Acoustic characterisation of "virtual" musical instruments: using MLS technique on ancient violins

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Abstract

In this paper the realisation of "virtual" musical instruments is analysed, in which the instruments are treated as linear systems, characterised by their impulse response. The impulse response is treated as a numerical filter, which is convolved with the force excitation signal coming from the chords, producing a signal containing all the acoustic characteristics of the instruments, avoiding all non acoustics phenomena.
The aim of this work is multiple: the "virtual" instruments can be used in subjective listening tests for the evaluation of the "sound quality" of different instruments, as reported in this paper. Other possible uses are for the (real or virtual) restoration of ancient instruments, and for preliminary listening test on new designed instruments.
Various measurements techniques of the impulse response have been tested, employing different transducers and numerical analysis.
For validating the accuracy of the new technique, a subjective listening test has been conducted. Some original music samples have been played by Maestro Marco Fornaciari, on 3 different violins and a viola in an anechoic chamber; both the acoustic radiated signal and the excitation of the chords on the bridge have been digitally recorded.
The results of the listening tests confirmed the excellent degree of similarity between the direct acoustic recording and the convolution technique.

1. Introduction

The violins are not mechano-acoustic linear transducers, but their non linear characteristics are due to the interaction between string and bow, and to the vibration of strings. However, the most important difference in timbric perception among ancient and modern violins depends on the body of the violins that behaves as a source of radiation, depending on the characteristics of the wood sound chest of each instrument.

The problem was circumvented taking into account only the system included between the bridge of the violin (in which the strings are in contact with the body) and the sound field radiated into an anechoic environment: this system is surely linear, as a consequence of the small displacements of the structure. The input signal, coming from the strings to the bridge, must be measured separately with an appropriate technique.

The impulse response can be obtained *directly* (excitation of the violin on the bridge with a known **force F** and measurements of the impulse response as **sound pressure p**) and *inversely*, using the reciprocity technique (excitation of the sound field with a known **volume velocity Q** generated by a loudspeaker, and measurements of the impulse response as **velocity u** of vibration of the bridge). These two different techniques yield the same results, provided that the requirements dictated by mechano-acoustics reciprocity theory are fulfilled (Fahy 1985): these ones require a point, omnidirectional sound source located exactly in the same place where the microphone is located, and a velocity transducer located exactly in the same position of the structure where the force load is applied.

The following equation state the theoretical equivalence between the Frequency Response Function evaluated in the two ways:

$$\mathrm{FRF} = \frac{p(\omega)}{F(\omega)} = \frac{u(\omega)}{Q(\omega)} \qquad (1)$$

Both techniques have been employed in the present work, but actually the direct method seems capable of superior performances than the reciprocal one, due to the better signal-to-noise ratio and to the greater linearity of the transducers.

The second problem is to properly measure the input signal (force), to make the “anechoic” input signal available to be used for convolution. The placement of force transducers (piezoelectric load cells) between the strings and the bridge during the violinist’s performance resulted unfeasible. However, a velocity transducer can be placed over the bridge, causing only a limited disturbance to the player: this is made with a phonograph needle, supported by a flexible arm. The sound track coming from the velocity transducer is not, however, what is required: in fact this is the response of the mechanical system, excited in a point with a complex mobility function (velocity versus force).

Two indirect techniques were developed to reconstruct the “anechoic” signal: the first is based on an inverse filtering of the velocity track, the second on the acoustic signal recorded by the microphone in the anechoic chamber. In the first case the inverse filter is the inverse of the mobility function, that is the mechanical impedance of the excitation point (force vs. velocity). In the second case, the inverse filter is the inverse of the mechano-acoustical radiation impulse response (acoustic pressure vs. force). The inversion of these complex functions is not easy, as they are mixed-phase types. The inversion of long, mixed-phase impulse responses is still an unresolved mathematical problem (Mourjopoulos 1994), so approximate solutions have to be used. In a first step the inverse IR was obtained with the Neely & Allen (1979) technique, that invert only the minimum phase component of the impulse response, obtained taking the modulus of its Fourier transform and forcing the phase to zero. This perfectly removes the timbric character of the violin on which the music sample was played, but still leaves in the signal a reverberation that can be heard. In a following step, the mean least squared technique (Mourjopoulos 1992) was employed, and it led to better results, removing also the “all-pass” component of the impulse response.

2. Theory

2.1 Impulse Response measurement techniques

The measurements of the mechano-acoustic impulse responses were obtained using a MLS (Maximum Length Sequence) signal, generated by a PC fitted with an A/D board equipped with a hardware MLS generator and software for deconvolution of the response. For direct measurements this signal, properly amplified, was sent to a Dunnwald-like copper wire force transducer (Dunnwald 1985): the force applied on the bridge was proportional to the current passing through the wire, as it was located in a strong, permanent magnetic field.
The violin was placed in the anechoic chamber of the Cremona Violin Making School, fitted with proper supports, microphones and preamplifiers. The signal coming from the preamplifier 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 & Vanderkooy 1989). Fig. 1 shows a schematic diagram of this direct measurement technique.

