiece. The signal coming from the preamplifier of the microphone was sampled by the A/D board, and cross-correlated with the original MLS signal to obtain the impulse response directly in the time domain, thanks to the Alrutz fast deconvolution algorithm.

This technique was already used from one of the authors on modern trumpet [1]. In the second case, a sine sweep signal ("stretched pulse") was used to calculate impulse responses; this novel technique for calculations of IRs yields better results. As demonstrated by the author of the system [2] the harmonic distortion typically found with MLS measurements can be avoided. Besides, a better signal/noise ratio can be achieved. All these techniques are available in the Aurora system [3].

THE VIRTUAL SOUND: INVERSION AND RECONVOLUTION

The inversion: formulation of the problem

The question of creating inverse filters causal, stable and of finite length has involved many researchers in many Countries in last decade. The question becomes more difficult when the impulse responses to be inverted are quite long and mixed phase. Mainly three different techniques have been developed: the *minimum/maximum phase decomposition* with separate inversion; the *least-squares approximation*, and a novel technique, the *frequency-domain regularisation technique*, which is still under development. This last technique, in collaboration with the researchers developing the software implementation, has been utilised.

The technique: frequency-domain regularisation

The frequency-domain regularisation technique has been first developed at University of Southampton, and recently upgraded This approach was first developed at the ISVR [4], and further refined with the co-operation of one of the authors [5], introducing a frequency-dependent regularisation parameter. Following this technique, the original response h(t) is first FFT transformed:

$$C(\omega) = FFT[h(t)]$$

(1)

Then the complex spectrum C is inverted:

$$C_{inv}(\omega) = \frac{Conj[C(\omega)]}{Conj[C(\omega)] \cdot C(\omega) + \varepsilon(\omega)}$$

(2)

And the result is back-transformed to time domain

$$h_{inv}(t) = IFFT[C_{inv}(\omega)]$$

(3)

This technique, after a proper developing of the computational parameters, has revealed to be quite good to be applied in systems, which are characterised by short and regular impulse response, like natural trumpets.

The reconvolution

After deconvolution of recorded music tracks with the inverse impulse response, a convolution with many different impulse responses is required. In this case frequency domain processing technique (select-save algorithm), developed in Aurora System, has been utilised.

EXPERIMENTS

Performances and impulse response measurements

The impulse responses of two natural trumpets have been measured, namely a Conforzi-Callegari (a tonal copy of the "Wilhelm Magnus Ehe I" model, XVII century), and a hunt-trumpet Meinl&Lauber. During the performances, copies of J.E. Altemburg' mouthpieces (XVIII century) have been utilised.

The measurements of impulse responses have been conducted both with MLS technique (using different orders of sequences) and stretched pulse technique. The second one gave better results, and was preferred.

The impulse responses were measured in many positions inside the bore, at a distance of 5 mm each other. Measurements of impulse responses were conducted also in front of the trumpet, at 1 meter from the bell. In this way, the near-field radiation point, located at a distance of 1 m on the axis of the instruments, was measured.

Fig. 1 - Measurements of IRs on "Conforzi-Callegaro" (left) and Mein&Lauber (right) trumpet

