2019 IEEE/RSJ International Conference on Intelligent Robots and Systems (IROS)
Macau, China, November 4-8, 2019

A Multi-Channel Embedded DSP Closed-Loop Control System for Musical Robots

Jason Long, Jim Murphy
New Zealand School of Music
Victoria University of Wellington
Wellington, New Zealand
jason.long@ecs.vuw.ac.nz, jim.w.murphy@gmail.com

Dale A. Carnegie, Ajay Kapur
School of Engineering and Computer Science
Victoria University of Wellington
Wellington, New Zealand
dale.carnegie@vuw.ac.nz, ajay@karmetik.com

***Abstract*—The continuous automatic calibration of musical robots is an important step toward enabling them to create accurate, reliable, and expressive performances. Several attempts have been made to achieve this using external laptops, single-board computers and off-the-shelf hardware, but these solutions have various drawbacks that preclude their use in many musical robotics contexts.**

This paper presents a new, custom-built embedded DSP system that is capable of up to 32 channels of high quality audio input and output, created to carry out musical information retrieval tasks on the sonic output of musical robots in real time while they are performing. The results of these analyses are continuously fed to musical robot control hardware in order to inform their performance. The design and construction of the circuit board is described, the Musical Information Retrieval (MIR)-based algorithms carried out by the board are outlined, the system's performance in representative applications are evaluated, and its various implications to the field of musical robotics and beyond are discussed.

By integrating this system into new musical robots, the result is instruments that are able to 'listen' to the audible results of their own actuations in real time, and continuously calibrate their actions to best represent the intentions of the programmed musical composition or live input.

***Index Terms*—Musical Information Retrieval, Musical Robotics, Digital Signal Processing, MIDI**

I. Introduction

Musical robotics is the field concerned with the actuation of real-world acoustic sound objects under computer control, with the objective of creating musical performances that would otherwise be impossible or impractical with live human practitioners alone. While the field has experienced considerable growth in the last decade, there remain significant barriers that prevent its widespread adoption in the world of mainstream musical composition and performance. Many studies in the area focus on the important aspect of equipping musical robots with additional parameters of expression, but as these machines become more complex, controlling and maintaining them becomes ever more involved and burdensome.

One of the primary reasons for this is that unlike human musicians, most musical robots are not able to listen to their own audible output and automatically make adjustments to their playing technique to refine their musical actions. To emulate this biological 'closed loop' in musical robots, it is necessary to equip the instruments with audio sensors such as pickups, coils, or microphones, audio acquisition circuitry, and digital signal processing hardware. To make musical sense of the audio that is being received, musical information retrieval algorithms must be implemented that extract pertinent data such as onset times, latency of actuation, envelope shape, fundamental frequency, and spectral content, among others.

While a number of attempts have been made to implement this functionality with commercially available hardware and personal computers, each of these options suffer from drawbacks that limit their utility in actual performance environments. This paper presents a new multi-channel embedded DSP system designed to address this issue.

To start with, a brief background of the area of musical automata and robotics is offered, to place this research in context. Developments in the more specific areas of the self-calibration and tuning of musical robots are then explored with the weak points of current approaches examined. The specifications and attributes of a system that improves upon the shortcomings of previous efforts are then outlined, and the selection of appropriate hardware is discussed. The newly created system is then presented, with descriptions of the sub-circuits and attributes of the system. Following that, the capabilities of the system are detailed by providing an overview of a range of application scenarios and use-cases, and the novel abilities that integrating this system within musical robots provides are described. Two representative applications of the system are then selected and evaluations are carried out in order to assess the effectiveness of the system. The paper concludes with a discussion of the potential for future work in the area.

II. Background

In the 1970s, inspired by the field of automatic music, Trimpin [1], Godfried Willem Raes [2] and Ken Caulkins pioneered a new wave of automatic instruments, this time under computer control. This new development enabled entirely novel methods of composing for and performing with musical instruments, providing synchronization between remote actuations, the ability to respond to external stimuli and sensors, and enabling the acoustic realization of live adaptive and algorithmic works. This new field has more recently been labeled 'musical robotics', and more detailed histories of the area can be found in [3] and [4].

