On the“Virtual” Reconstruction of Sound Quality of Trumpets

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Summary

In this paper the realisation of “virtual” wind instruments is analysed, in which the bores are treated as linear, invariant systems, characterised by their impulse responses.

The “dry” excitation signal has been obtained from the output of the system, sampled through a microphone placed at the flaring bell of a trumpet, by convolution with a proper inverse filter, obtained from the impulse response measured between the mouthpiece and the recording point.

This “dry” excitation signal is the convolved with the impulse response of different wind instruments, thus reproducing the sound of “virtual” instruments. By comparing the results of very similar and much more different instruments, it was possible to conclude that this method makes it possible to finely distinguish between subtle timbric difference among different trumpets. This is due to the fact that the excitation signal remains always exactly the same, whilst when an human performer plays on different instruments, he always compensates for the different response of each instrument, making the differences less audible, and often the execution is strongly modified by this unavoidable feedback.

The aim of this work is multiple: the “virtual” instruments can be used in subjective listening tests for the comparison of the “sound quality” of different instruments, for the evaluation of (real or simulated) restoration of ancient instruments, and for preliminary listening tests with newly designed ones, before they are actually built.

For validating the repeatability of the technique, a blind subjective listening test has been performed. Three different trumpets and a silver-flute have been analysed, and compared with each other. The statistical analysis of the listening tests confirmed the excellent similarity between the direct acoustic recording and the result of the convolution technique.

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1. Introduction

Musical instruments, especially wind instruments, cannot be considered globally as linear transducers, but in the trumpet their non linear characteristics are especially due to the interaction of lips and mouthpiece, and to the viscosity and turbulence of air flow near the lips. From a mechanical point of view, the bore of the trumpet and the flaring bell could be considered the most important source of timbric radiation of the instrument, because they operate like a horn loudspeaker, with all characteristics and limitations of such an acoustic transducer.

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. If we explicitly limit 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.

From a theoretical point of view, it is therefore possible to obtain the acoustical characteristics of the trumpet by measuring the impulse response from the beginning of the bore at the mouthpiece to the near acoustic field. The sound emitted by the trumpet can then be recreated by convolving a suitable input (dry) signal with the measured impulse response, as first suggested by one of the authors [Tronchin 1999].

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In order to measure the impulse response, it was necessary to put a very small sound source inside the bell, close to the mouthpiece.

The “dry” excitation signal could be *directly* recorded by placing a small pressure transducer in that same position, during the normal use of the trumpet. Such a technique turned out to be unfeasible and was discarded.

Instead, the dry signal was *indirectly* calculated from a recording made with a microphone placed in the flaring bell of the trumpet. The input signal was recovered by convolution of the sound-pressure recording with the inverse filter of the transmission system, between the mouthpiece and the microphone position. This way, all non-linear characteristics of the sound generation are still part of the calculated excitation signal, while all linear characteristics associated with the transmission function of the instrument are removed.

It must be noted that the excitation signal needs to be recorded on a complete trumpet, because the acoustical load, which is seen downstream, influences the not-linear interaction between lips and the airflow. In principle, such a feedback changes on a different instrument, but it is assumed that such differences are of minor importance in comparison with the much more evident differences in the transfer function between the excitation point and the radiated sound field.

The inversion of complex impulse responses is not easy, as they are mixed-phase type. The inversion of long, mixed-phase impulse responses is still an unresolved mathematical problem [Mourjopoulos, 1994], so numerical approximations have to be used.

In a first step, the time-domain, least squares technique [Mourjopoulos 1992] was employed, using a simplified algorithm, showing very good results, by also removing the “all-pass” (reverberation) component of the impulse response. In a following step, another simplified technique [Kirkeby et al. 1999] was developed and tested.

2. Impulse Response measurement techniques

The measurements of pressure impulse responses were obtained by using the Aurora system [Farina and Righini, 1997]. 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 [Rife and Vanderkooy, 1989]. The measurements have been repeated in many positions in the bore, inside the instruments, as well as in the flaring bell.

Furthermore, in one case, a sine sweep signal ("stretched pulse") has been used to calculate impulse responses; this novel technique for calculations of IRs yields better results. A signal/noise ratio of almost 80 dB can be achieved [Farina and Ugolotti 1999], and makes it possible to quantify exactly the amount of harmonic distortion [Farina, 2000]. For these reasons, this novel technique will be the preferred one in the prosecution of this research.

3. The inversion of Impulse Responses

3.1 The question of inverting mixed-phase impulse responses

Inverting long, mixed phase impulse responses and creating inverse filters that are causal, stable and of finite length was studied by many authors during the last years [Mourjopoulos 1984, Clarkson et al. 1985]. From their results two general techniques have been developed: the minimum/maximum phase decomposition with separate inversion, and the least squares approximation. In addition, a new technique, which is still under development [Kirkeby et al. 1999], has been tested in one case, but due to problems with the numerical computation, which are still unsolved, it was discarded.

