Sculpting the behaviour of the Feedback-Actuated Augmented Bass

Design strategies for subtle manipulations of string feedback using simple adaptive algorithms

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ABSTRACT

This paper describes physical and digital design strategies for the *Feedback-Actuated Augmented Bass* – a self-contained feedback double bass with embedded DSP capabilities. A primary goal of the research project is to create an instrument that responds well to the use of extended playing techniques and can manifest complex harmonic spectra while retaining the feel and sonic fingerprint of an acoustic double bass. While the physical configuration of the instrument builds on similar feedback string instruments being developed in recent years, this project focuses on modifying the feedback behaviour through low-level audio feature extractions coupled to computationally lightweight filtering and amplitude management algorithms. We discuss these adaptive and time-variant processing strategies and how we apply them in sculpting the system's dynamic and complex behaviour to our liking.

Author Keywords

NIME, augmented string instrument, electromechanical actuation, electroacoustic, string feedback, experimental lutherie

CCS Concepts

•**Human-centered computing** → **Sound-based input / output;** •**Information systems** → *Music retrieval;* •**Applied computing** → *Performing arts;*

1. INTRODUCTION

The project builds on a method for feedback string instruments developed in *halldorophones* [21], a cello-like, electroacoustic, feedback string instrument which has proven to be of interest to experimentally minded string players in recent years [8] [5].

In this project, a double bass is modified in that same tradition: individual pickups located under each string are routed through an amplifier to an embedded speaker at the back of the instrument, in our implementation with the addition of a Bela microprocessor[1] [10] running SuperCollider[2] between pickups and amplifier. This results in a feedback 'loop', as the electro-mechanically amplified string vibrations re-vibrate the entire instrument in conjunction with any excitation introduced by the player (plucking, bowing, hitting, etc.).

Figure 1 provides a diagram of the instrument system configuration, with a conceptualised visualisation of the acoustic and digital signal flow.

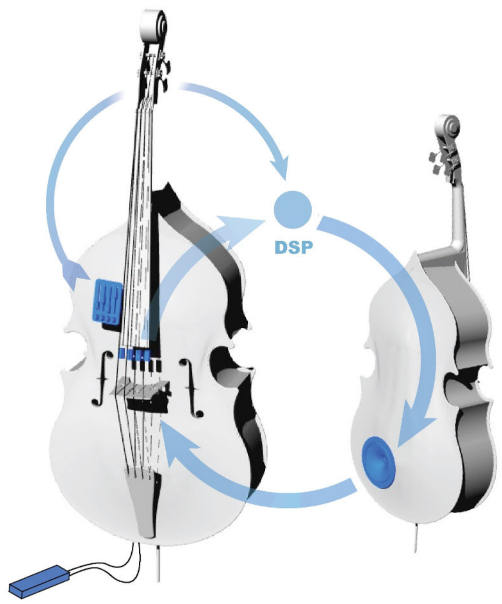

Figure 1: A diagram of the Augmented Double Bass. Component configuration and feedback pathway.

The design objective of this project is to create a robust, portable and self-contained feedback string instrument that compliments the basic electro-mechanical dynamics of the feedback system through digital signal processing, all while retaining the sonic fingerprint of the double bass as an aesthetic constraint. By extending the behaviour of a dou-

[1] `https://shop.bela.io/collections/bela-and-bela-mini/products/bela-starter-kit` (Accessed 30.04.2020)
[2] `https://supercollider.github.io/` (Accessed 30.04.2020)

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ble bass into electroacoustic and digital space, and employing a cybernetic approach to the ergodynamic[3] [9] design strategy, we cause the instrument-system to exhibit semi-autonomous and emergent behaviour. This calls for an explorative mindset on behalf of the player rather than a reliance on conceptions of technical and interpretative mastery. Currently seven months into the project, this paper reports on evaluation of the work with and the experience of negotiating our design intent within the constraints of the current system.