In the last decade, as microcontroller technology has become more accessible to artists, the popularity of musical robotics has increased, with notable examples of internationally acclaimed musicians such as Bj*ö*rk [5] and Pat Metheny [6] making use of musical robots as centerpieces of their live shows and albums. However, technicians involved in such shows have remarked about the many challenges that are encountered when undertaking performances with musical robots that are unable to tune or calibrate themselves [7].

A. Self-calibration and Tuning of Musical Robots

A number of robotic and manual instruments have utilized automatic tuning to carry out their functions. One example is the Gibson Robot Guitar, which is equipped with motors attached to its tuning pegs, carrying out tuning of the guitar on command. Another instance is Trimpin's 'If VI Was IX' installation, which consists of a large tower of chordophones, including many robotically actuated guitars. Trimpin utilized modified off-the-shelf guitar tuners to generate control signals for a number of tuning actuators [8].

A more flexible approach is outlined in [9], in which the audio output of a robotic chordophone is fed into a laptop running custom tuning software which then sends back the positions of 'virtual frets' to the machine. Another method was trialled in the STARI robot, in which the tuning software is embedded on a BeagleBone single-board computer so as to increase the convenience of the setup [7]. While these approaches prove useful in situations when it is impractical or burdensome to manually tune string instruments, they are all pre-performance systems, and they are not capable of *maintaining* a robot's calibration during a performance.

The Closed-Loop Robotic Glockenspiel remedies this by utilizing a pair of Arduino Due microcontroller boards to implement continuous, audio based, real-time closed-loop control to maintain the instrument's accuracy with regards to velocity and timing [10]. However, the specifications of the hardware used, including 12 bit ADCs, and the general purpose SAM3X8E ARM microcontrollers that were clocked at 84 MHz resulted in bottlenecks that reduced the sample rate, and limited the types of algorithms that could be implemented. The audio channels of the instrument also suffered from high noise resulting from the embedded analogue circuitry used.

III. A New Control System

The shortcomings of the current developments in musical information robotics have inspired the creation of a new hardware system specifically dedicated to the task. In order to be successful in fulfilling the needs of both the aforementioned applications and future ones, the system must provide the following capabilities :

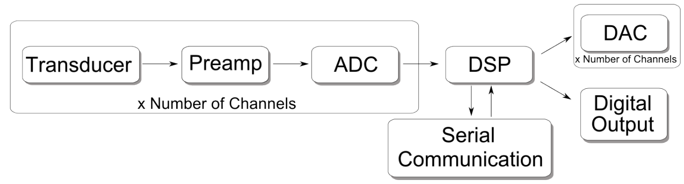

Fig. 1. Signal flow diagram of the new embedded MIR system.

- up to 32 channels of high quality, low latency audio input, to accommodate large installations or polyphonic instruments.
- digital signal processor (DSP) capable of carrying out processes commonly utilized in MIR, such as FFT and FIR filters at sufficient sample rates.
- up to 32 channels of digital audio output, for forwarding the acquired audio to subsequent processors, digital audio workstations or debuggers.
- up to 32 channels of high quality, low latency analogue audio output, for use in distributing the acquired audio or driving closed loop actuated or magnetic resonance instruments.
- modular architecture to facilitate smaller or larger systems.

In addition to the above-mentioned capabilities, the system should also be programmable with inexpensive or free tools where possible, to improve accessibility and affordability in its implementation. Figure 1 illustrates the desired signal flow of the system, with a number of channels being processed by the DSP, and bidirectional serial communication between the DSP and outside systems enabling it to share the musical information that has been retrieved from the audio streams with MIDI hardware or musical robot controllers.

A. Part Selection

In this case, high quality audio is regarded as utilizing a sample rate of at least 44.1 kHz, and a bit depth of at least 16 bits, equal to CD audio quality. Low latency is considered to be in the sub-millisecond range in order to facilitate a responsive 1 kHz control loop frequency. These requirements eliminate the possibility of choosing general purpose microcontroller boards, as their on-board ADCs and DACs rarely meet these bit depths. The low latency requirements also eliminate the use of commercial laptops and audio interfaces whose drivers and operating systems generally require larger block sizes. The Bela cape for the Beaglebone Black single-board computer fulfills the latency and audio quality requirements, but lacks the number of channels required [11].

