

## COMPARISON OF VIOLIN IMPULSE RESPONSES BY LISTENING TO CONVOLUTED SIGNALS

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### ABSTRACT

In this paper results obtained with a new measuring and playback technique are presented. The main assumption of the technique is that the body of a violin behaves as a linear mechanic-acoustic system, transforming input forces applied by the strings on the bridge into acoustic pressure waves irradiated through the air. Provided that the measured Impulse Response of the violin is sufficiently long as to reflect the complete decay of the instrument, properties of the violin timbre are fully contained in the transfer function (frequency domain) or in the Impulse Response (time domain).

As previously reported by the authors, the impulse responses or frequency responses of violins have been measured both with traditional techniques (2 channel FFT analysers) and with an MLS (Maximum Length Sequence) analyser.

In this study, these measured impulse responses were convoluted with an "anechoic" signal which was sampled during an actual musical performance, where a phonograph needle was mounted on top of the violin bridge and the acquired data then stored in digital format on a computer. This original signal was a velocity signal, and had to be converted into an equivalent force signal by inverse-filtering for the mobility of the excitation point of the particular violin, prior to applying the convolution process. This made the input signal largely independent from the particular violin used to record it. The convolution process can be used also to add the reverberation and colouring of a concert hall for direct headphone listening, otherwise the simulation is "acoustically anechoic" and is suitable only to be reproduced by loudspeakers in a good quality, reverberating listening room. A 16 bit sound card in the PC is then used to output the convoluted signal to loudspeakers or headphones, or the signal can be transferred digitally onto a DAT recorder.

In this paper the emphasis is put on a better understanding of the frequency response. An attempt is made to determine which resonance peaks are important to the sound impression of a violin. This investigation was conducted by editing the frequency response on the computer by parametric filtering, and then listening to the signal convoluted both with the original and modified. Subjective tests were conducted through direct comparisons. The results of these tests show that the Impulse Response technique clearly identifies the differences between violins of different makers, and that by adjusting the frequency response of a bad violin it is possible to make it sound better. However, this "virtual" playing on different violins is limited. For example, little is known about the dependence of the driving point mobility on its exact location and dynamic load (due to the bow). Therefore, a correct compensation as done here is difficult. Another drawback is the fact that one cannot take into

account the feedback loop that causes the musician to adapt to the peculiarities of a certain instrument to exploit its sound potential fully. This may be desirable since it confirms that the human adaptivity gives a musical performance its ultimate musical touch.

## 1. INTRODUCTION

The main assumptions made in this paper rely on fundamental concepts of linear system theory. It is therefore necessary to define a linear system model that most closely matches the ‘real world’ situation that it shall describe. In our case this ‘real world’ situation is given by a musician who is playing on a violin and producing sound. In a very simple model describing this scenario two blocks could be defined: The first block would represent the energy source of the system, e.g. the musician, and the second block would represent the violin body. Fig. 1 gives a sketch of this very simple linear system model.

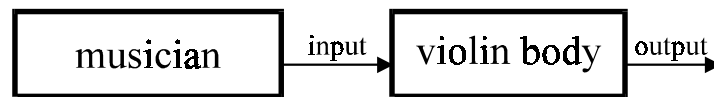


Fig. 1. A very simple linear system model describing the playing on a violin.

This model, in its simplicity, is already very useful and forms the basis for the comparison of violin impulse responses as described below. The difficulty with this model lies in the correct measurement of the input and output of the violin black box. For the case of the output of the violin this is quite feasible with a good microphone. More difficult to measure is the output of the musician or the input of the violin body. Two major problems arise here: First, what exactly is the intersection point between musician and violin, and second: Having defined a suitable intersection point, what sensor can be used to measure the signal at that point?

