LISTENER EVALUATION OF REAL AND SYNTHESISED IMPULSE RESPONSE DESIRABILITY USING CONVOLUTION WITH THE VIOLIN STRING SIGNAL

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1 INTRODUCTION

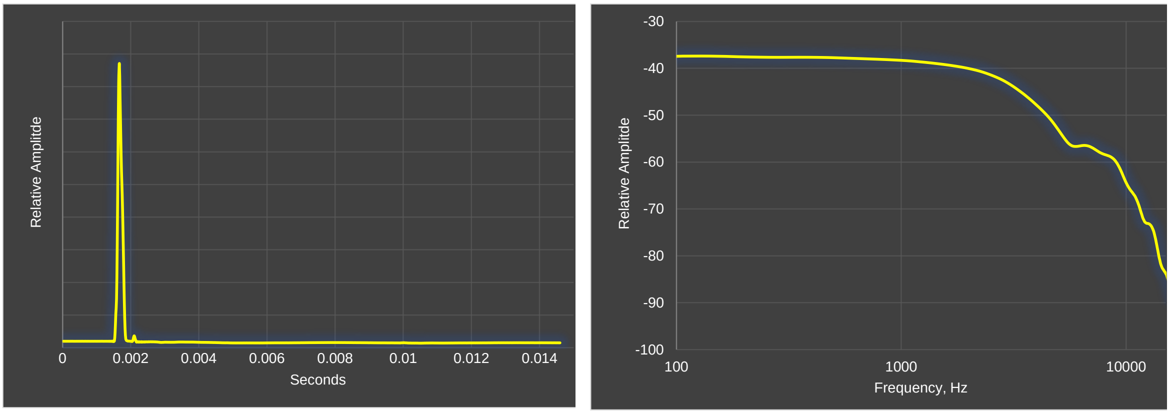
The timbres of both plucked and bowed acoustic string instruments are shaped by the vibrations of their hollow bodies [1-3]. These vibrations are initiated by the strings, which are either plucked or bowed, involving complex, non-linear frictional forces between the strings and the musician's fingers or bow hair [4, 5]. In an ideal case, a taut string oscillating between two fixed points would produce a fundamental tone, along with harmonics at frequencies that are integer multiples of the fundamental [6]. However, in real-world instruments like the guitar, this ideal is not fully realized due to inharmonicity arising from non-linear interactions, resulting in harmonics that are slightly offset from their expected positions. The string itself contributes minimally to the sound we hear, as it displaces only a small amount of air. Instead, the vibrational energy is transferred to the instrument's body, which, due to its physical properties, plays a crucial role in determining the tonal quality perceived by the listener. Skilled musicians can expertly exploit these acoustic properties, which is why the unique timbres, spatial projection, and interaction between string motion and the instrument body have been extensively studied over the years. Researchers have employed various methods to characterize the acoustic properties of these instruments by measuring their impulse responses and identifying the features that most significantly influence tonal quality. In contrast, the raw sound of an electric guitar or violin is primarily the result of electronically amplified string vibrations, as the bodies of these instruments are generally solid frames. This paper explores three key aspects of this area of research:

- A typical methodology for instrument body characterisation through response measurement.
- A software system for the synthesis of body impulse responses that utilizes cascade parametric filters
- Hardware for real-time convolution of the body impulse responses, for use by players of both acoustic and electric strong instruments.

We have reported previously on the nascent development of a system for generating synthetic impulse responses (IRs) of various string instruments, based on the dominant resonant modes of the body. Typically for a violin these will be A0 (265 Hz), C2 (395 Hz), A1 (430 Hz), C3 (550 Hz) and C4 (665 Hz). The software to synthesize IRs uses a cascade series of parametric filters, expressed as infinite impulse response (IIR) biquad equations. The user may adjust the centre frequency, gain and bandwidth of each mode through a set of controls within the graphical environment of the software. The system instantly generates both the frequency and impulse response as the user adjusts the various parameters; IRs are stored in a standard file format (16-bit WAV with user-selectable sample rates of between 8 kHz and 48 kHz). Listening tests have confirmed that while these IRs do not yet approach the quality of measured equivalents, there is sufficient evidence to suggest that with further refinement of the software, the timbre and tonality of synthesised IR will significantly improve.