Of the many DSP chips available, the Analog Devices ADSP-BF592 was selected for several reasons. The BF592 runs at a clock speed of 400 MHz and has dual 16 bit multiply-accumulators (MACs) for up to 800 mega MACs (MMACs). It has the necessary communication peripherals for transmitting the results of analyses to external hardware, complete with a UART, I^2C compliant two wire interface (TWI) and 2 SPI compatible ports. More significantly, it is equipped with 2 dual-channel, full-duplex synchronous serial ports (SPORTs) which are capable of supporting eight stereo I^2S channels, each supporting Time-Domain Multiplexing (TDM) with direct memory access (DMA). This results in a maximum 32 channels of audio input and 32 channels of audio output with very little processor overhead. It is also equipped with 64 Kilobytes of L1 instruction ROM, which contains user-accessible built-in optimized algorithms such as FFT and FIR. The BF592 is also capable of booting from its UART port, and is compatible with the open source GNU Blackfin toolchain. This means that it is possible to program the chip using an inexpensive USB to UART adapter, without the potentially prohibitive cost involved in purchasing proprietary software development environments and in-circuit emulators.

The Cirrus Logic CS5368 was chosen as the ADC, as it meets the resolution, sample-rate, latency, and channel count requirements, it is compatible with multi-channel TDM, and has I^2C and SPI control interfaces. For the DAC, the Analog Devices AD1934 was selected for its similar specifications. There are a total of 1085 components on board, and each segment of the system is described in the following section.

IV. The System's Hardware

The circuitry of the system is housed on a single, 4 layer circuit board 100 mm in width and 150 mm in length, comprised of 2 signal layers, one ground plane, and one power plane. The signal flow is structured similar to Figure 1 in that inputs are received on the left side, processed in the middle, and output on the right. These three stages are outlined in the following subsections, while referring to the annotated photograph of the top side of the PCB provided in Figure 2. Since the ADCs and DACs are the most expensive components of the board, unused banks of inputs or outputs may be left unpopulated, saving monetary cost.

A. Input Stage

The first section of the board comprises 32 analogue inputs split into 4, 8-contact Molex connectors. Banks of ground pins are interspersed with these connections to facilitate analogue audio connections with external equipment. These audio signals are then AC coupled, and passed through a conditioning stage based on the LMV722 dual op amp. This conforms the input signals to the ADC's levels, and converts the single-ended inputs to the differential format that the ADC expects. Each ADC is configured via I^2C on startup to run in 256 f_s mode, and clock their 8 channels of digital TDM audio to the central processor from a master clock generated by a crystal oscillator, and bit and word clocks scaled and distributed by the DSP.

B. Central Processor

The circuitry shown in the top middle of Figure 2 provides regulated 3.3 V and 1.4 V power for the external and core supply lines respectively, and a reset supervisor IC ensures that the unit only boots once both supplies are stable. A jumper provides the options of either booting directly from UART or from an external 16 Mb SPI flash memory on the board. Programming the chip and the external flash memory can be achieved either via JTAG using Analog Devices' proprietary tools, or via UART using open source tools. A Dual Pole Dual Throw (DPDT) switch at the bottom of the board switches the UART connections between the programming port, which is equipped with MOSFET-based voltage converters for 5 V compatibility, and separate communication lines which may be used in the application.

Upon booting, the DSP's firmware assumes control and proceeds to initialize its various peripherals before configuring each available ADC and DAC on the board for operation. This processes takes just a few milliseconds and is essentially instantaneous from the perspective of the user. The chip then begins the continuous transfer of streams of audio in and out by way of the DMA peripheral, and conducts musical information retrieval, digital audio processing, synthesis and any other operations specified in firmware.

C. Output Stage

The TDM outputs from the DSP's SPORTs are routed simultaneously to two destinations: the four AD1934 octal DACs, and four 3-pin Molex connectors to their left in Figure 2. This allows both line level audio output directly from the board, and lossless digital audio output to external circuitry. The analogue outputs from the DAC are passed to an array of TL074 Quad Low-Noise JFET Input Op Amps, running at ± 15 V, that amplify the signals and provide low impedance, AC coupled audio via the banks of outputs on the right.