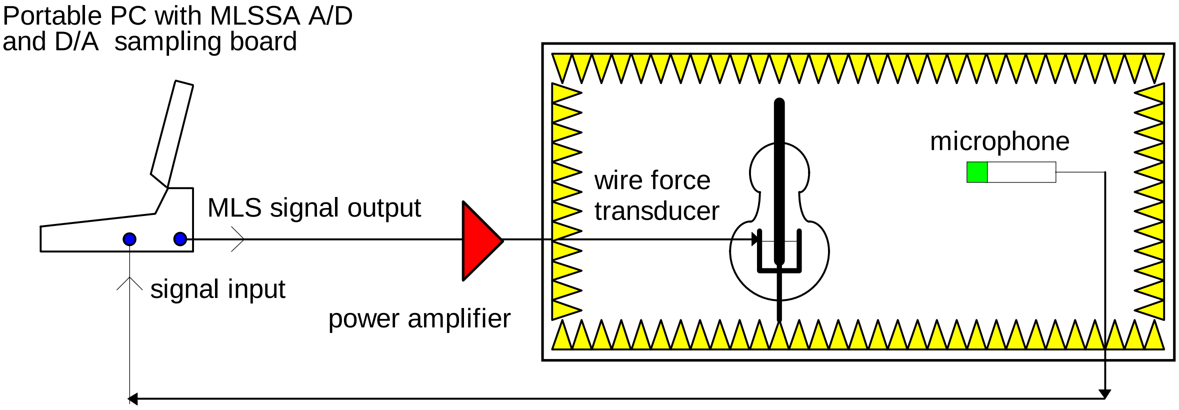


Fig. 1 - schematic measuring system for direct method

Also a reciprocal scheme was used, as shown in fig. 2. In this case, the MLS signal was fed into a loudspeaker placed in the anechoic chamber approximately in the same position previously occupied by the microphone. The velocity response of the violin bridge was detected by a phonograph needle, whose electric output was cross-correlated with the original MLS signal.

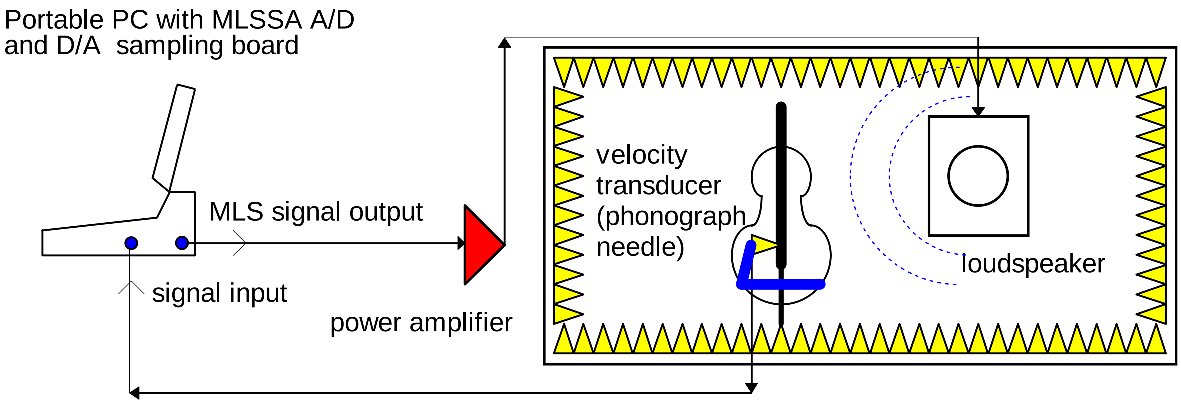


Fig. 2 - schematic measuring system for reciprocal method

Fig. 3 shows a typical measurement result according to the direct technique, while fig. 4 shows the result of a reciprocal measurement on the same violin. Both the time domain and frequency domain representations are shown. It is evident that the results are not equal: this is certainly due to noise contamination problems evident in the reciprocal measurement. The authors think that this “noise” is not actually acoustic noise present in the environment (the anechoic chamber has a background noise lower than 20 dB(A)), but it is a mathematical artefact due to non-linear distortion in the transducer chain, mainly in the loudspeaker and in the phonograph needle pickup (Vanderkooy 1995).