2. PREVIOUS WORK

The ergodynamics of feedback string instruments are fundamentally different to those of their acoustic and electroacoustic siblings in that the former are set up to introduce mechanical energy (vibration) into the instrument in a coupled relationship to the excitation of the strings. The points of coupling in the feedback loop (detecting and actuating) can be configured in different ways, affecting the feel and behaviour of the system. However, the central principle of electro-mechanically co-energising an acoustic resonating body is a defining quality of the feedback instruments described here. The halldorophone-method [21] and derived augmented instrument projects [5] [8] are organised to feed back through an actuator embedded in the body of the instrument, with individual pickups for each string allowing for the mixing of string-prominence in the resulting feedback. A different approach is Melbye's *What the Frog's Eye tells the Frog's Brain* [11] where the dominant coupling is through the body of the double bass, induced through the application of contact pickups and actuators to the soundbox. As a consequence, Melbye's system favours the resonant frequencies of the instrument body rather than the strings.

3. THE INSTRUMENT

Central to the design of the Feedback-Actuated Augmented Bass (hereafter the FAAB) is a conception of simplicity; a wish to keep all electronics embedded in the instrument, allowing for a self-contained portable system where the only protruding cables are for power and master amplitude control via a volume pedal. The Bela is placed underneath the fingerboard in close proximity to the pickups and control panel, while the amplifier is stored inside the double bass body. The Bela and amplifier are powered through a power supply (see figure 2), with the pickups running on separate a 9-volt battery (significantly reducing system noise as opposed to powering them from the power supply).

The design rationale for a self-contained instrument is practical as well as aesthetic: The instrument is quick to set up, with a single switch powering up the electronics and booting the software. Significantly excluding the ubiquitous laptop from the performance setting while retaining the ability to sculpt systemic behaviour afforded by DSP, the instrument is physically self-contained, allowing the player to focus on engaging with the familiar physical features of the instrument, rather than a laptop or external controllers. As such, this system favours tacit musical engagement, offering an opportunity to grow into and deepen an embodied relationship with the instrument.

[3]Thor Magnusson defines ergodynamics as a 'term that somewhat relates to the use of 'gameplay' in computer games, but further signifies an awareness and experience of the instrument in embodied, historical, and aesthetic practices'.

Figure 2: The FAAB, front

Figure 3: The FAAB, back

3.1 Hardware

Four electromagnetic Cycfi Nu[4] single-string pickups are routed through the Bela, an amplifier and finally to an eight-inch 150 watts mid-range speaker installed in the back of the double bass. Here, a hole is cut and a ring of birch plywood is fitted and glued to the rim for reinforcement. The speaker is mounted, facing outwards, on that plywood rim.

The Bela has been expanded with the CTAG Face cape giving us four input channels at a sampling rate of 48000 Hz,

[4]`https://www.cycfi.com/projects/nu-series-v2/` (Accessed 30.04.2020)

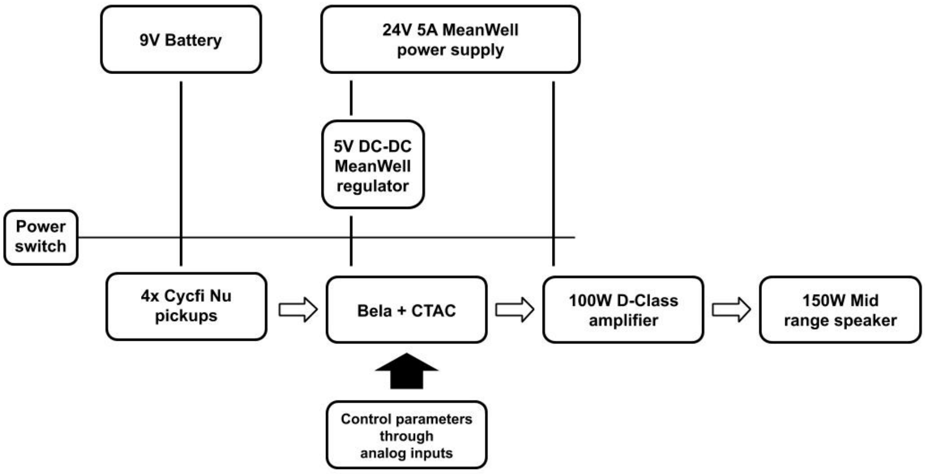

Figure 4: System electronics block diagram

crucial for sculpting feedback via signal processing[5]. Control parameters are constrained to the Bela's eight analogue inputs which are connected to slide and knob potentiometers on the right-hand side control panel. The four sliders are mapped to individual string volume controls (future system design may move this step to an analogue mixer, freeing up mappable control parameters). The remaining controllers are available for additional DSP-mappings, so far mainly adjusting the balance between the clean signal and various processing algorithms running concurrently. See Figure 4 for a block diagram overview of the system.