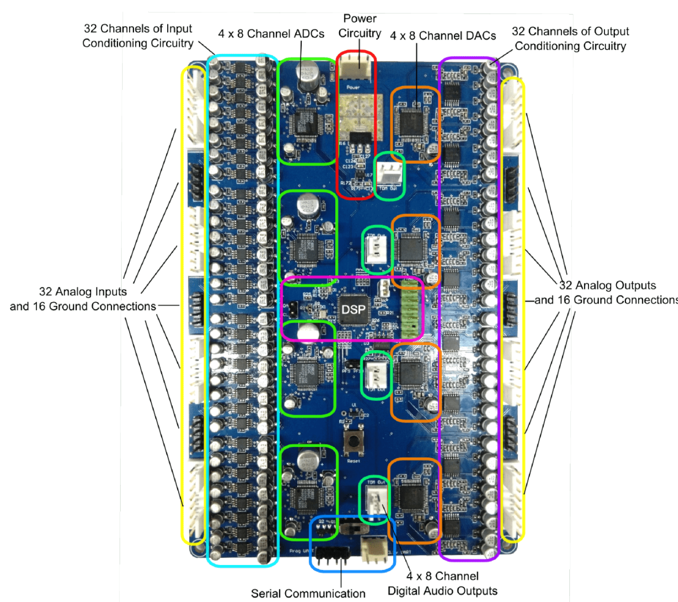

Fig. 2. Annotated photo of the system's circuit board.

V. Example Implementation

The system was designed primarily as a device for enabling musical robots to listen to themselves while they are playing, extract relevant musical information from the audio, and convey that musical information to the robot's control systems. The control systems then compare their intended musical results with the measured ones and continuously calibrate the control of the robot. The first complete integration of the new DSP board is in the modular robotic percussion controller shown in Figure 3.

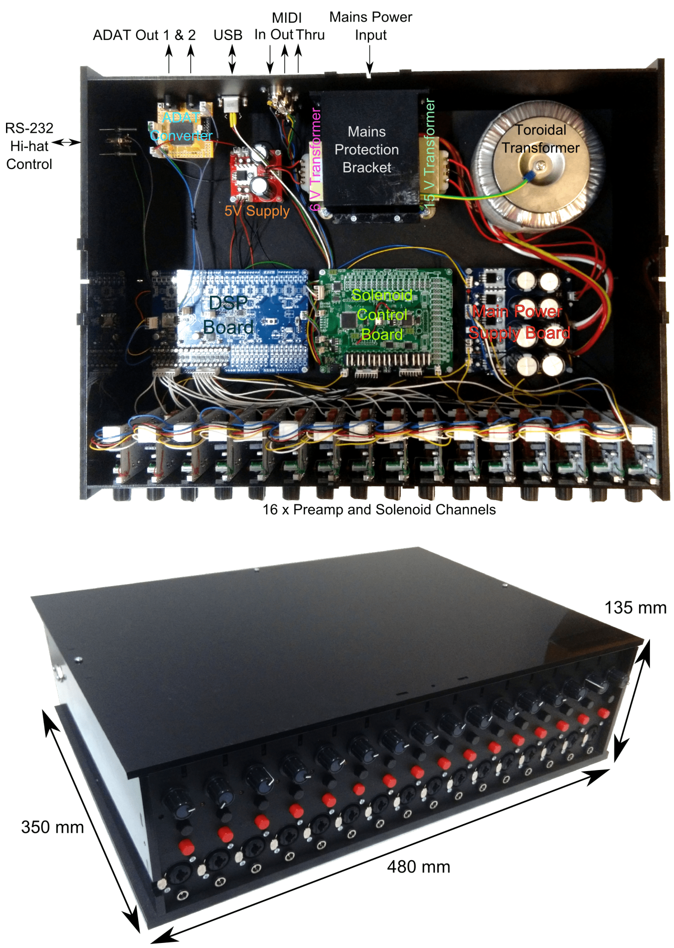

Fig. 3. Annotated photograph of the interior (top) and exterior (bottom) of the robotic percussion array control unit.