With all these measured impulse responses in the bore, will be possible in the next time to calculate the sound intensity inside the cylindrical tube.
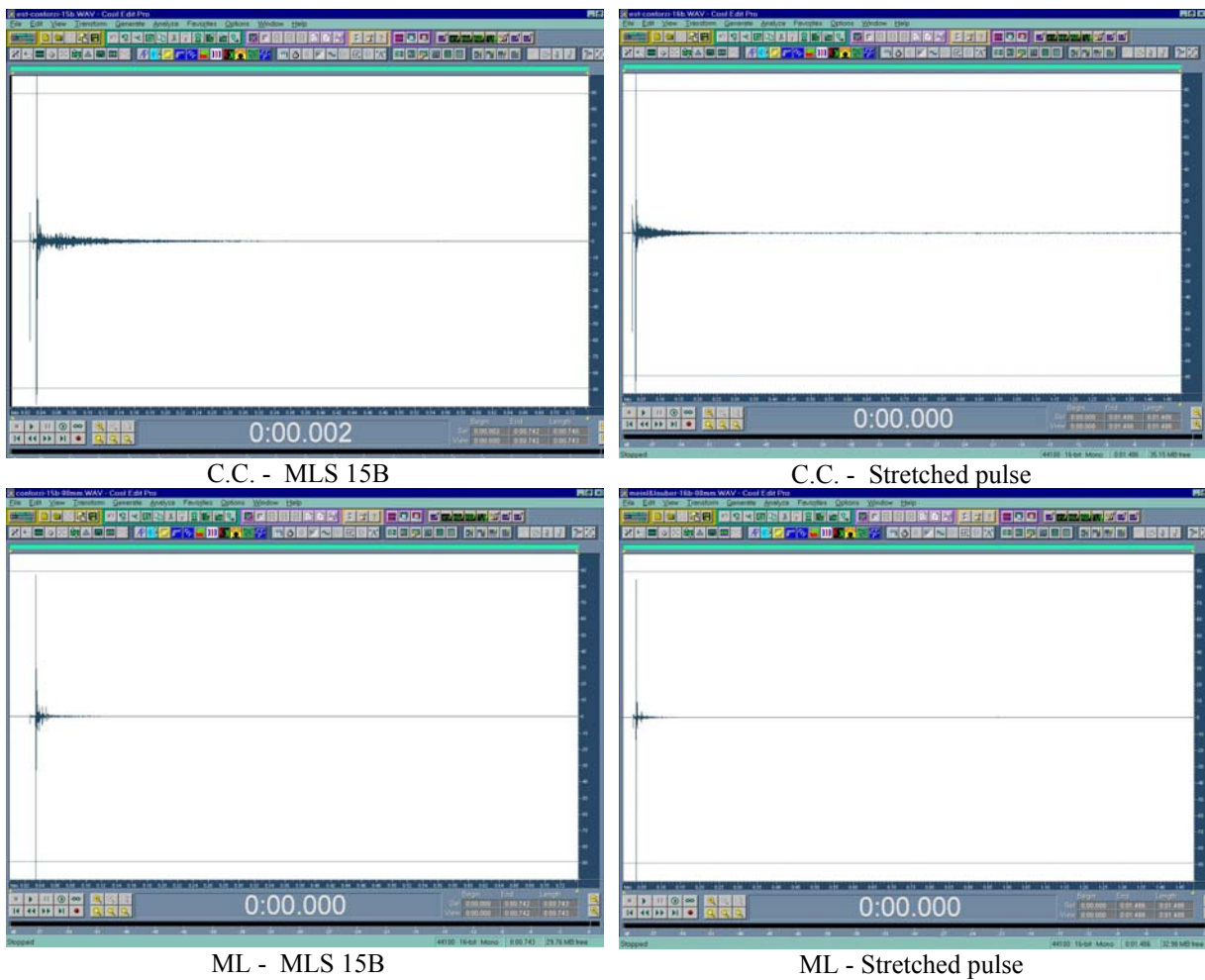

C.C. - MLS 15B

C.C. - Stretched pulse

ML - MLS 15B

ML - Stretched pulse

Fig. 2 - IRs measurements with different techniques in different trumpets at the end of the bell
Recording of music samples

The recordings have been performed in the laboratory of CIARM, at University of Bologna. Thanks to M° Conforzi, different pieces have been recorded in a semi-anechoic environment in a DAT and digitally stored as .WAV files on the PC hard disk using a professional digital sound board (Layla by Echo). Two free-field microphones have been utilised. The first one was placed in the final part of the bell of the trumpet, while the other one meter in front of the instrument, at exactly the same point where the "reference" impulse response has been measured.
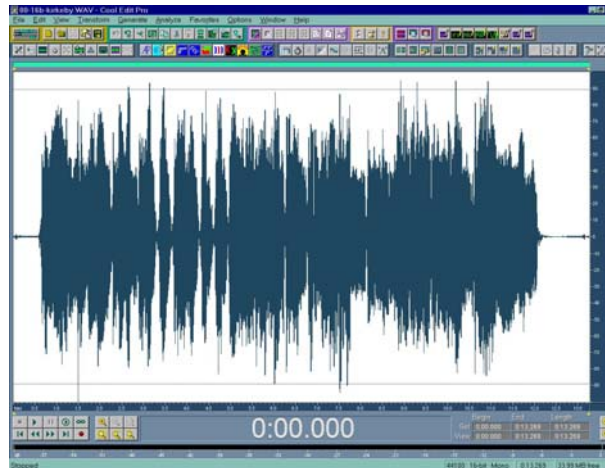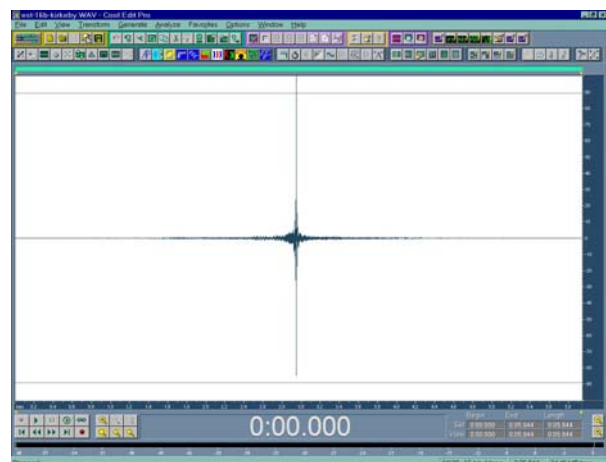