The violin itself is not a linear mechanic-acoustical transducer, but the violin’s non linear characteristics are due to the interaction between the string and the bow and to the vibration of the strings. The violin body alone, with the strings excluded, can however be considered linear as the plate displacements are extremely small. Any possible intersection point must therefore exclude the strings from the violin body black box in order to maintain the linear behaviour of it. This was done here where the top of the bridge was chosen as the intersection point between musician and violin body. The non-linear behaviour becomes in this way part of the (measured) input signal, while the linear behaviour of the violin body is characterised by its impulse response. In this way the most important timbric information of the violin body is contained in the impulse response, the little relevant information on the strings is not contained in it, and the again important information on the bridge itself is still included in the impulse response.

The purpose of this paper is to show that the linear system model described above can be realised and that hence the convolution algorithm can be applied. A more theoretical treatment can be found in [FAR 95.2]. At first, the measurement of the impulse response of the violin body and the measurement of the violin input signal is described. It transpired that the violin input signal could not be measured because the input point admittance was not known. Therefore an attempt was made to calculate the input signal from the measured output signal. This task, which involves the inversion of non minimum phase (mixed

phase) impulse responses turned out to be difficult. For the solution of this problem in this paper only standard methods available in the MATLAB Signal Processing Toolbox are reported.

Once the impulse response and the input signal is known, the convolution algorithm can be applied and the acoustic output signal can be calculated on the computer.

This output signal reflects then the timbral properties contained in the frequency response of the violin measured. This is a very important fact and can be used to compare different violin spectra by listening to the output signal obtained by the convolution of the impulse response of the violin body with the input signal. It is also possible to edit an existing frequency response in certain areas. Listening to the convoluted signal the timbral difference is due only to the editing. This is very appealing and could give way to many experiments in the psychoacoustic area.

## 2. IMPULSE RESPONSE MEASUREMENT

The violin impulse responses were obtained in two different ways, first by calculation of the inverse fourier transform of existing frequency response measurements contained in the database of the Cremona Violin making school and second, by measurements using a MLS (Maximum Length Sequence) signal. The MLS technique is based on a Maximum Length Sequence Pseudorandom Excitation Signal. The concept was developed by Schröder [SCH 65] for room acoustics applications. This excitation signal, although apparently similar to a random noise signal, has a well defined autocorrelation function. The system under interest is excited with this signal and the response signal is acquired with an A/D board on a computer. The acquired signal is then cross correlated to the original MLS signal and the impulse response is obtained directly in the time domain using the Alrutz fast convolution algorithm [RIF 89]. The MLS acquisition system used for the measurements described here consists of a Toshiba portable computer with an MLS generator and acquisition board in one of the expansion slots. Compared to traditional 2 channel FFT analysers the signal to noise ratio is much higher and the impulse response is measured with audio resolution. The MLS technique measures the impulse response in 16384 samples at 44.1kHz sample frequency, while the data obtained with a 2 channel FFT analyser has only 2048 samples at 16.384kHz sample frequency. The MLS system is usually used by one of the authors (A.Farina) for impulse response measurements in concert halls [COC 88], but by changing the excitation transducer this system can also be applied to violin body impulse response measurements.

### 2.1 The measurement set-up

The set-up for the measurement of the violin body impulse response ( or transfer function in the frequency domain ) used here is very similar to the set-up described in [LAN 94]. The exciter consists of an electromagnetic transducer based on a design first used by Dünwald [DÜN 84]. It consists of a small wire that is situated in a magnetic field. Any current passing through the wire will therefore exert a force on it that is proportional to the current. The wire touches the violin near the e-string as if it were a fifth string. Its excitation direction is parallel to the violin top plate, in correspondence with the string vibration direction. In Fig. 1a the measurement set-up and acquisition system is shown, fig. 1b. is a photo of the transducer as used in the physics lab. of the Cremona Violin Making School.