This percussion control system utilizes an array of 16 custom preamp boards to condition incoming audio from transducers attached to the musical robots for the DSP board. The DSP board communicates with the FPGA-based solenoid control board, the design of which is described in [12]. The unit interfaces with the outside world via MIDI, RS-232, and USB, and 16 channels of ADAT digital audio output are provided from the DSP board via a custom TDM to ADAT conversion board.

Thus far, the system's primary use has been in controlling a newly built robotic drum kit, utilizing onset detection, envelope following, amplitude detection, and latency compensation algorithms to improve the performance of the robotic instruments. Each of these functions is described henceforth.

A. MIR Algorithms

As MIDI is received by the solenoid control board, the messages are forwarded to the DSP board so that it can prepare to observe an incoming strike. This allows it to conduct analyses only on the short windows of audio where strikes are expected. The DSP board rectifies and maintains a running average of the audio samples in order to extract the amplitude envelope of the incoming signal. Once a predetermined amplitude threshold is crossed, an onset is determined to have occurred. The threshold is hard coded to be above the noise floor of the unit plus a band of safety.

By measuring the time between receiving the MIDI message commanding a strike and the musical onset occurring, a latency value for that channel at that specific velocity is recorded. Following the onset, the amplitude envelope is tracked, and the maximum value that the envelope reaches after the strike is recorded as the strike's amplitude. When the envelope falls back below the threshold, the strike is considered complete, and the time difference between the onset and the conclusion of the strike is recorded as the envelope length.

These attributes of musical information that have been retrieved from live audio of the actuation are immediately sent from the DSP board to the solenoid control board, where the calibration is undertaken. The solenoid control board amends its calibration tables to include the newly received information, with the goal of ensuring that the mapping of MIDI velocity values to resulting strike loudness is a relatively straight, rising line, that latency is made as consistent as possible, and that the envelope lengths of hi-hat strikes increase uniformly as the cymbals are opened.

The solenoid control board uses the latency values to inform a latency compensation algorithm which adds delays to faster strikes in order to create a consistent amount of latency for all MIDI velocity values. Once the latency is consistent, the software or hardware used to generate the MIDI sequences can be set to send them earlier, greatly reducing the musical timing issues that inconsistent latencies cause.

VI. Evaluations

In order to quantitatively evaluate the performance of the new embedded DSP system as part of the example implementation, measurements are carried out that compare the performance characteristics of musical robotic mechanisms with, and without closed-loop calibration. New robotic percussion striking units and hi-hat mechanisms were created for this, detailed in the following subsections, and these were used in conjunction with the instruments of the drum kit, with an example shown in Figure 4. Following a description of these mechanisms, the methodology of the evaluations is explained, and the results are presented.

A. The Robotic Striking Mechanism

The new robotic striking mechanism utilised for this experiment was created following analysis of the results of the robotic percussion studies carried out in [13] and [14]. Figure 4 illustrates the design of the new robotic percussion striking mechanism in its default configuration. The design draws inspiration from several elements of the 'linear solenoid with pivot', and 'Trimpin Hammer' mechanisms outlined in [15], and the 'Kaltron' device described in [16], in that it utilises a pair of male and female ball joints with a 6 mm rod inserted through them, held in place by shaft collars.

A pair of robotic striking modules are shown mounted to a 13" snare drum in Figure 4. The strikers shown are mounted with standard maple 5A sized drum sticks that have been cut to an appropriate length, and have had 5.9 mm holes drilled in the centre of the newly created back plane of the stick. The 6 mm silver steel rod is then pressure fit into the drum stick for a tight, secure connection.

A custom aluminium mount is created to enable the robotic strikers to firmly attach to the snare drum. The bracket attaches to a pair of the snare drum's adjacent UNC 12-24 tension rods, with a hex nut on each side securing the bracket in place. The bracket is secured after tuning of the drum head takes place, to ensure that tuning using the rods does not affect the mounting strength.

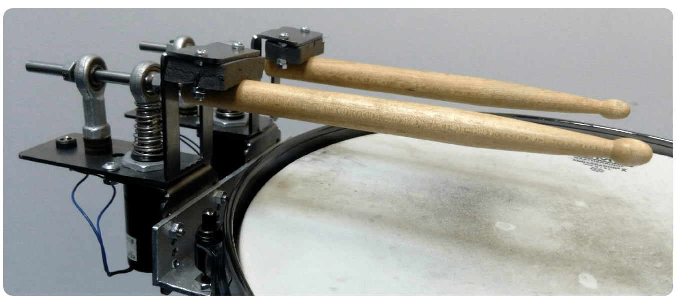

Fig. 4. The robotic striker and snare drum used for the evaluations.