The flexibility of working with individual audio streams from the four pickups places a computational strain on the Bela processor. We are currently using a buffer frame size of 32 at a sampling rate of 48000 Hz, which we consider an acceptable compromise between high-quality audio, computational flexibility and latency, without audio dropouts and artefacts. In the painstaking work of reducing system noise we have found that all cabling should be shielded. The current iteration of the system excludes a preamplifier circuit designed to boost the signal of the Nu pickups before the Bela stage. This served to reduce system noise during early stages of development but has become redundant as we shielded more cabling and introduced a better 5V regulator.

3.2 Ergodynamic design intent

We are designing a feedback string instrument and as such, focus on and attempt to magnify the qualities of the two constituent main parts: feedback, as a very distinct electroacoustic phenomenon as well as control principle, such as described by Collins [1], and the string instrument itself, being a highly familiar and refined mechano-acoustic interface. Both parts allow for considerable depth of intimate and intuitive sonic exploration.

For simplicity we strive to keep the number of manually controllable DSP parameters to a minimum. This is a major design concern since the performing bass player rarely has any hands free. To facilitate this strategy we derive the majority of control signals from simple audio feature extraction algorithms such as frequency detection and amplitude tracking as well as higher-level analysis and comparisons of those parameters. To that end, the DSP part of the system depends on the careful scaling and mapping of control signals to signal processing algorithms, facilitating adaptive and self-organising behaviour through systemic couplings. We optimise this behaviour to play out with maximum reciprocality between the acoustic and digital aspects of the system while attempting to retain a certain acoustic feel in regards to the behaviour of the processing algorithms (avoiding obviously digital-sounding artefacts).

The DSP design builds on works by, amongst others, Agostino Di Scipio [2] and Sanfilippo and Valle [16] whose practices rely on feedback and are often to be found on an instrument-installation continuum. Yet where the ambient acoustic environment is often a determining factor in the systems developed by these composers, we are concerned with the relationship between instrument and performer and the ergodynamics of the playing experience. Structural and systemic couplings of the constituent mechanical (acoustic), electronic and digital parts of the system allow us to compose a dynamic signal processing flow, shaped in the interaction with a physically familiar instrument. The control-signal feedback described by Sanfilippo and Valle [16] is crucial in this context: By coupling audio feature analysis back into the system, an inaudible feedback coupling between control signals is created, facilitating adaptive behaviour while increasing the number of attractors in the system's state space. The result is an increased variety in how the system responds to its own states as well as inputs from the player, resulting in a lively interface with complex, yet comprehensible behaviour that may ultimately become familiar to the performer.

3.3 Audio feature extraction

Our experiments have shown that FFT-based feature extraction used for identifying higher-level features such as spectral centroid and flatness explored in earlier laptop-based feedback instruments such as *What the Frog's Eye tells the Frog's Brain* is not computationally possible in our current DSP (Bela) environment due to lack of processing power. In order to shape the behaviour of the instrument, our approach is instead to focus on frequency and amplitude tracking and calculating differences between this data across individual time scales and separate audio streams (the four string inputs).

For detecting the center frequency of all four strings summed together, we use SuperCollider's *Pitch* U-Gen, while computationally much cheaper and less precise zero-crossing

[5] `https://shop.bela.io/collections/ctag-multi-channel-audio-system/products/ctag-face` (Accessed 30.04.2020)

detections are used for individual strings. For amplitude we use a number of RMS measurements, all based on applying a low-pass IIR filter on the squared input signal and taking the square root of the output of the filter [14]. By applying two RMS measurements with different filter cutoffs (between 0.2 and 50 Hz) to the same signal and calculating the resulting difference, we obtain a dynamically changing signal useful for mapping to other parameters, such as the smoothing factors of additional RMS-calculations as well as band-pass and comb-filter variables mentioned later (3.4.3). We find that using time-derivatives such as these is a musically satisfying and computationally efficient way of generating systemic variety.

3.4 Signal processing

In the following we describe a number of simple algorithms used for processing the signals entering the Bela from the pickups[6]:

3.4.1 *Time-variant high-pass filter*

Because of the large amount of power needed to drive low-frequency feedback combined with resonant 'dead spots' on the instrument, not all pitches can easily be provoked into feedback, especially at lower volumes.