Three possible approaches for the inversion of mixed-phase impulse responses are described here theoretically. The second technique was actually applied in the experimental part of this work, and a more detailed explanation is given for it.

3.2 Minimum and maximum phase signal decomposition in the Z plane

The impulse response is first decomposed into two components: a minimum phase one, containing all the zeroes which fall inside the unit circle in the Z-plane, and a maximum phase component, containing all the zeroes which fall outside the unit circle (it is assumed that there are no zeroes which are located exactly on the unit circle). Whilst the minimum phase component can be inverted easily, the maximum-phase component, since its inverse is unstable, needs to be time-reversed before getting inverted, and then time-reversed again. The acausality introduced by this process has to be eliminated by means of a time delay. Finally, the inverse of minimum and maximum phase component are convolved, getting the inverse filter.

3.3 Time-domain least squares technique

Inversion of an impulse response in the time domain can be accomplished by solving a classic least squares problem. The unknown inverse impulse response (containing N+1 unknown quantities), convolved with the original impulse response, has to approximate a delayed Dirac's delta function δ.

Providing that $h(t)$ and $h_{inv}(t)$, are, respectively, the original impulse response and the inverted one, $h(t)*h_{inv}(t)$ should approximate an ideal Dirac δ pulse as much as possible. This creates an optimisation problem, where the sum of the squared differences between the convolved result and a perfect delta function are to become a minimum.

$$\min\left\{ I=\sum_{k=0}^{2N}\left\{\sum_{i=0}^{N}\left[h(k-N+i)\cdot h_{inv}(N-i)\right]-\delta(k)\right\}^2\right\} \tag{0}$$

$$\left.\frac{\partial I}{\partial h_{inv}(l)}=0\right|_{l=0,1,\dots N} \tag{1}$$

From the calculation in (0), we found:

$$I=\sum_{k=0}^{2N}\left\{\delta^2(k)-2\delta(k)\cdot\sum_{i=0}^{N}\left[h(k-N+i)\cdot h_{inv}(N-i)\right]+ \right.$$
$$\left.+\sum_{i=0}^{N}\sum_{j=0}^{N}\left[h(k-N+i)\cdot h_{inv}(N-i)\cdot h(k-N+j)\cdot h_{inv}(N-j)\right]\right\} \tag{2}$$

that is to say:

$$I=1-2\sum_{k=0}^{2N}\delta(k)\cdot\sum_{i=0}^{N}\left[h(k-N+i)\cdot h_{inv}(N-i)\right]+$$
$$+\sum_{k=0}^{2N}\sum_{i=0}^{N}\sum_{j=0}^{N}\left[h(k-N+i)\cdot h_{inv}(N-i)\cdot h(k-N+j)\cdot h_{inv}(N-j)\right] \tag{3}$$

Since the only term different from zero in $\delta(k)$ is $\delta(N)$=1, we obtain:

$$I=1-2\sum_{i=0}^{N}\left[h(N-N+i)\cdot h_{inv}(N-i)\right]+$$
$$+\sum_{k=0}^{2N}\sum_{i=0}^{N}\sum_{j=0}^{N}\left[h(k-N+i)\cdot h_{inv}(N-i)\right. \tag{4}$$
$$\left.\cdot h(k-N+j)\cdot h_{inv}(N-j)\right]$$

from which, calculating (1):

$$\frac{\partial I}{\partial h_{inv}(l)}=0-2\sum_{i=0}^{N}\frac{\partial}{\partial h_{inv}(l)}\left[h(i)\cdot h_{inv}(N-i)\right]+$$
$$+\frac{\partial I}{\partial h_{inv}(l)}\sum_{k=0}^{2N}\sum_{i=0}^{N}\sum_{j=0}^{N}\left[h(k-N+i)\cdot h_{inv}(N-i)\right]\cdot \tag{5}$$
$$\cdot h(k-N+j)\cdot h_{inv}(N-j)\Big]$$

with the positions:

$$N - i = l \Rightarrow i = N - l$$
$$j = N - l$$

we obtain:

$$\frac{\partial I}{\partial h_{inv}(l)} = -2\sum_{i=0}^{N} \frac{\partial}{\partial h_{inv}(l)}\left[h(N-l)\cdot h_{inv}(l)\right] + \sum_{k=0}^{2N}\left\{2\cdot\sum_{i=0}^{N}\frac{\partial}{\partial h_{inv}(l)}\left[h(k-N+i)\cdot h_{inv}(N-i)\cdot h(k-l)\cdot h_{inv}(l)\right]\right\} \quad (6)$$

that could be written in the following way:

$$\begin{aligned}\frac{\partial I}{\partial h_{inv}(l)} &= -2\cdot h(N-l) + \\ &+ \sum_{k=0}^{2N}\sum_{i=0}^{N-l-1}\left[h(k-N+i)\cdot h_{inv}(N-i)\cdot h(k-l)\right] + \\ &+ 2\cdot\sum_{k=0}^{2N}\frac{\partial I}{\partial h_{inv}(l)}\left[h(k-l)\cdot h_{inv}(l)\cdot h(k-l)\cdot h_{inv}(l)\right] + \\ &+ \sum_{k=0}^{2N}\sum_{i=N-l+1}^{N}\left[h(k-N+i)\cdot h_{inv}(N-i)\cdot h(k-l)\right] = 0\end{aligned} \quad (7)$$

gathering a *N + 1* equations system, expressed by:

$$\begin{aligned}&\sum_{k=0}^{2N}\sum_{i=0}^{N-l-1}\left[h(k-N+i)\cdot h_{inv}(N-i)\cdot h(k-l)\right] + \sum_{k=0}^{2N} 2\cdot\left[h^2(k-l)\cdot h_{inv}(l)\right] + \\ &+ \sum_{k=0}^{2N}\sum_{i=N-l+1}^{N}\left[h(k-N+i)\cdot h_{inv}(N-i)\cdot h(k-l)\right] = h(N-l) \\ &\text{for } l = 0,1,\ldots,N\end{aligned} \quad (8)$$

that in matrix form become:

$$[R]\cdot\{h_{inv}\} = \{g\} \quad (9)$$

where the l^{th} element of $\{g\}$ is expressed by $h(N-l)$, whilst the matrix $[R]$ is composed by the elements:

$$\begin{aligned}R(l,j) &= \sum_{k=0}^{2N}\left[h(k-j)\cdot h(k-l)\right] \quad \text{for} \quad l \neq j \\ R(l,l) &= \sum_{k=0}^{2N}\left[h^2(k-l)\right] \quad \text{for} \quad l = j\end{aligned} \quad (10)$$

The matrix is symmetric, i.e. $R(l,j) = R(j,l)$; Furthermore, since $h(t)$ has been defined with a *zero-padding*, on the left and right side, of *N* zeros, and since *l* varies between the values [*0* , *N*], in the diagonal every element of the matrix are equal. Thus, for *l = 0* we obtain:

$$R(0,0) = 2\cdot\sum_{k=0}^{2N}\left[h^2(k)\right] = 2\cdot\sum_{k=0}^{N}\left[h^2(k)\right] \quad (11)$$

whilst for *l = N* we have:

$$R(N,N) = 2\cdot\sum_{k=0}^{2N}\left[h^2(k-N)\right] = 2\cdot\sum_{k'=-N}^{N}\left[h^2(k')\right] = 2\cdot\sum_{k'=0}^{N}\left[h^2(k')\right] \quad (12)$$

after taking into account just the terms different from zero.

Besides, the terms outside of the diagonal are depending only from the distance $|l - j|$, not separately from *l* and *j*. With the position *d = l - j*, we obtain:

$$R(l,j) = \sum_{k=0}^{2N}\left[h(k-j+l-l)\right]\cdot h(k-l) = \sum_{k=0}^{2N}\left[h(k+d-l)\right]\cdot h(k-l) \quad (13)$$

not depending from *l*. With the position *l = 0*, he have:

$$R(l,j) = \sum_{k=0}^{2N}\left[h(k-j+l)\right]\cdot h(k) \quad (14)$$

that is to say:

$$R(l,j) = R(l+1, j+1) \quad (15)$$

This is a very simple matrix, called *Toeplitz matrix*, that allows a simplified solution of the problem following an algorithm suggested by Wiener-Levinson, and modified by Durbin. The matrix has the following structure:

$$[R] = \begin{bmatrix} r_0\, r_1\, r_2 \ldots r_N \\ r_{-1}\, r_0\, r_1 \ldots r_{N-1} \\ r_{-2}\, r_{-1}\, r_0 \ldots r_{N-2} \\ \ldots\ldots\ldots\ldots \\ r_{-N}\, r_{-(N-1)}\, r_{-(N-2)} \ldots r_0 \end{bmatrix} \quad (16)$$

It is necessary to point out that the vector *r*(*N*) represents the auto-correlation function of *h*(*t*)

The results are accurate only if the length N+1 of the FIR inverse filter exceeds the length of the original impulse response, the optimal length being usually around twice that of *h*(*t*).
When N becomes large, it is possible to significantly reduce the memory storage requirements by using the mathematical properties of the Toeplix matrix R. In practice, only the first row of the matrix needs to be stored, and the solution is obtained by an iterative process, working “in place”. This way it was possible to create inverse filters with a length of up to 32 kpoints in a few seconds. No special effort was required for developing the computer code for the inversion of the Toeplitz matrix, as the above outlined procedure is commonly found inside packages of publicly-available subroutines (i.e. Numerical Recipes).

3.4 Frequency-domain regularisation technique

This approach was first developed at the ISVR [Kirkeby and Nelson 1998], and further refined with the co-operation of one of the authors [Kirkeby et al. 1999], introducing a frequency-dependent regularisation parameter.