B. The Robotic Hi-hat Mechanism

Unlike the other instruments of the drum kit, hi-hats utilise a pedal to adjust the distance between the two cymbals during a performance in order to expressively manipulate the envelope of strikes to the cymbals, alter the timbre of strikes, and to create a 'chick' sound by closing them quickly, among other techniques. In order to enable these modes of expression with this instrument, a movement system that actuates these types of techniques was built. The hi-hat pedal is replaced by the assembly pictured in Figure 5, which employs a pair of digitally controlled soft-shift solenoid actuators to raise and lower the hi-hats in a fast and strong manner with very low levels of extraneous acoustic noise. Details about the design and construction of this novel robotic hi-hat control system can be found in [17].

Each time a hi-hat is set up, the spacing between the cymbals is configured manually by hand. In human performance situations any inaccuracy in this initialization is automatically compensated for by the drummer via a biological closed-loop, and so is not of significant consequence. However, in robotic contexts without this closed-loop control, even very small variations in the setup lead to significant inaccuracies in the musical output. The embedded DSP system described in this paper seeks to remedy this situation by using acquired envelope length values from audible strikes to the hi-hat for automatic calibration, as these directly correspond to hi-hat position.

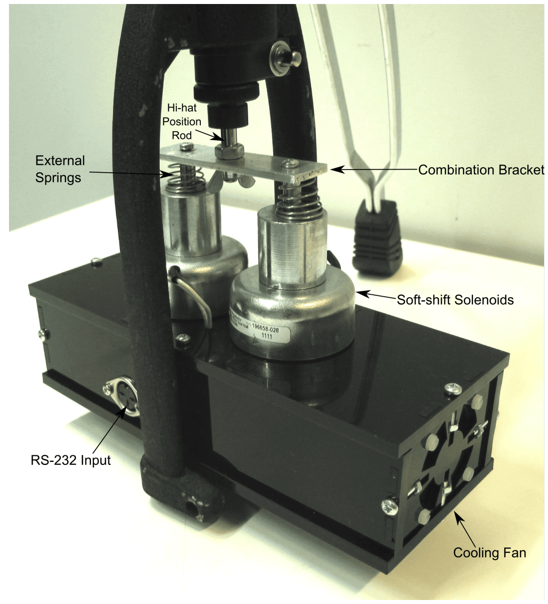

Fig. 5. The robotic mechanism used to raise and lower the hi-hat.

C. Snare Evaluation Methodology

The closed-loop mechanism of the system requires that an audio transducer be directed at the struck instrument in order to feed the instrument's audio back to the DSP board via the preamps. For the snare drum, an Audix i5 dynamic cardioid microphone is attached to its rim using a specialist Audix clip that maintains its position pointing at the center of the drum's head at a 45 degree angle, 50 mm from its surface.

The microphone is connected via balanced XLR cable to channel 1 of the robotic control unit, with the solenoid control signals connected to the same channel. To provide an external measurement of the system, a second microphone is utilised, pointing directly down at the center of the snare drum, 100 mm from its surface. A Shure SM57 dynamic cardioid model is used for this purpose, as it is an industry standard microphone for snare drum recording. The microphone's output is recorded into a digital audio workstation (DAW) via a MOTU 828 mk3 audio interface at a sample rate of 44.1 kHz and 16 bit resolution.