Fig. 3 - Recorded music at 1000 mm - Conforzi trumpet

Creation of the inverse filters and deconvolution of "dry" excitation signals

To get the anechoic signal it could be theoretically possible to *directly* measure the sound-pressure signal in the mouthpiece. To make measurements directly in the mouthpiece it is necessary to insert miniaturised microphones that inevitably modify the acoustical field in such a peculiar position in the instrument.
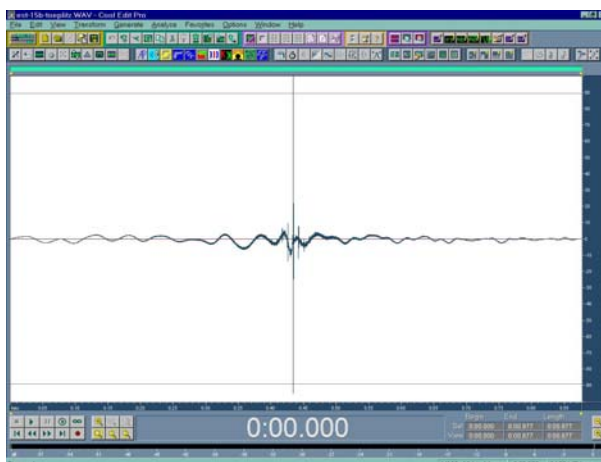
inverse filter: Toeplitz' technique

inverse filter: Kirkeby' technique

Fig. 4 - Inverse IR at 00 mm - Conforzi trumpet

These reasons provoke to choose an *indirect* way to get the anechoic signal in the mouthpiece, that is to convolve the music recorded near to the end of the bell with the inverse filter obtained by the impulse response measured exactly in the same position.

Following the reciprocity theory, the signal could be recorded at 1m from the bell, and the consequent IR that should be inverted at the same position (1m). This second manner revealed to be less satisfactory, probably due to the fact that some environmental effect is included in the impulse response to be inverted, and that the microphone positioning error is greater.

After the computational of the inverse filter, the anechoic signal was obtained. Different experiments have been conducted on different performances, recorded in different positions.

In order to recreate the sound characterisation of other wind instruments, the dry excitation signal have to be convolved with their impulse responses, measured in the same environment conditions. When the impulse responses used for this purpose are covering all the transmission system between a certain instrument's mouthpiece and the listeners ear, oh this case all acoustical influences of that instruments as well as of the room, where the impulse response has been measured, are included. In this way the sound characteristics of the instrument is fully reconstructed. Furthermore, in order to fully represent the sound quality in a theatre for headphone listening caused by a virtual instrument, it become necessary to convolve the anechoic signal also with the impulse response of the theatre. Otherwise it is simply necessary, to reproduce the sound through a loudspeaker in real room.
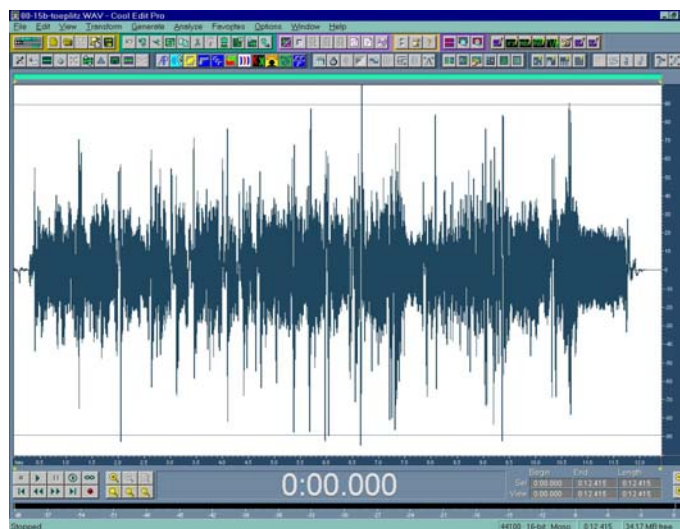

Fig. 5 - Dry signal after convolution with inverse filter (Kirkeby)

After inversion, a band-pass, second-order Chebychev filter was applied with cut-off frequencies of 80 and 12000 Hz, for reducing the excessive gain at extreme frequencies.