2 MEASURING THE IMPULSE RESPONSE

An acoustic instrument can be closely approximated as a linear system that consistently acts as a filter, regardless of the type or source of excitation. Since the system is linear, it can theoretically be characterized by its impulse response, the Fourier transform of which provides the frequency response. Various methods are used to obtain the impulse response, with a helpful summary provided in [1]. One approach used by the authors involves tapping the violin bridge with a small hammer equipped with an accelerometer, while recording both the force signal and the resulting sound with a microphone placed in the far field [7-10]. Although a single recording does not account for directivity, it does capture the low-frequency Helmholtz resonance (breathing mode) through the instrument's sound holes (such as the circular hole in a guitar or the f-holes in bowed instruments), which is largely isotropic. These measurements must be conducted in an anechoic environment to minimize the effects of unwanted room reverberations. Additionally, it is necessary to correct for the limitations of the hammer strike; ideally, if the hammer delivered a perfect impulse, the instrument would produce a perfect impulse response, encompassing all frequencies equally. However, in practice, the impact is not ideal, displaying a profile of rapid attack followed by a more gradual decay, as shown in Figure 1. This figure (a) illustrates the force signal from an actual hammer strike on a violin bridge, while (b) shows the strike's spectrum, which diminishes in amplitude at higher frequencies. Consequently, the instrument's response is less energetic in this upper range, necessitating compensation through deconvolution.



(a) (b)

Figure 1. Hammer strike and its spectrum.

This is achieved as follows. The Fourier transforms of the recorded response $y(t)$ and the hammer strike $x(t)$ are computed, yielding $Y(\omega)$ and $X(\omega)$ respectively. The spectrum $X(\omega)$ is inverted, using a guard operator s, i.e.

$$\tilde{X}^{-1}(\omega) = \frac{1}{X(\omega) + s} \quad (1)$$

The use of a guard operator obviates the risk of an ill-conditioned division, and furthermore, by controlling the value of s, the degree of compensation may be controlled explicitly. The estimated, true impulse response of the violin may now be obtained by calculating the product of the Fourier vectors $Y(\omega)$ and $\tilde{X}^{-1}(\omega)$ and taking the inverse Fourier transform thus:

$$\tilde{h}(t) = F^{-1}[Y(\omega)\tilde{X}^{-1}(\omega)] \quad (2)$$

Other more complex schemes exist based on adaptive modification of s, but the method described above yields results that are repeatable and convincing in the context of performance and listening.

3 SYNTHESIS OF BODY IMPULSE RESPONSES

The goal of response synthesis in the context of performance differs significantly from that of traditional characterization or modal analysis. In performance, the aim is to create a response that, when convolved with the string signal of an electric instrument, results in a sound with desirable tonal qualities, meaning it is pleasing to the ear. On the other hand, traditional response derivation, whether through the measurement methods mentioned earlier or through modal analysis, seeks to accurately estimate the acoustic properties of the instrument's body. A more straightforward approach involves approximating the general behaviour of an instrument body by using a response generated from a tuneable parametric equalizer, which is then applied in the convolution process. In previous blind listening tests we have conducted, we often found that violin music created by convolving a raw electric signal with a response measured from a violin is frequently indistinguishable from music played on the original instrument. This is not yet the case with synthesized responses, mainly because it is very difficult to replicate the complexity of the response – especially with the higher frequencies, where modal overlap takes place.

3.1 The complexity of the measured impulse response

If we consider the violin body as an example, it is evident that each frequency response will vary according to instrument quality and details regarding its construction. What is less certain is the quantitative relationship between the tonal preference and representation of its spectral characteristics. Nevertheless, it is generally agreed that the most important resonant modes of a violin are given as C1 (~185 Hz), A0, usually termed the Helmholtz resonance (260-290 Hz), C2 (~405 Hz), A1 (430-490 Hz), C3 (490-590 Hz) and C4 (~700 Hz). In addition, there is a "bridge hill" residing between 2-3 kHz. In reality, due to modal overlap, the frequency response is much more intricate than these discrete modes suggest; Figure 2, for example, shows a typical response measured from a Stradivarius violin.