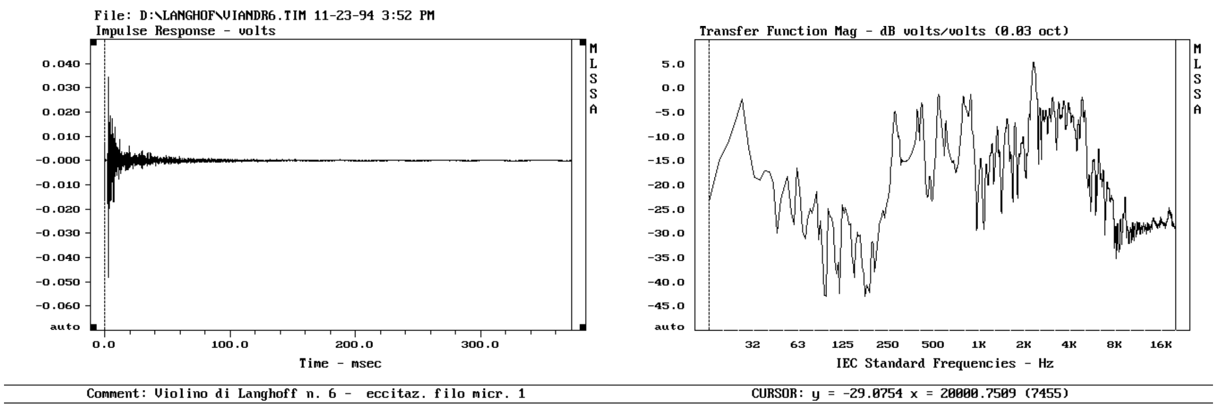


Fig. 3 - Impulse response and Frequency Response Magnitude - Direct method

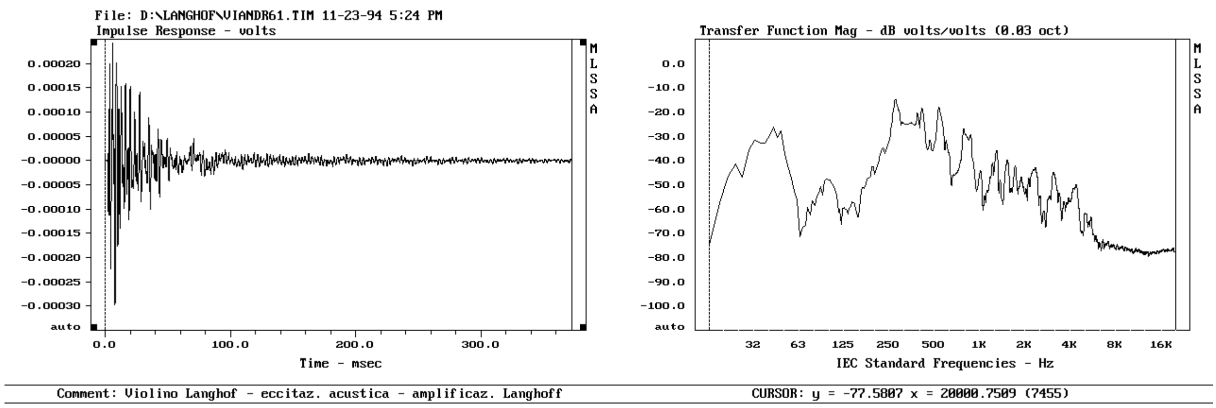


Fig. 4 - Impulse response and Frequency Response Magnitude - Reciprocal method

For this reason only the direct-type impulse response measurements were considered in the following.

2.2 "Anechoic" recording of input force signals

To compare the responses of different violins, an anechoic input signal is required, which is then convoluted with the impulse response of each "virtual" violin, producing a filtered signal containing the whole characterisation of that particular instrument. Then a pair-type comparison technique can be used to subjectively assess the perceptible differences between the responses of different violins.
The obvious solution consists of measuring the force caused by the vibrating chords on the bridge, during a musical performance. However this is not an easy task: miniaturised load cells have to be inserted between the chords and the bridge, and their mass and stiffness are certainly too large to prevent from causing any change in the dynamic response of the instrument. For this reason, it was chosen an indirect technique, in which the input force signal is reconstructed by inverse filtering of response signals. Two kinds of response signals during musical performances were simultaneously digitally recorded on a 2-channel DAT machine and transferred to the hard disk of a PC through an optical digital link: the velocity of the top of the bridge (by the phonograph needle pickup) and the acoustic pressure inside the anechoic chamber.
In principle, the input force can be obtained by both these response signals (pressure or velocity), provided that the input/output transfer function of the system is accurately measured, and that it is possible to create an inverse filter to deconvolve out this transfer function from the response signal.
The transfer function for the acoustic pressure signal is the impulse response already measured as explained in the previous paragraph; the velocity vs. force transfer function is the mechanical mobility of the bridge, and this can be measured by exciting the bridge with the Dunnwald force transducer and measuring the velocity output by the phonograph needle. in This way of operating has the advantage that, during the deconvolution process, the response of the transducers (the phonograph pickup and the Dunnwald force transducer) are filtered out together with the response function of the particular violin over which the music was played.
In theory these deconvolution techniques reproduce an "anechoic" input signal which has lost any colouring coming from the particular violin employed, and this signal can be used for convolution with any other violin's impulse response. Obviously this does not happen in reality, as the performer changes his way of playing depending on the sonority of the instrument used, as he tries to correct for defects of it, or is anyway conditioned from some particularities of the violin (not necessarily acoustic, but for example related to the instrument embracement, to its tactile feeling, to the vibrational feedback through the shoulder and the chin). However, a performance on a medium-quality violin, with a not particularly evident character can produce a reasonably "universal" input signal, suitable for further convolution with different violins. The employment of the same input signal on different violins permits to produce more evident acoustic differences than separate performances of the same music piece over these instruments, as no "compensation" is made by the performer.