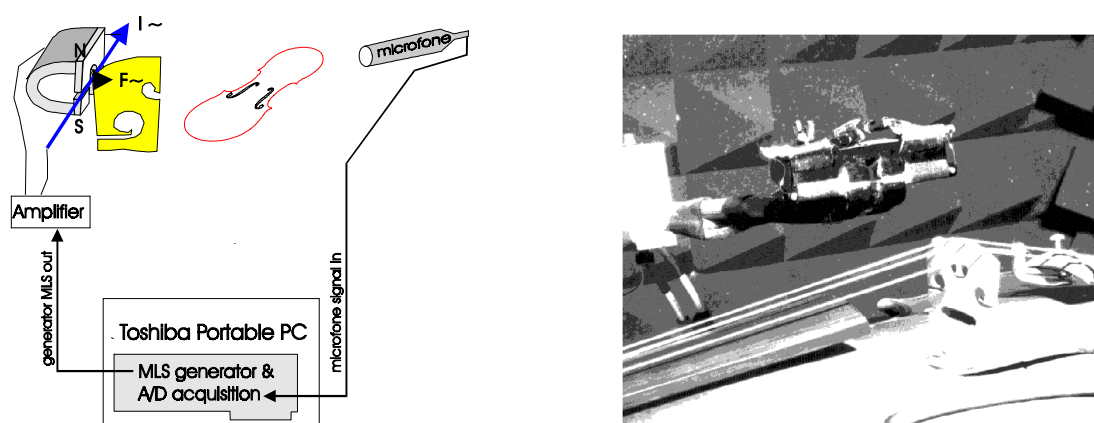


Fig. 2 (a) measurement set-up and (b) a photo of the transducer.

The sound pressure is registered with a B&K 1/2" 4176 microphone. The microphone was placed about 80cm distant from the violin, above it and inclined towards the e-string side. This microphone position was chosen because it corresponds to the direction that points directly to the audience during the musician's performance. All measurements were conducted in the semi-anechoic chamber of the Cremona Violin Making School, its dimensions are  $2 \cdot 2 \cdot 3 \text{ m}^3$ .

### 3. INPUT SIGNAL MEASUREMENT

In a first discussion with the musician it was decided what musical pieces to record. The aim was to pick pieces that make it easier to hear the strong or weak sides of a violin. Four pieces were chosen, and they cover most of the musical relevant situations. One piece was the theme of the 1.st concert of N. Paganini, played exclusively on the g-string. The next was the theme of the 2.nd concert of N.Paganini, this piece is played only on the e-string. A musical piece that exemplifies the balance of a violin and its capability to play chords was found in the Siciliana of the first sonata for violin solo by J.S.Bach. The cadenza of the D-major concert by W.A.Mozart was also chosen because it contains many scales covering the whole range of the violin.

For the registration the violin was instrumented with a velocity sensor that could measure the velocity signal on top of the bridge. The sound pressure was registered with a microphone at about 80cm distance in the direction towards the fictitious audience. The registration was carried out inside the semi-anechoic chamber of the Cremona Violin Making School. The data was registered on the 2 channels of a Casio DAT recorder. In parallel the data was registered directly onto the computer hard disk via a low cost 16 bit audio board. On the left channel the velocity signal on top of the bridge was recorded, on the right channel the microphone signal was recorded.

As already mentioned above, we are interested in the force signal that is exerted by the vibrating string on the violin bridge. The sensors available to the authors did not allow a direct measurement of the force. Therefore two approaches were taken to obtain this force signal.

### 3.1 Measurement of the velocity input signal

In this first approach the velocity signal at the bridge was measured and an attempt was made to deduce the force signal from it. A modified phonograph pickup was used to measure the velocity signal. The phonograph needle was mounted on top of the bridge in such a way that the needle was resting on the bridge. The needle was fixed with beeswax. A special holder provided for a soft mounting of the pickup; the whole holder was designed in such a way that the musician can still play normally on the violin with the holder mounted.

To obtain the force from the velocity the knowledge of the impedance of the bridge point is necessary. Once the impedance is known, one can calculate the force response of violin body with respect to its measured velocity signal. The force input signal can then be calculated by performing a convolution of the velocity signal with this force response. In this investigation the impedance of the bridge point was not measured; although measurements of the impedance are possible in principle and have been carried out in the past [BEL 74] [ALO 82].