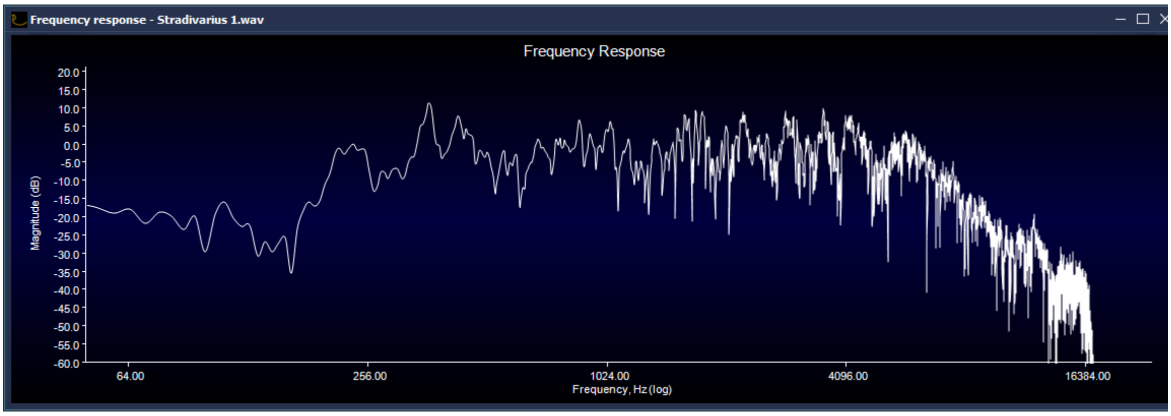


Figure 2. Frequency response of the body of a Stradivarius violin, measured using the impact hammer and microphone method.

Clearly any representation of such a frequency response using parametric modelling would be a gross simplification; however, it is not possible to know, without conducting listening tests, which of the overlapping modes contribute in any significant manner to a desirable timbre – which is why a readily tuneable parametric tool would be of benefit to both instrument makers and players.

3.2 *Lucy Boo*: software for impulse response synthesis using a parametric approach

The parametric equalizer consists of several IIR filters, each of which allows independent adjustment of centre frequencies, gains, and Q factors. By default, each filter unit is set to a gain of 1, or 0 dB, which is why the filters are configured in a cascading arrangement.

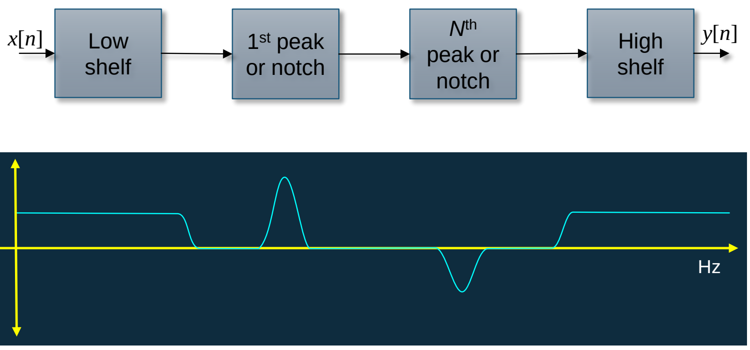


Figure 3. Algorithmic structure of a software parametric equalizer, comprising *n* tuned biquad filters in series.