The idea is incredibly simple: after transforming the original impulse response to the frequency domain by means of FFT, it is inverted by simply taking the (complex) reciprocal at each frequency, and then re-transformed back to the time domain by means of IFFT. Obviously this does not work with mixed-phase signals: to make it feasible, a small regularisation parameter (positive and real) is added to the denominator while taking the reciprocal. This small quantity, called ε, can be varied along the frequency axis, to ensure more accurate inversion in the most interesting frequency band, avoiding to waste the processing power of a limited length FIR filter at very low or high frequencies.
The original response h(t) is first FFT transformed:

$$C(\omega) = FFT[h(t)] \tag{17}$$

Then the complex spectrum C is inverted:

$$C_{inv}(\omega) = \frac{Conj[C(\omega)]}{Conj[C(\omega)] \cdot C(\omega) + \varepsilon(\omega)} \tag{18}$$

And the result is back-transformed to time domain

$$h_{inv}(t) = IFFT[C_{inv}(\omega)] \tag{19}$$

Although Kirkeby attempted to find an automatic way of optimising the value of the regularisation parameter ε, it resulted that the computation of the inverse filter is so fast that it is more convenient to adjust the optimal value of ε by a trial-and-error approach. The actual implementation requires that the user chooses the proper values of these parameters:

- regularisation parameter in the central frequency band (usually very small)
- regularisation parameter in the extreme frequency bands (usually quite bigger)
- frequency of the transition between low and central bands
- frequency of the transition between central and high bands
- bandwidth of the transitions (in octave units)

A proper logarithmic interpolation between the two values of ε is automatically done during the cycle that computes the above inversion (18) at various frequencies. No matrix operation is required in this case, and therefore the computation is incredibly fast (less than 0.1 s for the inversion of a 32k-points impulse response).
This technique revealed to be very good for the inversion of the response of systems which are reasonably damped and have regular frequency response, such as loudspeakers [Kirkeby et al., 1999]. In this case, however, it turned out to be quite difficult to obtain satisfactory inverse filters, because the impulse response of a trumpet contains very sharp peaks with very little damping, which cause the Kirkeby inverse filters to become unacceptably long and slightly unstable.

Fig. 1 - The measurements of IRs on "Vincent Bach" trumpet

4. Convolution by Frequency Domain Processing

The Convolution algorithm can be implemented very efficiently making use of the Frequency Domain Processing technique: the well known “select-save” algorithm [Oppenheim and Schafer 1975] can be used for this task.
For the following experiments, the Aurora system [Farina and Righini 1997] has been utilised. Real time convolution is possible with this system, especially for impulse responses which are not very long, as in the case of trumpets: on today’s personal computers, real-time convolution can be sustained with an impulse response length up to 256000 taps, much longer than the impulse response of a trumpet, which is typically less than 16 kpoints at a sampling rate of 44.1 kHz.

5. Experiments

5.1. Impulse Response Measurements

The impulse responses of three widely employed trumpets have been measured, namely a Vincent Bach, tuned in C, a Yamaha, tuned in B-flat, and a "Yamaha Custom piccolo trumpet", tuned in B-flat. The measurements have been conducted with the technique presented in chapter 2. In order to compare results for quite different wind instruments, measurements have also been made with a silver flute.
The first two trumpets are rather similar, while the last one is quite different, both in shape and in sound timbre.
The measuring points considered in each instrument, covered almost all the bore. They have been chosen with a distance of 5 mm each other, beginning from the flaring bell until to reach the cylindrical tube. Furthermore, the “reference” near-field radiation point was measured, located on the axis of the instrument at a distance of 1m.
The main goal for taking so many measurements at closely spaced positions was to get data suitable for the computation of the sound intensity inside the trumpet, treating each consecutive pair of microphone positions as a pressure-gradient sound intensity probe. This technique was already employed in the study of the sound field inside standing wave tubes [Farina and Fausti, 1994], and it will lead, in the prosecution of this research, to a deeper understanding of the acoustical phenomena inside the trumpet.
Some other tests have been conducted with different conditions of the trumpets, e.g. weights on the valves, to check differences in sound timbre.