By introducing a high-pass filter with a low cutoff (< 70 Hz), some pitches can be induced to feed back, even when they are of a fundamental frequency lower than the cutoff. The reason for this is not clear to us and requires further research. However, our experiments suggest that the critical cutoff frequency for inducing feedback is a nonlinear function of the relationships between string frequency, amplitude and resonant nodes on the instrument and as such, varies dynamically with minute shifts in playing positions. Consequently, any linear mapping from frequency detection to filter cutoff is unfeasible. A simple solution has been to map the inversely scaled RMS of each string to the frequency of a triangle wave oscillator, scaled to sweep between values of 30 and 70 and mapped to the cutoff of a high-pass filter. The effect of this mapping is such that when the driving amplitude approaches the point where the string starts to self-oscillate, the sweep slows down, ideally stopping completely when feedback becomes prominent. Since the sweep is required to be rather slow to allow for the amplitude tracker to correctly measure the variations in amplitude, the algorithm is slow in adapting. However, this is aesthetically desirable as too quick a sweep would be too sonically obvious. This algorithm is particularly helpful at low volume, where consistent feedback across playing positions is desired.

3.4.2 *Amplitude sync*

This patch is designed to make every string attempt to sync its amplitude to the mean of the individually weighted amplitude of its two immediate neighbours, considering the 1st and 4th string neighbours for the sake of symmetry.

The difference in RMS between individual strings and the weighted RMS of their neighbours is mapped to the frequency of four triangle wave oscillators. These oscillators scale the input gain of each string to between 0.0 and 1.0, meaning that the closer strings are to each other in amplitude, the slower their gain is adjusted. The amplitude variance across all strings is used as a measure of amplitude homogeneity and mapped (via sample and hold) to the frequency of a phasor sweeping from 0.0 to 1.0. When the sweep crosses a pre-defined threshold (currently set to 0.9) and the variance exceeds 0.01, a new set of weights is generated and the phasor reset while the new distribution range is sampled and mapped to the phasor frequency. The weights are evolved by a genetic algorithm [6] [4], the crossover points and mutation of which constitute the only randomized features of the system[7].

See figure 5 for a diagram of the patch. For the sake of simplicity, only the second string is represented.

Inspired by Strogatz' writing on locally coupled oscillators [18], the syncing algorithm rarely achieves perfect sync, yet exhibits dynamic and adaptive behaviour conducive to performative exploration.

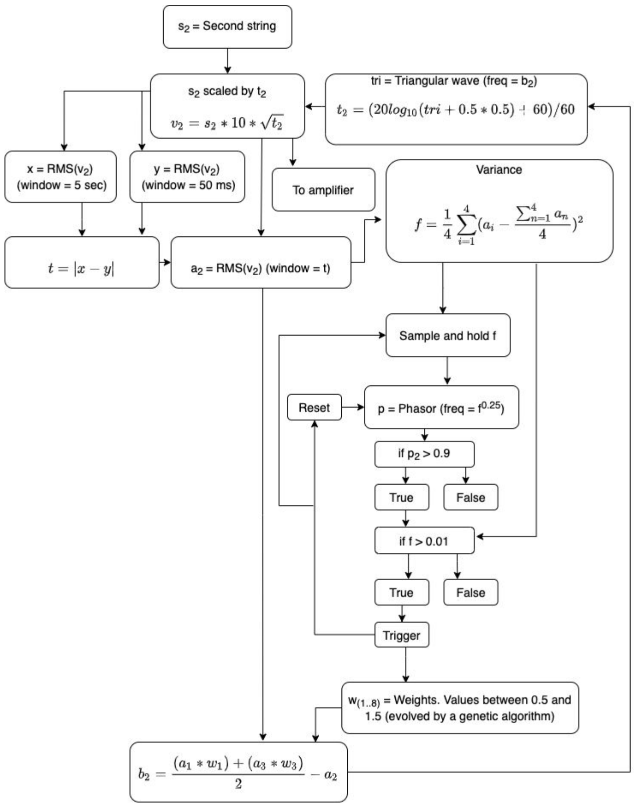

Figure 5: Amplitude sync

3.4.3 *Adaptive filtering*

Since the instrument generally favours low-end feedback, we are currently exploring two approaches to amplifying upper partials through the use of adaptive filtering:

One approach is to apply an array of band-pass filters tuned to the prime ratios of a fixed fundamental relative to the tuning of the instrument (currently 55 Hz). In order to avoid static behaviour due to the fixed fundamental of the filters, we let the difference between slow and fast amplitude tracking determine a frequency offset so that amplitude spikes will push the filters into an inharmonic range, returning in proportion to the convergence of the two amplitude measurements[8].