2.3. Inversion of mixed-phase impulse responses

The latter question is how to invert long, mixed phase impulse responses, producing inverse filters that are causal, stable and of finite length. The question was addressed in the last years by many authors, but the most efficient results are those of Mourjopoulos (Mourjopoulos

1984, Clarkson et al. 1985). He proposes two general techniques: the minimum/maximum phase decomposition with separate inversion, and the least squares approximation.

2.3.1 Minimum and maximum phase signal decomposition

Following the first technique, the original impulse response is first decomposed in two components: a minimum phase one, containing all the zeroes which fall inside the unit circle on the Z-plane, and a maximum phase component, containing all the zeroes which fall outside the unit circle (it is assumed that no zeroes fall exactly over the unit circle).
The decomposition of a mixed-phase impulse response in the minimum and maximum phase component is not easy. It was tempted both by homomorphic decomposition (Mourjopoulos 1984) and by complex cepstral separation (Clarkson et al. 1985), but in general the results are poor.
Once the components have been separated, the minimum phase component can be directly inverted, because simply taking the IFFT of the reciprocal of its FFT transform yields a finite, stable and causal inverse impulse response. The same approach is unsuccessful for the maximum-phase component, as its inverse is unstable (the response is not decaying to zero with increasing time, but the values get larger and larger...). However, if the maximum-phase component is time-reversed, then its response becomes stable, but infinite and acausal; after the inversion, the inverse impulse response is time-reversed again. If the time window is long enough, the truncation of the inverse response does not cause any appreciable error; furthermore, the acausality can be corrected adding a simple time delay, which causes no practical problems to not-real-time processing. The inverse of the minimum and maximum phase components is then convoluted, producing the final approximate inverse filter.

2.3.2 Approximate inversion of the zero-phase component

The decomposition of the impulse response in minimum and maximum phase components resulted difficult, because the phase response of the system exhibits many wraparounds, and it is difficult to unwrap correctly these jumps.
Nevertheless, the Neely and Allen approximate technique resulted in a viable method to remove at least the timbric coloration caused by the particular violin used. Following this technique, the original impulse response is decomposed in a zero-phase component, obtained taking the modulus of the frequency response, and an all-pass component, containing the phase of the frequency response with unitary modulus. The first is obviously a minimum phase component, and it can be inverted directly by taking the inverse Fourier transform of its reciprocal. The second is again mixed-phase, and cannot easily be inverted: it carries just “reverberation” information, not any timbric coloration, and it is left not inverted.
It must be noted, however, that prior to applying the inversion of the zero-phase component a small amount of “frequency smoothing” is required, in order to reduce the length of the inverse filter; this allows the creation of an inverse impulse response which is stable within the length of the original impulse response.
Applying the zero-phase inverse filter to the measured response signals, the timbric coloration of the violin is removed, but the “reverberation” is still present. When this “anechoic” signal is convoluted again with a violin impulse response, the spectral coloration is properly restored, but the reverberation is a little too big. However this effect is noticeable only comparing signals directly recorded in an acoustically-anechoic space with “virtual” violin convolutions. If the comparison is repeated in a non-anechoic acoustic space (i.e. in a room), the directly recorded acoustic signal becomes indistinguishable from the original input signal

convoluted with both the violin's impulse response and the room's impulse response; this because the room's reverberation is usually larger than the violin's reverberation (1s against 0.2 s), and it masks completely the latter one.

2.3.3 Least squares technique

The last technique set up a classic least squares problem, with an unknown inverse impulse response (containing N unknown quantities) that, convoluted with the original impulse response, has to approximate a delayed Dirac's delta function. Imposing the minimisation of the sum of the squared differences between the convoluted result and the wanted delta function, a set of N linear equations is formed:

$$[R(i,j)] \cdot \{h_{inv}(i)\} = \{h(d-i)\} \tag{2}$$

in which each row of the matrix [R] is a sample of N points taken from the autocorrelation function of the original response $h(\tau)$, starting at time 0 for the first row and one sample left for each subsequent row (-1 for the second row, and so on); the column of known terms is a time-reversed, delayed copy of the original impulse response $h(\tau)$ (d is the delay in samples, usually taken around N/2).

Accurate results can be obtained only if the length N of the inverse filter exceeds the length M of the original impulse response. A reasonable value for N is the double of M.

For efficient computations, when N becomes very large, it is possible to significantly reduce the memory storage requirements by using the mathematical properties of the matrix R, which is a symmetric Toeplitz matrix. A recursive algorithm for fast inversion of Toeplitz matrixes can be found in many standard mathematical libraries. This way it was possible to create inverse filters of length up to 32 kpoints in a few minutes.

2.4 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 (Oppheneim & Schafer 1975) can be used for this task.

Although specialised hardware tools for implementing this algorithm in real time already exist (Connoly 1992, Connoly 1995), a complete software implementation of the convolution process was used in this case, based on a fast convolution program (Farina 1993, Farina 1995). The processing speed of a modern PC is high enough to give real time convolution with impulse responses of even larger length than that required to completely describe the violins.