Measurements are first taken of the instruments in their non-calibrated 'open loop' state, where the solenoid control board activates the striking mechanisms and the functionality of the DSP board is disabled. Recordings are made of the percussion mechanism striking the snare drum, powered by a range of different pulse widths, which results in strikes of varying loudness. The latency of each strike is manually measured in the DAW as the time difference between the MIDI note leaving

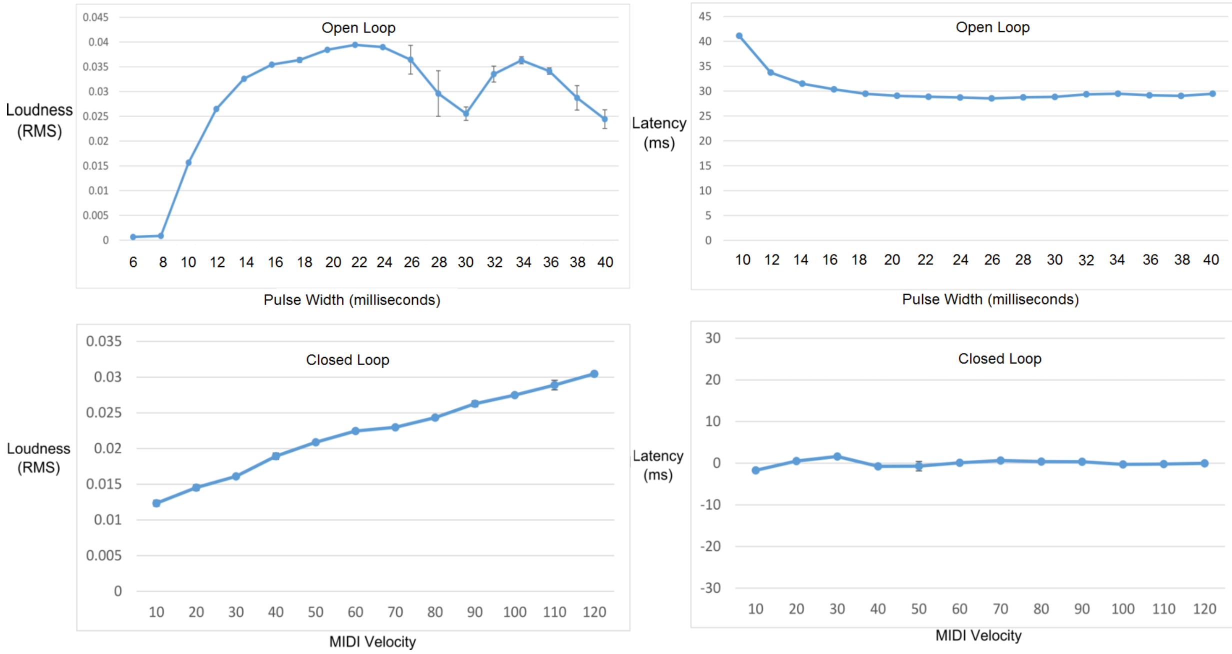

Fig. 6. Graphs showing the loudness (top-left) and latency (top-right) of strikes of varying strengths on the snare drum without calibration, and loudness (bottom-left) and latency (bottom-right) curves achieved by the system following automatic calibration.

the same DAW, and the onset of movement in the recorded audio as observed by a manual operator. Two-second renders from the onset of each strike are exported from the DAW, and the loudness of each strike is extracted using a script written in the ChucK musical programming language. The script carries out spectral RMS analyses on 1024-bin blocks of the audio and sums them, generating a loudness value for each strike. Each of these measurements is undertaken 10 times, and the standard deviation of these are illustrated as error bars in the upper graphs presented in Figure 6.

Following the open loop measurements, the closed-loop automatic calibration functionality of the DSP board and the latency compensation functions of the solenoid control board are enabled. A sequence of MIDI messages of varying velocities across the entire range are sent to the instrument, simulating a short musical performance. While each strike is taking place, the analyses described in Section V-A are being conducted and the instrument is being automatically calibrated. After this simulated performance, the calibration is frozen so that an accurate snapshot of it can be acquired. The measurements of the loudness are then repeated with respect to the MIDI velocity values in order to establish the extent to which the system is capable of successfully mapping the velocity values to a linearly rising range of loudness. Measurements of the latency of the strikes are also taken, including the compensation that has been conducted by the DAW.

1) Open Loop Measurements: Figure 6 presents two upper graphs which show the latency and loudness of strikes powered by a range of pulse widths. The pre-calibration loudness graph shows that pulse widths of 6 ms and 8 ms are too short for the striker to make contact with the drum head. The loudness of strikes rises from 10 ms to 16 ms, and there is little change in loudness from 16 ms to 24 ms. Pulse widths longer than 24 ms show much higher standard deviation values, and the relationship between higher pulse widths and loudness is broken. This is because these longer pulses prevent the drum stick from bouncing freely from the drum head, dampening the head's vibration to varying degrees. If working as intended, the automatic calibration mechanism should map MIDI velocity values to the range of approximately 9.5 ms to 16 ms to ensure linearly rising loudness for linearly rising velocity values.