Secondly, we have implemented an array of comb filters, the resonant frequencies of which are tuned to prime ratios of a zero-crossing frequency-estimate of individual strings. In order to avoid fast or apparent filter sweeps due to changes in the tracked pitch, the signal is smoothed over a window

[6]The code is available at:
`https://github.com/adampultz/faab_nime_2020`
(Accessed 22.04.2020)

[7]Amplitude sync example: `https://vimeo.com/413504674`
(Accessed 30.04.2020)

[8]Band-pass filter: `https://vimeo.com/413504674#t=33s`
(Accessed 30.04.2020)

of 5 seconds. The filter amount is the inverse of the difference between a fast frequency tracker and a slow-moving frequency average, meaning that many filter sweeps will occur while the dry signal is being favoured[9].

Our preference for prime number odd partials produced by the filters is due to the ambition of adding spectral complexity to the signal. A variety of filter tunings can be applied and will be explored in future work.

Both filter implementations result in an extended frequency response that varies dynamically with changes in amplitude and sound spectrum. Through the emergence of additional harmonic and inharmonic content, the filters allow for extended explorations of complex spectra produced by multiphonics and dyads but additionally effect a coupled amplitude-to-timbre relationship, increasing the instrument's sensitivity to bow-placement and pressure, which is a creative asset to our ears.

3.4.4 Amplitude saturation

This patch is a modified adaption from Chris Kiefer [7] and uses the SuperCollider *Integrator* UGen - a simple low-pass filter with the formula:

$$out(0) = in(0) + (coef * out(-1))$$

By calculating the reciprocal of the Integrator and multiplying this with the input signal, the system becomes saturated with gain. The balance between the clean and saturated signal is a function of RMS: when amplitude is low, the wet signal is favoured, whereas higher amplitude causes the dry signal to dominate. Where Kiefer uses the mean amplitude to drive a single Integrator with a summed input of all four strings, we have opted for applying an individual saturation process to each string. Additionally, by letting the RMS time window become a function of RMS changes over time (as mentioned in 3.3) we add yet another layer of adaptive dynamic behaviour. The behaviour of the saturator is sometimes reminiscent of a sustainer, pushing amplitude up to promote feedback, even at low volume levels. This is a desirable quality for us when, for example, used in combination with pizzicato harmonics.

An interesting side effect of the gain saturation is that analog and digital system noise will inevitably become amplified when the strings are dampened. While generally aiming for a low-noise system, the creative possibilities of exploring the balance between string feedback and noise saturation are very attractive and a current path for exploration[10].

4. DISCUSSION

We like this instrument. Considering the FAAB from the perspective of our previous work along similar lines, it has definite benefits and it is clear to us that here is a project on which we will keep working.

From Úlfarsson's perspective, in comparison to halldorphones, the lower register, larger resonant body and higher power ratings for the amp and speaker give it a distinct 'rolling-thunder' quality. To give into sentiment, this author feels: 'its aural presence is awe-inspiring.'

An experimental version of the halldorophone is already Bela-endowed and Úlfarsson's experience of working with that instrument as well as the FAAB is gradually making the DSP design-space more familiar for these instruments. To this author, there is a latent risk of losing interesting systemic behaviour and rendering the instrument predictably manageable by pursuing a route of implementing algorithms that render this manner of feedback as purely a sustain feature (which is certainly possible). The most salient affordances seem to be devising strategies of dynamically controlling gain based on monitoring of the system state, an opinion supported by early findings of Eldridge and Kiefer on their Feedback Cello project [5].