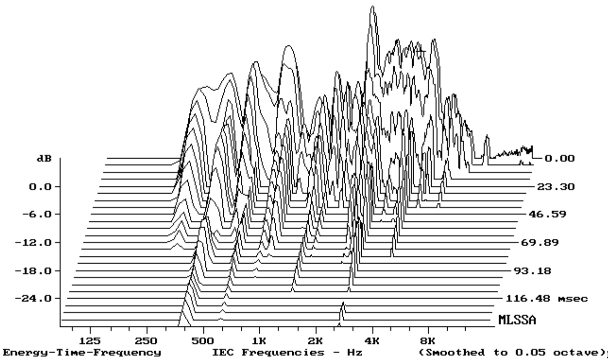
3. Measurements

3.1 Impulse Response Measurements

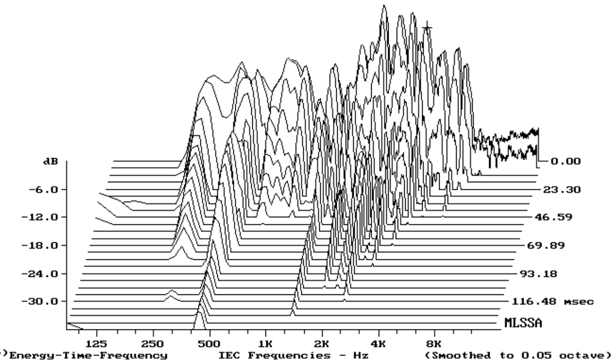
The impulse responses of three violins and of one viola were measured with the direct technique presented in paragraph 2.1. The three violins were identified by the name of the builder:

- Calcanius
- Klotz
- Langhoff

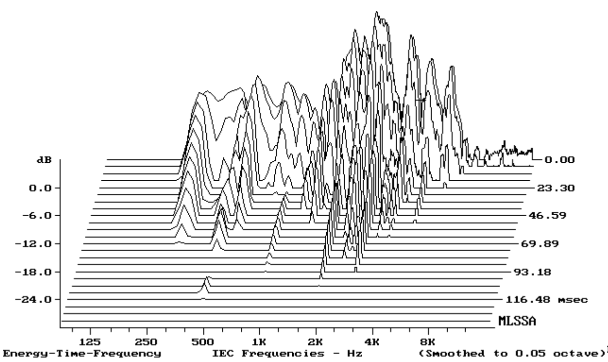
The viola was introduced only to have a very different instrument, making audible to everyone the different timbric coloration.
Fig. 5 reports the time-frequency responses of the 4 instruments. It can be observed that the viola is noticeably different, whilst the differences between the three violins are not so clearly evident from these waterfall representations.



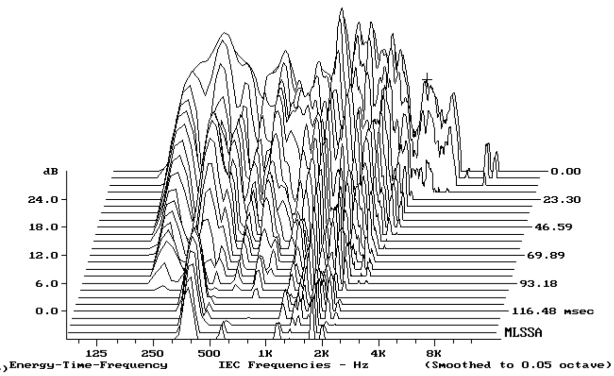
Langhoff Violin



Calcanius Violin



Klotz Violin



Viola

Fig. 5 - Energy-Time-Frequency responses of the four instruments

3.2 Recording of music samples

Four music samples were played on the three violins: they were two music pieces of Paganini, one of Bach and the last of Mozart. The violins were kindly played by Maestro Marco Fornaciari inside the anechoic chamber of the Violin Making School of Cremona.
A 2-channel DAT recorder was used: the “Left” channel was connected to the phonograph needle pickup, placed on the violin’s bridge. The “Right” channel was connected with the free-field microphone, placed inside the anechoic chamber in the same position relative to the violin as when measuring the impulse responses.
The DAT recordings were digitally transferred as .WAV files on the PC hard disk through an optical linkage.

3.3 Creation of the inverse filters for the deconvolution of “anechoic” signals

Also the mobility functions of the 4 instruments were measured. However, musical trials were conducted only over the three violins, and furthermore the *velocity* music tracks recorded with the phonograph needle were noisier than the acoustic pressure tracks. So the inversion of the

mechanical-acoustic impulse response function was preferred to the inversion of the mechanical input mobility.
Both the zero-phase and least-squares inversion techniques were employed. The first resulted in a faster computation and a more robust procedure, and thus was preferred for the subsequent subjective tests. On the other hand the latter technique, when properly applied, resulted in a more accurate validation test.
Applying the inverse filter to the “right channel” acoustic recordings, “anechoic” input signals were obtained.