A parametric equalizer will typically comprise a low frequency shelving filter, several peak/notch filters (typically between four and seven) and a high frequency shelving filter, Figure 3. The basic structure of each filter block is a biquad with a transfer function of

$$H(z) = \frac{a_o + a_1 z^{-1} + a_2 z^{-2}}{1 + b_1 z^{-1} + b_2 z^{-2}} \tag{5}$$

And a difference equation given by

$$y[n] = a_0 x[n] + a_1 x[n-1] + a_2 x[n-1] - b_1 y[n-1] - b_2 y[n-2] \tag{6}$$

In which

$$\begin{gathered} a_0 = 1 + \alpha A \quad a = -2\,cos(2\pi f_c/f_s) \quad a_2 = 1 - \alpha A \\ b_0 = 1 + \frac{\alpha}{A} \quad b_1 = -2\,cos(2\pi f_c/f_s) \quad b_2 = 1 - \frac{\alpha}{A} \\ A = 10^{g/40} \qquad \alpha = \frac{sin(2\pi f_c/f_s)}{2Q} \end{gathered} \tag{7}$$

Where f_c is the centre frequency and f_s is the sampling frequency. Since each filter stage acts independently, it may be used to replicate the behaviour of the dominant frequency modes of an acoustic instrument body. By stimulating the resulting cascaded difference equations with an impulse, we obtain the estimated impulse response and associated frequency response.

Figure 4 depicts the software tool *Lucy Boo* which performs very simple synthesis of impulse/frequency responses of acoustic instrument bodies. The control panel on the left allows the user to control the centre frequency, gain and selectivity of a low-shelf filter, six peak/notch filters and a high-shelf filter. The resulting frequency and impulse responses are shown in the lower two plots. In addition, the software supports audio track playback – for example, an electric violin recording – shown here in the upper plot. When the track is played, the changes in timbre take place in real time as the filter controls are adjusted. Once the user is satisfied with the tone, the impulse response can be exported and used with the hardware pedal (for live performance) described below.

Figure 4. The *Lucy Boo* software.

4 HARDWARE FOR LIVE PERFORMANCE

As mentioned earlier, the impulse response is crucial for instrument makers and musical acoustics researchers because it defines the instrument's timbre or tonality. However, it is important to note that the impulse response varies with location; for instance, the impulse response captured from the front of a guitar or violin differs from that captured from the sides. Additionally, the sound experienced by the performer can differ significantly from that perceived by the listener. Despite these nuances, the impulse response is valuable not only for analysis but also for synthesis, which is the focus here. A device called *Prosody*, commonly referred to as a pedal, has been developed to process the raw electrical output from an electric string instrument, such as a guitar or violin, to generate an output signal that, when amplified and played through a speaker, closely mimics the timbre of the corresponding acoustic instrument, as shown in Figure 5 [7]. This system includes a high-impedance preamplifier, a 24-bit sigma-delta codec, and a high-speed digital signal processor (DSP). It can store up to sixteen far-field impulse responses of wooden acoustic instruments, allowing any of these responses to be selected and convolved in real-time with the input signal to produce the modified output. The initial stage of convolution is implemented with near-zero latency using a finite impulse response (FIR) structure. Additionally, the system features various enhancement effects, with a precision spectrum editor being the most notable. The pedal is widely used by professional musicians around the world and comes with a diverse set of impulse responses from guitars, violins, violas, cellos, and upright basses.

Although the IRs supplied with Prosody are measured from real instruments, the impulse responses synthesised by *Lucy Boo* are also completely compatible with the *Prosody* pedal. We now plan to make this software available to players and gather opinions on the merits of the IRs it can produce. It may be that whilst it might not – yet - accurately reflect the timbral properties of actual wooden instruments, it nevertheless allows tonal modifications that players of electric instruments find interesting and desirable.

Figure 5. The Prosody pedal, which performs real-time convolution of an acoustic instrument impulse response with a signal from its electric equivalent.

5 CONCLUSION

In the realm of musical instrument bodies, frequency and impulse response play a crucial role in linking the instrument's tonal characteristics with its physical acoustics, which are influenced by its design, materials, and construction. Moreover, impulse responses are now employed with real-time convolution engines to improve the timbre of electric instruments. Here, the goal is less about understanding the physical principles and more about achieving a pleasing sound quality. Consequently, current software developments are eliminating the need for traditional impulse response measurements by enabling the creation of synthetic responses that can be tailored to the performer's artistic preferences.

6 REFERENCES

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