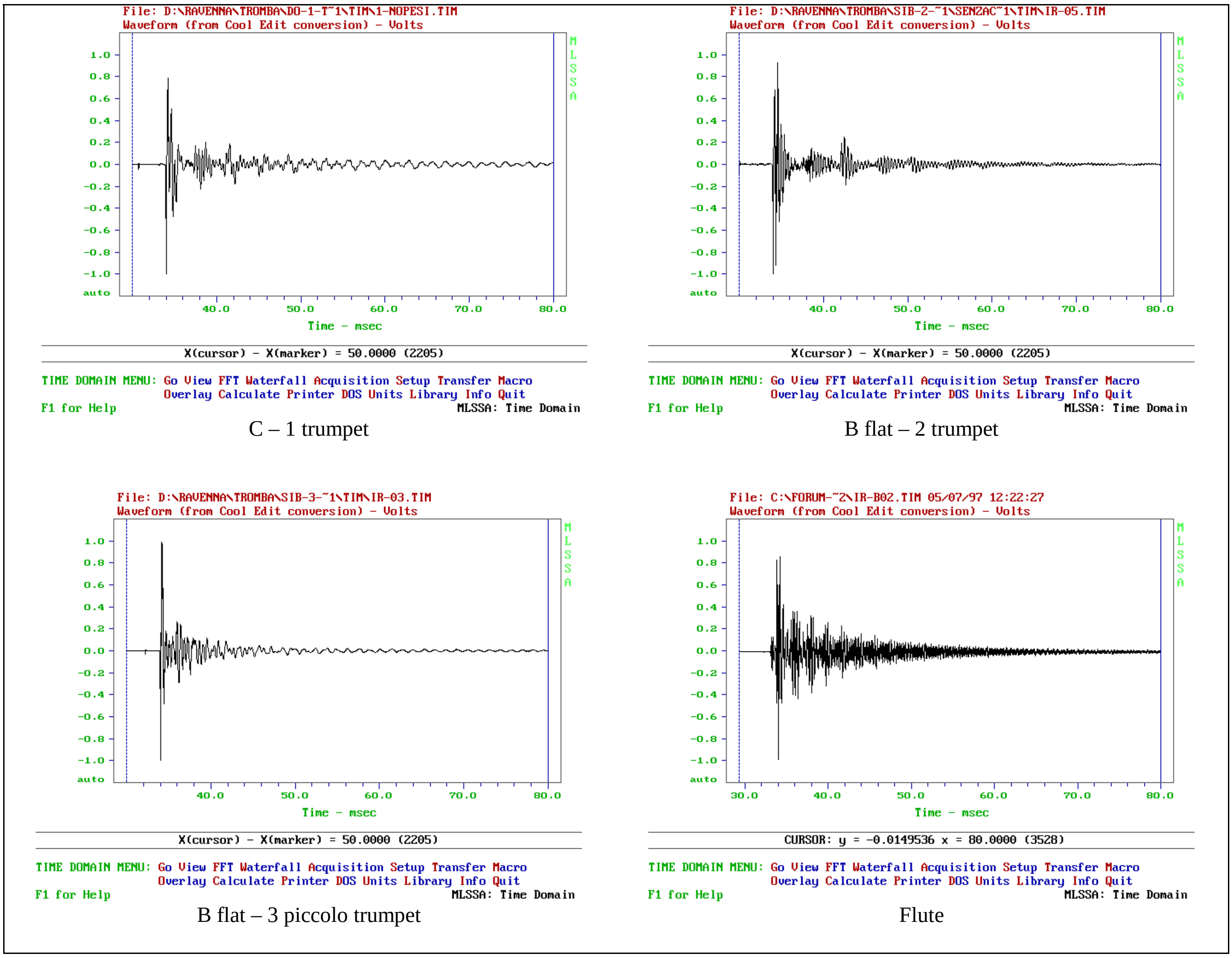

Fig. 2 - IRs measurements in different trumpets.

From the time-frequency responses of the 4 instruments it can be observed that the flute is remarkably different, whilst the differences among the trumpets are not so evident.

5.2. Recording of music samples

Different pieces of music were performed in a semi-anechoic environment in the laboratory of University of Bologna. Two free-field microphones were used, placed at two different positions. One, inside the final part of the flaring bell of the trumpet, and another one meter in front of the instrument, at exactly the same position where the “reference” impulse response has been measured. The recordings were digitally stored as .WAV files on the PC hard disk using a professional digital sound board (Layla by Echo).

5.3. Creation of the inverse filters and deconvolution of “dry” excitation signals

An obvious way to get the 'anechoic' input signal for further convolution with IRs is to directly measure the sound-pressure signal in the mouthpiece. However this technique requires miniaturised microphones that have to be inserted in the mouthpiece, which is unfeasible without severely disturbing the delicate not-linear processes happening inside the mouthpiece.

The better technique for obtaining the dry excitation signal revealed to be convolving the signal recorded at the point near to the end of the bell with the inverse filter obtained by the impulse response measured exactly in the same position.

The same process, applied to the signal recorded in the radiated near field, at 1m from the bell, 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.

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.