Here we assumed that the bridge impedance was high relative to the string impedance for the frequency range of interest. If this was the case, the force and velocity signal differ only by a constant factor. A test of this assumption was carried out by listening to the measured velocity signal. If the bridge impedance was evenly high relative to the string impedance over the whole frequency range of interest, then its influence on the velocity signal should be negligible: This means that there should be no contribution from the violin body and bridge to the measured velocity signal; or in other words: listening to the velocity signal no timbral character from the violin should be present in the signal. `velocity\viandr6.wav` is an example of such a measured velocity signal. Listening to this sound sample it turns out that part of the timbral character of the violin is already contained in this signal. `velocity\calcani.wav` is the same musical example, but played on another violin. The timbral difference between the two samples is perceptible and an experienced listener can clearly distinguish between the two different violins used.

This indicates that the violin body impedance cannot be considered constant. Hence the force and velocity signal are not only different by a constant factor.

### 3.2 Velocity signal as convolution input

In order to find out how great the error is by using the velocity signal instead of the force input signal a simple test was performed: The velocity signal was convoluted with the impulse response; this signal is given in the sound example `convolve\velocity\vian-dr6\viandr6.wav`. Parallel to the velocity signal on the second channel of the DAT recorder the microphone signal was measured. This signal is given in sound example `micro\viandr6.wav`. The measured sample `micro\viandr6.wav` has its balance shifted noticeable towards the lower frequencies.

The velocity signal can though still be used in the convolution process to obtain the acoustic output signal. `Convolve\velocity\calcani\poggil.wav` is the result of the convolution of `velocity\calcani.wav` with `poggil.h_1`, `convolve\velocity\calcani\ianuar2.wav` is the result of the convolution of `velocity\calcani.wav` with `ianuar2.h_3`. Since both output signals are obtained using the same input signal `velocity\calcani.wav` in this case, the timbral difference between the two samples is only related to the difference in the impulse response (or frequency response) of the two violins used. Differences are clearly perceptible, and one can detect a difference timbral character either when the musician is playing on the g-string (at low frequencies) or when the musician is playing on the e-string.

The two violin impulse responses were obtained from violin body frequency response measurements conducted previously [LAN 93]. Thus the convolution concept enables one to re-listen to an instrument that is unavailable now. The output signal obtained in this way does not however correspond to the real sound impression. A correction of the signal for the influence of the impedance cannot at this stage be done; but it is still possible to detect relative changes between different instruments.

### 3.3 Calculation of the Force input signal

Another approach to obtain the force input signal is to derive it from the (measured) acoustic output signal. Following the assumptions made earlier the force input signal can theoretically be calculated from the output signal by a deconvolution with the violin body impulse response. The deconvolution process on the computer is however sensitive to noise and the zeros in the data have to be treated separately [see for example Numerical Recipes, Cambridge University Press]. For the measured data the direct deconvolution could not be used.

Instead one can try to find the inverse impulse response to the violin body impulse response. The data set is then confined to only the impulse response (about 16000 points ~ 64Kb) and it is not necessary to consider the output data (several Mb); this makes data fitting and elaboration much easier on the computer.

**The inverse impulse response:** For the impulse response `viandr6.tim` it was tried to find the inverse impulse response. The inverse impulse response is not uniquely defined for one impulse response since all algorithms tried here take into account only the magnitude of the impulse response to calculate the inverse. The phase can be chosen arbitrarily, but most algorithms adjust the phase such that the real and the imaginary part of the impulse response are related to each other via a Hilbert transform. Filters that fulfil this requirement are called minimal phase filters.

Several algorithms were tried out to find the best inverse impulse response, most are taken from the Signal Processing Package of MATLAB.

Three different algorithms were tried out to determine the inverse impulse response. One was a Finite impulse response filter design (MATLAB: `fir2`), the second was a least squares linear phase FIR filter design (MATLAB: `firls`), and the third was a recursive IIR filter design using a least squares fit in the time domain (MATLAB: `yulewalk`).