From Melbye's perspective, the ergodynamic shift from an acoustic double bass to the extended instrumental properties of the FAAB demands a new approach to developing performance skills. The adaptive and often nonlinear behaviour of the instrument imbues it with a sense of autonomy that, on the one hand supports exploration, and on the other, resists the traditional notion of instrumental mastery [17] [12]. The intimate coupling between string and bow required by a number of bowing techniques (such as spiccato, multiphonics, subharmonics and sul ponticello) is greatly impacted by the increased amplitude throughput and self-sustaining string oscillations caused by the feedback. Whereas the execution of some of these techniques flow easier on the FAAB, traditionally simple bowing tasks such as the retention of consistency in amplitude and timbre across the instrument become an uncertain enterprise, given the impact of the feedback.

The embedded electronics are an important feature to Melbye: The double bass, DSP engine, amp and speaker, are perceived as a coherent instrument and allow for an embodied engagement with a physically bounded object rather than spatially distributed discrete entities. Phenomenologically speaking, the feel of a physically singular instrument is carried over from the double bass and into the FAAB, allowing the player to keep a strong baseline feeling of familiarity with the system even when the augmented behaviour of that system is unfamiliar.

5. CONCLUSIONS

In the DSP design for a self-contained, feedback-configured double bass with the constraints of onboard processing, we have developed strategies which make for complex, structurally coupled behaviour in the ergodynamics of this hybrid instrument.

The self-imposed computational limitations of aiming for a low-latency and responsive four-channel system running on an on-board microprocessor such as the Bela, is a creative constraint calling for specific coding solutions that could otherwise be executed from a laptop. We have developed low-level audio feature extraction strategies, specifically through the smoothing of control signals over different and varying time scales. Coupling these to filters and amplitude management algorithms, we have created a complex and adaptive environment within the constraints of the current configuration of the instrument. However, being in the preliminary stages of development, we recognize that there are additional computationally cheap time-domain processes to explore, such as those recently proposed by Dario Sanfilippo [15].

The instrument exhibits complex, time-variant and adaptive behaviour and as such, demands the development of new performance practices, inviting of explorative interaction rather than an aspiration toward mastery. For experimentally minded players the FAAB offers an open-ended avenue for the research and development of new techniques and playing idioms, which counts as a positive to the authors.

Feedback and signal processing, by amplifying and gener-

[9] Comb filter: `https://vimeo.com/413504674#t=82s` (Accessed 30.04.2020)

[10] Amplitude saturation: `https://vimeo.com/413504674#t=149s` (Accessed 30.04.2020)

ating spectral features that cannot otherwise be produced by purely acoustic means, has fundamental consequences for how this instrument can be played. Extended and alternative bowing and pizzicato techniques recently developed in contemporary acoustic double bass performance practices [3] [20] [19] are fundamentally applicable to performance with the FAAB. Yet, the physical excitation and timbral colouration supplied by amplification and signal processing requires the performer to adapt any skill set to the additional electromechanical response of the instrument, being offered an extended sonic and dynamic field for exploration in exchange.

A multi-performer survey analysing the ergodynamics of the instrument is warranted and of interest. However, the focus here is on the design of an instrument tailored to the preferences of a specific performer (Melbye) to whom it holds a promise of continued fascination during future exploration. As such, the work to date may be a guide to other designers and artists looking for discrete solutions to bespoke instruments of similar constraints and characteristics.

6. FUTURE WORK

With the majority of the work in this research project being focused on DSP feature design, it is beyond the scope of this paper to discuss the complex relationship between the acoustics of the double bass and the electro-mechanically actuated feedback network. We recognise this as a valuable topic for further research.

A priority for Melbye is the exploration the instrument as it is currently set up, in order to achieve a degree of familiarity (comparable to what he has with a traditional double bass). This work will include developing new techniques and performance tactics while slowly adapting to - and co-evolving with the physical and digital features of the instrument.

Working towards lower system noise and structural stability, we will incorporate shielded casings for the system components, and potentially shield all cabling (as shielding has given good results in some instances).

In an effort to optimize the processing power of the Bela we may reprogram the SuperCollider code in C++ or Heavy Audio Tools [13] to potentially gain more computational headroom, allowing for increased processing possibilities and/or lower audio buffer frames.

Despite the benefits of having a petite, easy to work with, fast to boot, on-board DSP engine, we intend to implement a version of the software on a laptop, allowing us to work with several higher-level FFT-based feature extractions that we have been unable to successfully run on the Bela. Opening up to off-board processes would afford more processing possibilities on the individual string signals and additionally expand the number of inputs, facilitating the creation of coupled networks of multiple instruments.

7. ACKNOWLEDGEMENT

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