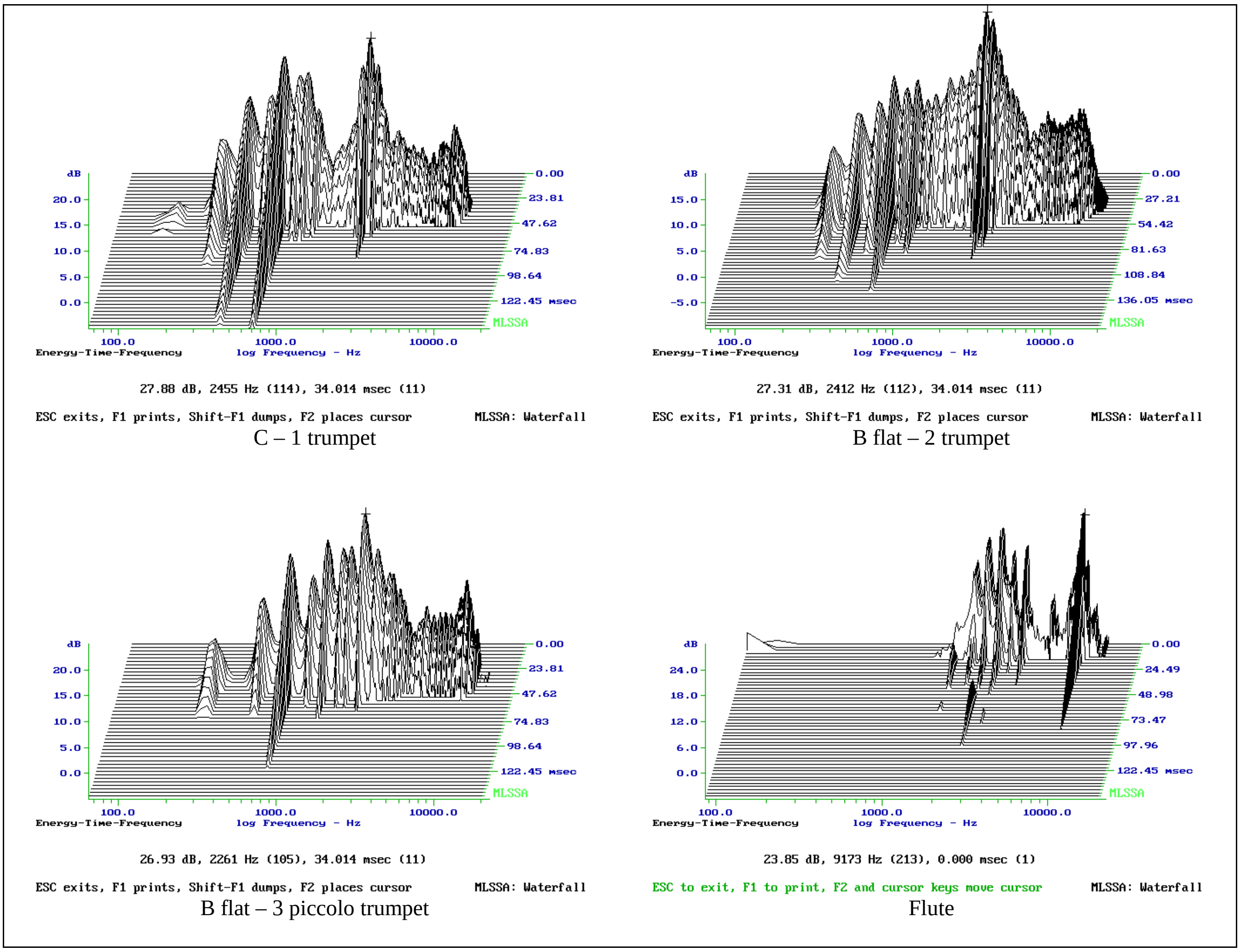

Fig. 3 - Energy-Time-Frequency responses of the four instruments

If this dry excitation signal is convolved with impulse responses of different instruments then their sound characteristics can be simulated. If the impulse responses used for this purpose are covering the complete transmission system between a certain instrument's mouthpiece and the listeners ear, then all acoustical influences of that instruments as well as of the room, where the impulse response has been measured, are included in the reconstructed sound. If impulse responses, measured between mouthpiece and bell of an instrument are used for sound reconstruction, then an anechoic radiated sound will be produced. In order to reproduce the complete sound experience of a listener this anechoic radiated sound needs to be reverberated, and this can be accomplished by reproduction through a loudspeaker in a real room, or by a further convolution process with a room's impulse response for headphone listening.
While different inversion techniques were tested in a previous work [Farina et al., 1998], for these experiments the time-domain least-squares inversion technique was employed. The impulse response of the C-tuned trumpet was inverted, after some minor manipulation required to remove electrical cross-talk and to reduce its length to less than 8192 point. The computed inverse filter was 32768 points long. 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. In fig. 4 the whole inversion process is reported: first the original impulse response is shown, then the corresponding inverse filter obtained with the least-squares method, and finally a check is made, convolving the original impulse response with the inverse filter, and checking that the results is very close to an ideal Dirac's delta function.