In all cases it was necessary to use very high filter orders in order to describe the magnitude response of the violin body correctly. For the FIR filters orders of 200 to 2000 were used, the IIR filter could only work up to order 60, above this it became unstable. The quality of the inverse filter was checked in the following way: First, the microphone sound signal `micro\viandr6.wav` was convolved with the inverse impulse response, the result is the force input signal. This force input signal was then convolved with the original impulse response, the result then is the calculated microphone signal:

$$\langle \text{microfone signal} \rangle_{\text{meas.}} * \langle \text{impulse response} \rangle_{\text{calc.}}^{-1} * \langle \text{impulse response} \rangle_{\text{meas.}} = \langle \text{microfone signal} \rangle_{\text{calc.}}$$

The sound samples `calcmic\fir1kmf3.wav`, `calcmic\firlsmf2.wav` and `calcmic\yu60b.wav` are all calculated using the procedure described above. All examples use the measured microphone signal `micro\viandr6.wav`, and for all examples the measured impulse response `viandr6.tim` was used. For example `calcmic\fir1kmf3.wav` the inverse impulse response was calculated using the `fir2` MATLAB function, the filter order was 1000. For example `calcmic\firlsmf2.wav` the inverse impulse response was calculated using the `firls` MATLAB function, the filter order was also 1000. For example

`calcmicr\yu60b.wav` the inverse impulse response was calculated using the *yulewalk* MATLAB function, the filter order was 60. All these functions require the magnitude spectrum of the desired filter as parameter. This magnitude spectrum, which results from inverting the magnitude spectrum of the frequency response of the violin body, was artificially restricted to the range 100Hz-6kHz by multiplying it with a smooth window. This was necessary to minimise influence of out of band effects during the filter coefficient determination. The calculated microphone signals are not identical to the measured signal, with artificial reverberation effects being particularly noticeable in these signals.

#### 4. CONVOLUTION AND SYNTHESISED SOUND OUTPUT

For the convolution the software AURORA, developed by A. Farina [FAR 93] [FAR 95] was used that does fast convolution on standard PC hardware (Dos, Windows, Windows NT). The software stores both input and output files in standard Multimedia WAV format. In conjunction with a low cost 16bit sound card (SoundBlaster 16, Turtle Beach MultiSound, etc.) the WAV files can be listened to with speakers or earphones; once the convolution is performed, comparative tests can be easily conducted with just a “point and click” delay. Convolution is performed through the well known “select save” algorithm and the convolution time is about 1.2 to 4 times the data length on a PENTIUM 60 CPU depending on the impulse response length.