3.4. Validation of the inverse filtering to reproduce the “anechoic” input signal

To validate the procedure of extracting the “anechoic” impulse signal from the microphone recordings taken in the anechoic chamber, a preliminary test was conducted.
The reconstructed “anechoic” samples were convoluted again with the impulse response of the same violin on which they were measured. These signals are revealed almost indistinguishable from the original ones when listened to in a normally reverberant space, whilst in headphone listening a little increase in the reverberation can be evidenced for the re-convoluted signals (Farina et al. 1995a). In any case, the timbric perception was almost perfect, as already shown in Farina et al. (1995b) and in Langhoff et al. (1995), and this is the most important aspect for violins.
On the other hand, convoluting the same input signals with the impulse response of the other three instruments yielded noticeably different results as shown by the following subjective tests.

4. Subjective tests to compare the acoustic quality of Violins

For a long time a lot of people have been studying the acoustic characterisation of musical instruments using conventional methods (i.e., by comparing the music played on different instruments); now, by using the proposed convolution technique, it is possible to correlate the objective acoustic properties of violins with subjective evaluations without the need to collect dozens of performance recordings over different instruments under reproducibility conditions.
To test the feasibility and robustness of the new technique, a large pair-type comparison test has been carried out, with the purpose of evaluating the subjective perceptibility of differences among different instruments both with the new convolution technique and with the traditional technique of comparing acoustic recordings.
The three previously characterised violins and the viola have been used for these tests.
Two “anechoic” input samples (one of Paganini, the other of Bach), obtained as explained in paragraph 3.3, were convoluted with the measured impulse responses of the 4 different instruments. Each “virtual” violin (A,B,C) had to be compared with the remaining 2, so that 3 pairs of convoluted samples were obtained from each music piece (AB, AC and BC). On the other hand, 3 pairs of “microphonic pieces” were obtained coupling the direct acoustic recordings of each “real” violin with those of the other two.
Furthermore, two “control groups” each of 3 pairs of samples were mixed with the 6 “true comparisons” pairs: the first control group is made of 3 “truly equal” pairs (obtained playing twice the same sample) and 3 “truly different” pairs (obtained with convoluted pairs in which each violin is compared to the viola).

These 12 pairs were prepared for each of the two music samples obtaining 24 total pairs, with randomly shuffled presentation order: 9 subjects were asked to listen to the 24 pairs, filling up for each pair the following questionnaire:

Pair no. .... Are the two violins A and B the same? yes p no p

If Your response is no, explain why:

	a lot (-2)	slightly (-1)	no difference (0)	slightly (+1)	a lot (+2)	
A is better	p	p	p	p	p	B is better
A has more pronounced bass	p	p	p	p	p	B has more pronounced bass
A has more pronounced treble	p	p	p	p	p	B has more pronounced treble
A is softer	p	p	p	p	p	B is softer

4.1 Subjective results

The following table summarises the results of the first question (percentage of "equality"):

Convoluted samples	Microphonic samples	Truly equal samples	Truly different samples
14.6 %	12.5 %	75 %	0 %

These percentages show that convoluted simulations have actually almost the same degree of dissimilarity as microphonic recordings. But the truly equal and truly different samples are clearly recognised by the listeners.

Analysing the other 4 responses, the following three tables were obtained for the three violins studied averaging the subjective responses over all the pairs. They show the average value and the standard deviation of each response, as a function of the type of pair (convoluted, microphonic, etc.).

It can be observed that the values of convoluted samples pairs are very near to those obtained from microphonic pairs, as the differences are always lower than the standard deviations.

Langhoff Violin

	Conv. samples	Microph. samples	Equal samples	Different samples
better	-0.69 ± 0.83	-0.28 ± 1.01	0.0 ± 0.71	0.37 ± 1.41
pronounced bass	-0.28 ± 0.81	-0.31± 0.98	0.0 ± 0.35	–0.56 ± 1.22
pronounced treble	-0.37 ± 1.01	-0.05 ± 1.05	0.0 ± 0.50	1.44 ± 0.93
soft	-0.16 ± 1.18	-0.19 ± 1.11	0.06 ± 0.56	-0.75 ± 0.97

Klotz Violin

	Conv. samples	Microph. samples	Equal samples	Different samples
better	-0.16 ± 1.17	-0.03 ± 1.01	0.19 ± 0.39	0.75 ± 1.48
pronounced bass	0.34 ± 0.93	0.12 ± 1.25	-0.13 ± 0.33	-1.06 ± 0.97
pronounced treble	-0.19 ± 1.03	-0.09 ± 1.05	0.13 ± 0.48	1.25 ± 0.83
soft	0.03 ± 1.03	0.09 ± 1.06	-0.06 ± 0.43	-0.75 ± 1.20

Calcanius Violin

	Conv. samples	Microph. samples	Equal samples	Different samples
better	0.44 ± 1.05	0.72 ± 1.17	-0.13 ± 0.48	0.12 ± 1.05
pronounced bass	-0.03 ± 1.10	0.16 ± 1.05	-0.19 ± 0.53	-1.06 ± 0.97
pronounced treble	0.19 ± 0.93	0.47 ± 1.06	0.19 ± 0.53	1.25 ± 0.83
soft	0.16 ± 1.09	0.06 ± 1.17	-0.13 ± 0.60	-0.19 ± 1.18

On the other hand, the control groups show very different responses, that approach almost perfectly zero for the truly equal pairs, and exhibit extreme values for the truly different pairs.