5.4. Subjective tests for validation of the virtual reconstruction by convolution

In order to validate the procedure, a comparison with the second recording has been made. The "dry" input signal, obtained from the convolution with the inverse filter already described, has been re-convolved with the other impulse response measured outside the trumpet, one meter in front of the bell, and compared with the sound signal recorded at the same position outside the trumpet. In the following figure it is possible to compare the two waveforms, the original recording and the "virtual" one.
The only appreciable difference is a slightly more "reverberant" reconstructed signal, no timbric alteration can be perceived.
In practice, it turned out that the recorded and re-convoluted sound samples are indistinguishable, especially when they are reproduced through loudspeakers in a normally reverberant room, where the minor differences between the true recording and the

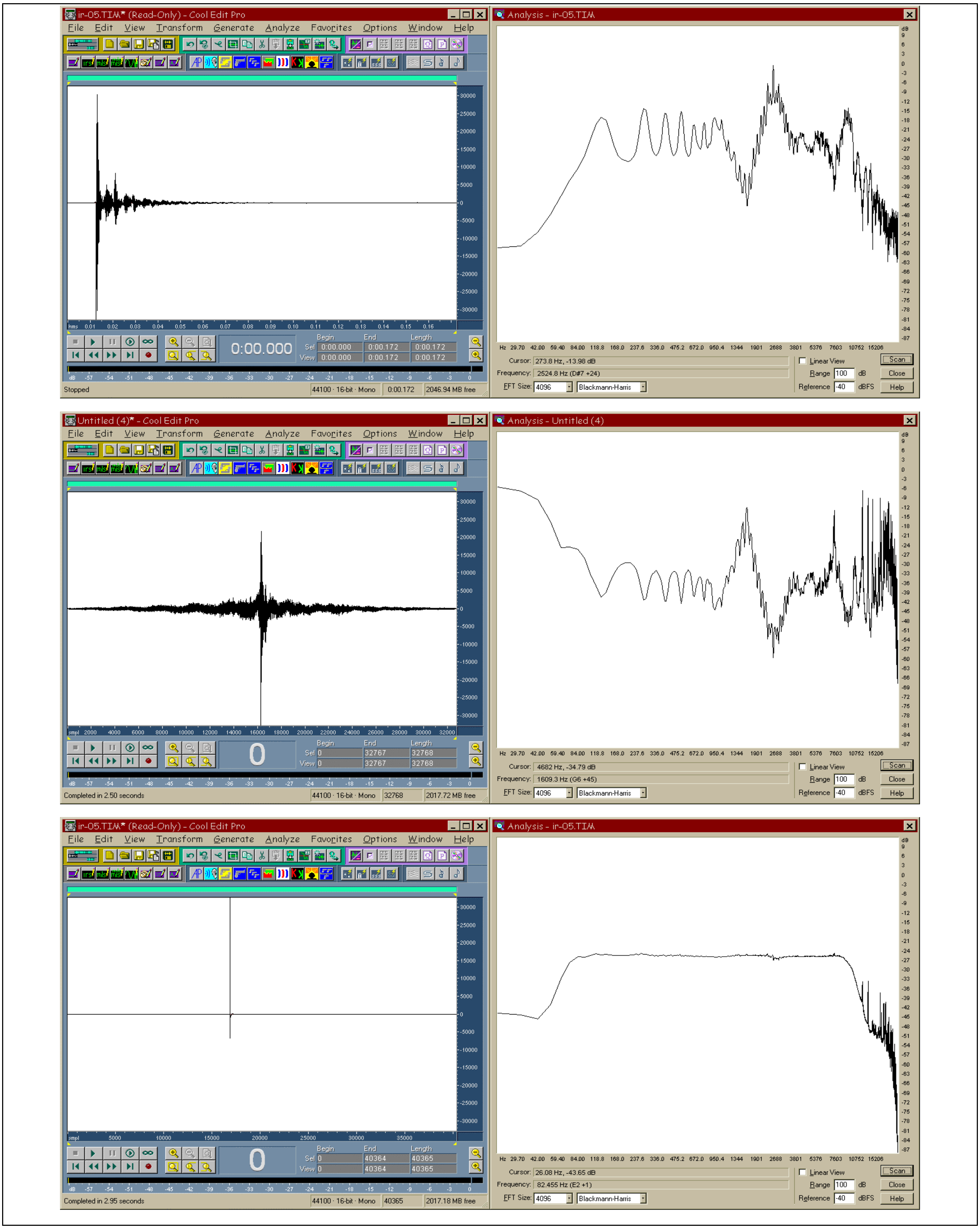

Fig. 4 - I.R. (above), inverse I.R. (mid) and convolution check (below) for trumpet “V. Bach”

virtual one obtained by convolution are covered by the acoustic field of the listening environment. This was tested with a panel of sharped-ear musicians, who were attending to the School of Cultural Heritage Preservation of the University of Bologna.