#### 5. EDITING THE FREQUENCY RESPONSE CURVE

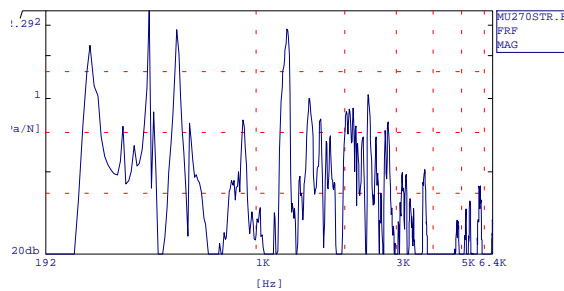
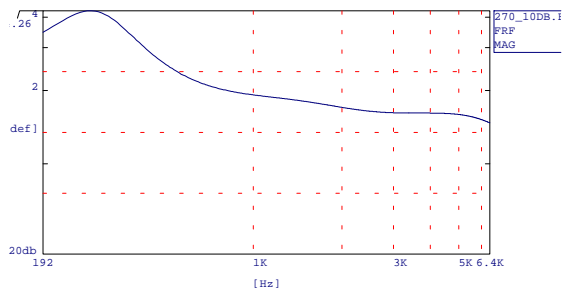
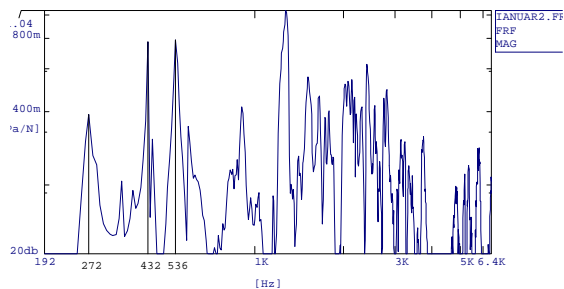
One acoustic spectrum was chosen as reference spectrum and then specific areas in the spectrum were changed. An input signal was then convoluted with the modified spectrum and the output signal listened to. As a starting point only three simple modifications were applied to the spectrum: One modification was an enhancement of the Helmholtz resonance peak, the second change introduces a smooth decay towards higher frequencies, and the third modification enhances the mid-range (or nasal) frequency region around 1.7kHz. The reference spectrum `ianuar2.frf` and the weighting functions (`multiply\270_10db.frf`, `multiply\10dbdecay.frf`, `multiply\1k7.frf`) together with the resulting modified violin spectra (`mu270str.frf`, `mu10dbdy.frf` and `mul1k7.frf`) are shown in fig.3. The sound samples `piece\ianuar2.wav`, `piece\mul270str.wav` and `piece\10dbdecay.wav` represent the output of the convolution of the velocity signal with the relative violin spectrum. *Piece* denotes the relative musical piece used, possible values are *mozart*, *sol-pag* and *mi-pag*.

Remark: Before going on with a brief description of the sound differences one should remember that the following description is only a subjective impression by one of the authors (A. Langhoff) that does not claim to be scientifically provable. It is the authors opinion that any attempt to find ‘true’ criteria for the ‘quality’ of a violin sound will fail. This is because everyone has a (at least slightly) different ideal of sound. However, owing to the convolution method the subjective impression can be much better presented to others.

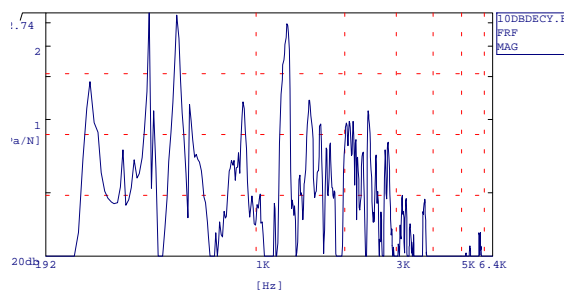
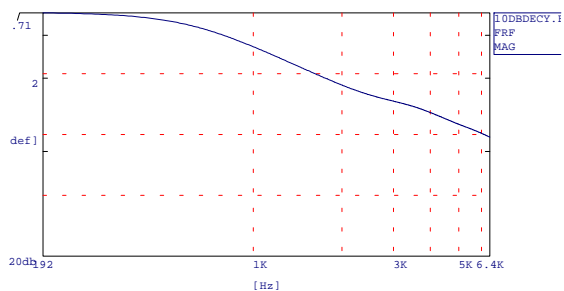
The two modifications `10dbdecay.frf` and `mul270str.frf` have similar effects, when compared to the original spectrum `ianuar2.frf`. Two musical pieces were used, one played on the g-string, the other played on the e-string. At first, the piece *sol-pag* was used. Both modifications enhance as expected the bass volume in the output. While the sample `sol-pag\mul270str.wav` gives mainly the impression of more ‘filling’ volume, on the `sol-pag\10dbdecay.wav` another effect is also present: Here one would like to ascribe the attribute ‘closed’ to the sound impression. Changes are also noticeable when playing on the e-string. The sample `mi-pag\10dbdecay.wav` sounds clearly ‘darker’ than the original sample