The dry signal, obtained by convolution with the inverse filter calculate with Kirkeby technique from the impulse response measured at 00 mm out of the Conforzi-Callegari

trumpet, has been convolved with the impulse response measured in the same trumpet at 1000 mm in front of the bell, and compared with the recording made in the same position.

Fig. 6 - Recorded signal and virtual reconstruction obtained by convolution

The comparison between recording music and reconstructed music has revealed quite good degree of similarity. The only appreciable difference was a slightly more "reverberant" reconstructed signal, no timbric alteration can be perceived.

The recorded and the reconstructed music were almost indistinguishable. When the sound samples were reproduced through loudspeakers in a normally reverberant room, the differences between the true recording and the virtual one obtained by convolution were very little. The acoustic characteristic of the listening environment is quite more important and covers all residual differences between the real and virtual sound.
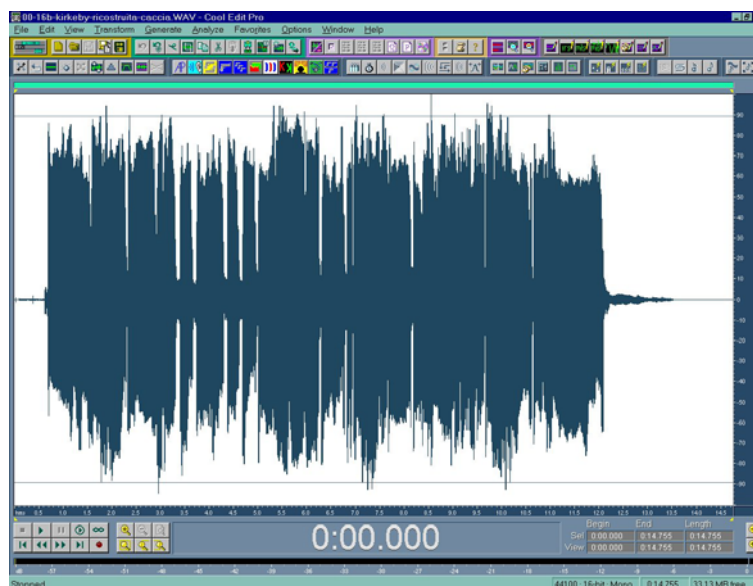

Fig. 7 - Virtual reconstruction obtained by convolution on Mein&Lauber trumpet

Following the procedure analysed, it was therefore possible to obtain the virtual sound of the hunt-trumpet (Mein&Lauber), as shown in the picture.

CONCLUSIONS

The creation of virtual string instruments by convolution methods has been already developed, regarding to violin [6]. The "stretched pulse" method revealed to be more satisfactory than MLS method. The frequency-domain regularisation procedure, properly utilised, revealed to be quite powerful and preferable with respect to the least-squares method. The procedure analysed permits to investigate many different conditions of the instruments, like influence in material and obsolescence [7]. Furthermore, being possible to predict the impulse response of not existing instruments, by using FEM-BEM models, it will allow to proper design new instruments or recreate ancient musical instruments.

REFERENCES

[1] Tronchin, L., Cocchi, A., (1999) A virtual reconstruction of the trumpet, *Proc. of* Forum Acusticum, Berlin, (Germany)

[2] Farina A. (2000) Simultaneous measurement of impulse response and distortion with a swept-sine technique *Pre-prints of the 108th AES Convention*, Paris 18-22 February 2000.

[3] Farina A., Righini F. (1997) Software implementation of an MLS analyzer, with tools for convolution, auralization and inverse filtering. *Pre-prints of the 103rd AES Convention*, New York, 26-29 September 1997.

[4] Kirkeby O. and Nelson P. A. (1998) Digital Filter Design for Virtual Source Imaging Systems, *Pre-prints of the 104th AES Convention*, Amsterdam, 15 - 20 May.

[5] Kirkeby O., Rubak P., Farina A. (1999) Analysis of ill-conditioning of multi-channel deconvolution problems *1999 IEEE Workshop on Applications of Signal Processing to Audio and Acoustics* - Mohonk Mountain House New Paltz, New York October 17-20, 1999

[6] Farina A., Langhoff A., Tronchin L. (1998) Acoustic characterisation of "virtual" musical instruments: using MLS technique on ancient violins, *Journal of New Music Research (ISSN 0929-8215)*, Swets & Zeitlinger Publ., vol. 27, N. 4, pp. 359-379

[7] Internet References: many sound examples are available at the web site of CIARM Bologna office , at the address: HTTP://www.ciarm.ing.unibo.it/researches/trumpet