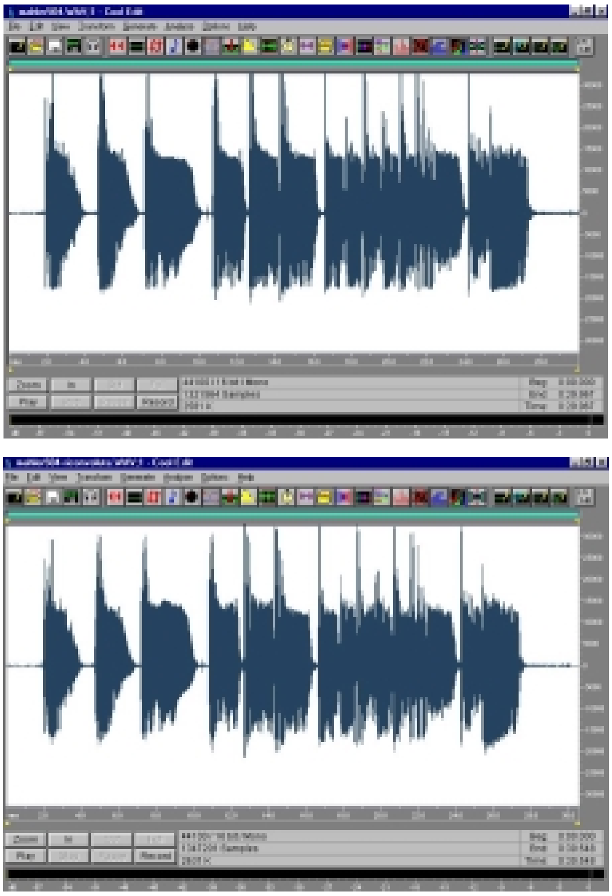

Fig. 5 - Recorded signal and virtual reconstruction obtained by convolution

In practice, a pair-comparison test was performed, in which 14 subjects had to simply state if the two sound samples of each pair were considered different or equal. Each test consisted in three pairs, one containing a “virtual” sound and an original recording, the other two pairs being simply “controls”, containing a pair of “truly equal” sounds (the original recording repeated twice) and a pair of “truly different” sounds (virtual convolutions with the IRs of two different trumpets).
The following table shows the average “equality” score, with its standard deviation, for each of the three pairs:

Pair	“Equality” score (%)	Standard Deviation (%)
Real/Virtual	89	7
“Truly equal”	96	6
“Truly different”	18	11

Even without advanced statistical tests, it can be concluded from the above-reported subjective results that the difference between the scores of “truly equal” and Real/Virtual pairs is not significant: the virtual reconstruction of trumpets can be considered substantially indistinguishable from the original recordings. Furthermore, the difference between two similar, but not completely identical instruments is clearly perceived by sharped-ear listeners.

6. Conclusions

The proposed technique enables the creation of virtual wind instruments by the convolution method, similarly to what was already done with string instruments, i.e. violins, [Farina et al., 1998]. The MLS method revealed to be satisfactory for the measurement of impulse responses of trumpets, although in future this technique will probably be superseded by the "stretched pulse method".
The least-squares method for the inversion of impulse responses revealed to be suitable for the calculation of inverse filters which are required to obtain the “dry” excitation signal, corresponding to the sound pressure in the mouthpiece; once the excitation signal has been obtained, it can easily be convoluted with calculated or measured IRs of different instruments, producing “virtual” wind instrument sound. The possibility of measuring impulse responses and making modifications on them permits to reconstruct real trumpets and to investigate the influence of different materials on the sound quality of wind instruments. Furthermore, the impulse response can be numerically estimated (for example through a structural FEM model coupled with an acoustical BEM model), and this makes it possible to listen at designed, but not existing, instruments.
The only evident limitation in the present work was the fact that the whole process was applied to an “invariant” system, which means that the valves were maintained not depressed. Fortunately, although a trumpet cannot be in general considered an “invariant” system due to the fact that the player acts on the valves during his performance, nevertheless it can be considered a system that can vary only between a limited number of repeatable states, given by the possible combinations of depressed valves. This means that it is possible to extend the procedure here described to a more realistic “variable” system, provided that separate impulse response measurements are made for each possible valve configuration of the instruments, and that during the performance the valves position is sampled together with the audio signal coming from the microphones. Then, each segment of signal, which was produced by a particular valves combination, needs to be convolved with the corresponding inverse filter for producing the “dry” excitation signal. And, during the reconvolution with the set of impulse responses of a different trumpet, again each time segment, which has been marked with a particular valve configuration, has to be convolved with the corresponding impulse response.
Such an extension to the method will be developed in the prosecution of this research.

7. Internet References

Some of the sound samples described in this paper (impulse responses, inverse impulse responses, dry and convolved music pieces) can be downloaded from the following URL:
HTTP://ciarm.ing.unibo.it/researches/trumpet

The Aurora plugins can be freely downloaded from:
HTTP://www.ramsete.com/aurora

The personal web pages of the authors are: **HTTP://pcfarina.eng.unipr.it** and **HTTP://ciarm.ing.unibo.it/lamberto**

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