This effect can be observed in a more evident way by looking at the graphs of fig. 6. The truly equal pairs always show a strong peak on the "0" (equality), while the truly different pairs show an evident trend toward an extreme of the scale. This is due to the fact that actually the viola is a very bad instrument compared to three violins (question n. 1), it has more pronounced bass (question 2), it does not have treble (question 3), and is certainly softer than the violins (question 4).

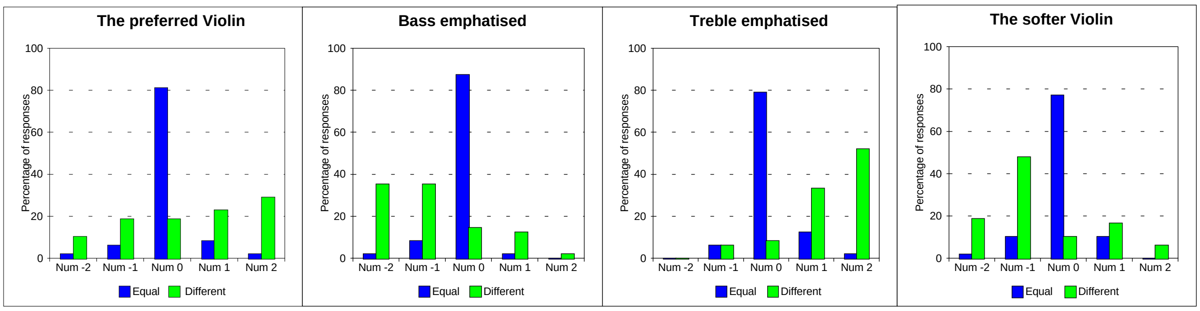


Fig. 6 - Distribution histograms for "truly equal" and "truly different" pairs

From the analysis of the control groups it can be concluded that the subjective test is reliable, and that the subjects were able to correctly identify acoustically evident differences or similarities.

Looking at the average results of the comparison of the three violins, as shown in fig. 7, it can be concluded that the Langhoff violin was judged the best of the three ones, the Klotz was the medium, and the Calcanius was the worse. Similar conclusions can be drawn for the other three questions. It has to be pointed out that the results obtained from the convoluted pairs are quite close to those obtained from the microphonic pairs: the absolute difference is always far lower than the standard deviation, and the sign is always maintained. This means that the same subjective ranking is obtained with the new proposed method, and thus it can be used in substitution of the traditional comparison technique, without any systematic bias of the subjective results

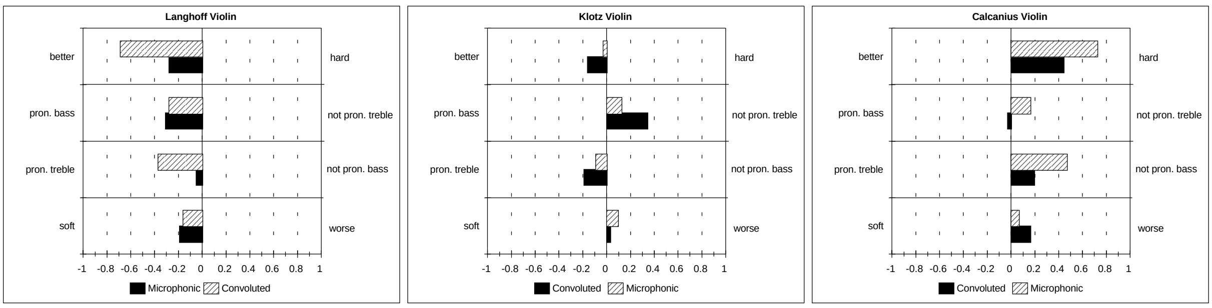


Fig. 7 - Averaged subjective results for the three violins

To obtain a deeper look at the individual responses, the graphs of fig. 8 and 9 compare the statistic distributions of the responses obtained with the new "virtual instrument" technique and with the traditional acoustic recordings of different violins. If the new technique is behaving correctly, the graphs of fig. 8 should be equal to those of fig. 9.