(a) ianuar2.frf

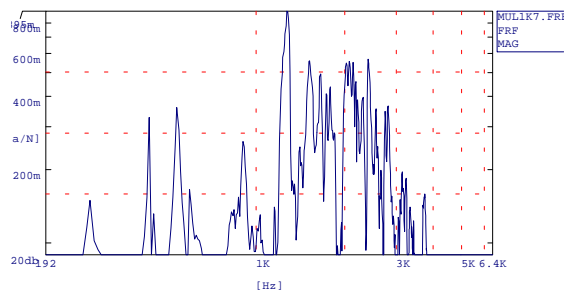
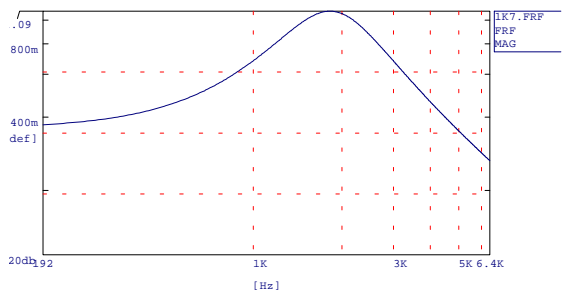
This frequency response is used as a basis for the editing process.



(b) 270\_10db.frf. This weighting function was used to enhance the Helmholtz resonance in the spectrum



(c) 10dbdecy.frf. This weighting function introduces a linear decay towards higher frequencies



(d) 1k7.frf. This weighting function enhances the nasal region in the spectrum of the violin

Fig. 3. Different spectra derived from ianua2.frf that were compared by listening to convoluted signals. Three different envelopes were used to modify the frequency response curve. Each of the three envelopes is multiplied with the original spectrum ianuar2.frf to yield the modified spectrum. 270\_10db.frf enhances the Helmholtz resonance peak of the violin by about 10db; 10dbdecy.frf imposes a decay toward higher frequencies and 1k7.frf enhances the nasal region of the violin about 1.7kHz.



mi-pag\ianuar2.wav. It is interesting to note that a modification mainly in the Helmholtz resonance area is clearly perceptible also when playing on the e-string.

Another test was conducted comparing the modification 1k7.frf (enhancement of the nasal area) to the reference. Here the beginning of the cadenza of the Mozart g-major concert was used as musical piece. One can clearly notice the more ‘tinny’ sounding version of the cadenza mozart\1k7.wav. Finally, the reference sample mozart\ianuar-2.wav is given, where the basses seem to be more balanced than in the modified version.

## 6. CONCLUSION

It has been shown that it is possible to compare violin spectra by listening to convoluted signals. The method of editing the frequency response curve and re-listening to the convoluted signal is a very powerful method to get a better understanding of the violin’s spectrum. This paper demonstrates the applicability of this method with some simple examples, a more thorough work confined to parametric editing of the violin spectrum would be a further step and will hopefully follow in the future.

Many problems still remain unsolved. In particular, the measurement and calculation of the force signal at the violin body bridge must be improved. A lightweight measurement device to measure the force directly would be an elegant solution. A related unresolved problem, that is encountered when one tries to calculate the force signal from the microphone output signal, is to find the inverse impulse response of a non minimal phase filter. Due to these limits a completely violin independent input signal is not available. This means that the sound output does always contain a component of the violin with which the measurements were conducted, therefore it does not yet represent the violin’s real sound correctly. For relative comparisons between different spectra the input signal is well suited however. It remains an open question whether it is possible to measure a completely violin-independent input signal, perhaps this ‘clean’ signal would not actually sound like a violin signal any more.

The measurement quality is already quite good, especially considering that the spectra presented here were mostly sampled at 16kHz. The old data, which was used in the editing section, is surely limited at high frequencies. By using the new MLS technique this limitation can be eliminated.

### LIST OF SOUND SAMPLES:

All sound examples were played by Maestro Marco Fornaciari from Padova, 1.violin of the “Solisti Veneti”

- 1 velocity signal measured on the bridge top with a phonograph needle, W.A. Mozart D-major concert cadenza, played on a Calcanius violin (WAV file: velocity\calcani.wav).
- 2 velocity signal measured on the bridge top with a phonograph needle, W.A. Mozart D-major concert cadenza, played on a modern violin (WAV file: velocity\viandr6.wav).
- 3 microphone signal measured for 2 (WAV file: micro\viandr6.wav).
- 4 inverse filter test (fir2): The sample 3 was first convoluted with the inverse filter for the impulse response of viandr6.tim, then it was convoluted with the original impulse response. Theoretically the outcome should be identical to the input (3). The MATLAB fir2(1000,f,minv) function was used. f denotes the frequencies array, minv denotes the inverse magnitude array of the original impulse response. (WAV file:calcmicr\fir1kmf3.wav).
- 5 inverse filter test (firls): The sample 3 was first convoluted with the inverse filter for the impulse response of viandr6.frf, then it was convoluted with the original impulse response. Theoretically the outcome should be identical to the input (3). The MATLAB firls(1000,f,minv) function was used. f denotes the frequencies array, minv denotes the inverse magnitude array of the original impulse response. (WAV file:calcmicr\firlsmf2.wav).

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- 6 inverse filter test (yulewalk): The sample 3 was first convoluted with the inverse filter for the impulse response of viandr6.frf, then it was convoluted with the original impulse response. Theoretically the outcome should be identical to the input (3). The MATLAB yulewalk(60,f,minv) function was used. f denotes the frequencies array, minv denotes the inverse magnitude array of the original impulse response. (WAVfile:calcmicr\yu60b.wav).
- 7 Convoluted sound output. As input signal the velocity signal 2 was used. The violin impulse response belongs to a Klotz school violin (impulse response: klotz.tim) (WAV file: convolve\velocity\calcani\klotz.wav).
- 8 Convoluted sound output. As input signal the velocity signal 2 was used. The violin impulse response belongs to the modern violin (impulse response: viandr6.tim) (WAVfile: convolve\velocity\calcani\viandr6.wav).
- 9 Convoluted sound output. As input signal the velocity signal 2 was used. The violin impulse response belongs to a N. Amati workshop violin (spectrum file: ianuar2.h\_3)(WAVfile: convolve\velocity\calcani\ianuar2.wav).
- 10 Convoluted sound output. As input signal the velocity signal 2 was used. The violin impulse response belongs to a A. Poggi violin (spectrum file: poggi1.h\_1) (WAV file: convolve\velocity\calcani\poggi1.wav).
- 11 Convoluted sound output: input signal is velocity signal, impulse response calculated from the spectrum ianuar2.h\_3 (Amati workshop violin), piece is N. Paganini 1. concert theme (on g-string) (WAV file: ianuar2\sol-pag\ianuar2.wav).
- 12 Convoluted sound output: similar to 11, but a 10db Helmholtz resonance region boost was introduced (weighting function: mul270str.frf) (WAV file: ianuar2\sol-pag\mul270str.wav).
- 13 Convoluted sound output: similar to 11, but a 10db decay towards high frequencies was introduced (weighting function: 10dbdecay.frf) (WAV file: ianuar2\sol-pag\10dbdecay.wav).
- 14 Convoluted sound output: input signal is velocity signal, impulse response calculated from the spectrum ianuar2.h\_3 (Amati workshop violin), piece is N. Paganini 2. concert theme (on e-string)(WAVfile:ianuar2\mi-pag\ianuar2.wav).
- 15 Convoluted sound output: similar to 14, but a 10db decay towards high frequencies was introduced (weighting function: 10dbdecay.frf) (WAV file: ianuar2\mi-pag\10dbdecay.wav).
- 16 Convoluted sound output: input signal is velocity signal, impulse response calculated from the spectrum ianuar2.h\_3 (Amati workshop violin), piece is W.A. Mozart D-major concert cadenza (WAV file: ianuar2\mozart\ianuar2.wav).
- 17 Convoluted sound output: similar to 16, but a medium frequency enhancement was introduced (weighting function: 1k7.frf) (WAV file: ianuar2\mozart\10dbdecay.wav).

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