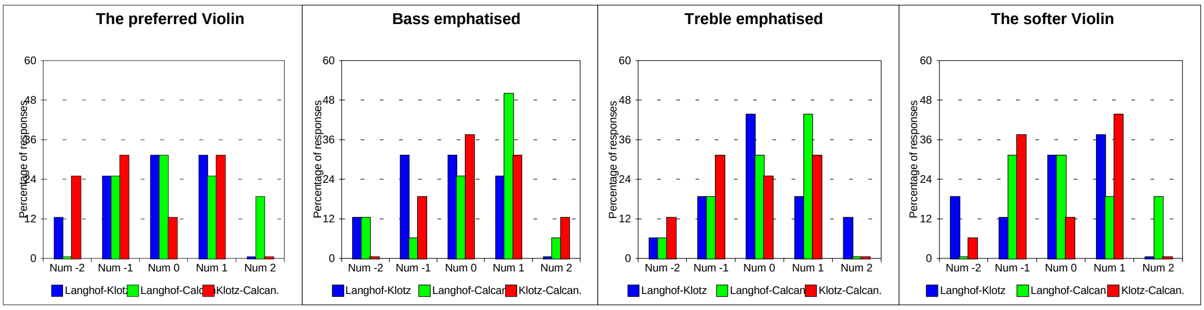


Fig. 8 - Distribution histograms of the four questions: microphonic pairs

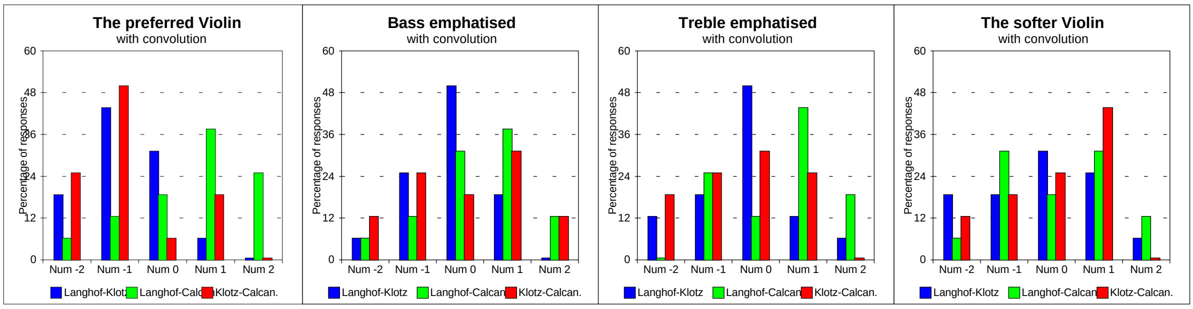


Fig. 9 - Distribution histograms of the four questions: convoluted pairs

4.2 Discussion of the results

Although a certain degree of similarity can be seen, there are still some differences between the graphs in fig. 8 and the corresponding in fig. 9, differences which however are not significant compared to the wide spread in the subjective responses. From the analysis of the answers it's shown that the spread is so large that actually no significant difference is found between the responses to the three violins, and this happens almost at the same degree both with convoluted pairs and with the microphonic pairs. The average differences found and shown in fig. 7 are in fact not significant if compared with standard statistical tests. This contrasts with the results of the control groups, and also with those of the first question, where it was clear that the subjects correctly identify as different these three instruments: they realise that the violin is changed, but then they are neither able to explain with precision what the difference is, nor are they capable of recognising which of the two instrument sounds better! Probably the test has to be repeated with a larger panel of trained, sharp-eared musicians, instead of a little group of students at the Engineering Faculty of the University of Bologna, as this was the case.

Anyway these results validate the novel technique proposed: the convoluted pairs give almost the same results as the direct microphone recordings, requiring a smaller effort, as a simple and fast measurement is taken on each violin, without the need of a musician performing various samples in an anechoic chamber. In this way, a large number of violins can be quickly compared.

The computation time required to convolute the "anechoic" input signals with the impulse response of each violin is also very short. So comparative tests can be conducted with many different music pieces, provided that suitable "anechoic" input signals are prepared as previously shown.

5. Conclusions

The implemented measuring techniques were able to accurately measure the mechanical-acoustical transfer function of different violins, represented by the time-domain impulse response. The MLS excitation system circumvented the previously encountered limitations, caused by limited time window length, poor noise rejection, limited sampling rate and long measurement time. Furthermore this technique doesn't need the use of large excitation amplitudes, that possibly cause non-linear distortions in the vibro-acoustic system.

The approximate inversion of these impulse responses made it possible to reconstruct the force input signal produced by the strings to the bridge during the playing time, starting from an anechoic acoustic recording. This "anechoic" input force signal can then be used as a starting point for producing music samples played on "virtual" violins, by the well known fast convolution process. These music samples can then be used for subjective comparisons among different violins, without the bias caused by the player's reaction to different instruments. By numerical manipulation of the impulse responses, the subjective effect of removing/adding spectral components or damping/increasing the reverberant character of an instrument can easily be assessed.

The results of the conducted subjective experiment show that the new convolution technique make it possible to create "virtual violins", that are digital filters capable of reconstructing almost perfectly the time and frequency response of a real instrument.

For these reasons it is expected that, employing the convolution technique, a better correlation between objective acoustical parameters and subjective evaluations will be found, even if the acoustic directivity of the instruments was not considered, as the sound field was sampled just in one point.

The prosecution of the research will be a large objective measurement campaign on dozens of instruments, followed by a subjective listening test with a selected panel of musicians and violin experts.

Furthermore the possibility of making numerical modifications to the measured impulse responses will be explored, with the goal of evaluating the virtual restoration of ancient, damaged instruments and the creation of new, non-existing instruments.

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