The latency graph shows higher latency for strikes of lower pulse width, decreasing as the loudness increases. Latency values are not recorded for pulse widths of 6 ms and 8 ms, as these values did not make contact with the drum. By automatically recording this latency curve, the algorithm can apply corresponding amounts of extra delay to make the resulting latency consistent between strikes of varying velocity.

2) Closed Loop Results: The bottom two graphs of Figure 6 show the latency and loudness of strikes at a range of MIDI velocity values after closed-loop calibration has taken place. The loudness graph shows a relatively linear relationship between increases in MIDI velocity and increases in the loudness of strikes to the snare. This shows that the system correctly mapped pulse widths to the range of MIDI velocity levels.

The latency graph shows the measured latency values of drum strikes over the range of MIDI velocity values after latency compensation has been conducted by the DAW. An ideal result would be a straight line with all values at zero latency. The results show that the latency detection, mapping,

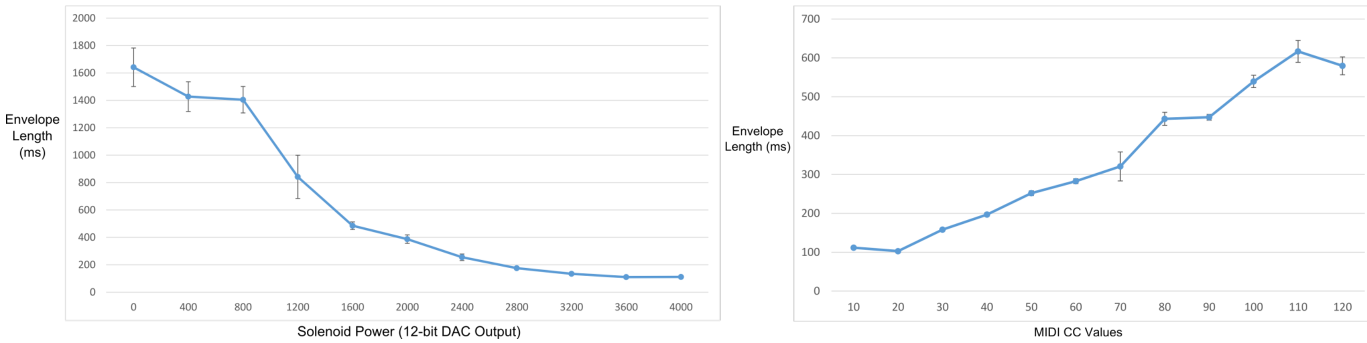

Fig. 7. Graphs showing the hi-hat envelope lengths before calibration (left), and the envelope lengths of strikes at various MIDI CC values following auto calibration (right).

and compensation functionality works as intended, with all but two of the values less than 1 ms away from zero, and the remaining two less than 2 ms away. The standard deviation over the entire range is 0.84 ms, a significant improvement over the standard deviation of 3.16 ms that would be observed with an ideal latency compensation value in the open-loop scenario.

D. Hi-hat Evaluation Methodology

The hi-hat setup is similar to the snare's, except that the closed-loop transducer is an Audix ADX-51 small diaphragm cardioid condenser microphone positioned 100 mm above the top hi-hat, pointed vertically downward at the edge of the cymbal.

The hi-hat calibration system measures the duration of amplitude envelopes of strikes to the instrument in order to inform the positioning system. To verify its functionality, external measurement of the envelope lengths are carried out. As manually distinguishing the end of an exponentially decaying envelope to a degree of accuracy in a DAW is problematic, the envelope lengths are verified using the pulse width measuring function of a Rigol MSO1104 mixed-signal oscilloscope. This technique provides much greater reliability of measurement than manual methods, though variation in the measured length tends to be proportional to total length, instead of absolute.

The microphone is connected to channel two of the robotic percussion array control unit, with the hi-hat striker control line connected to the same channel. All strikes of the hi-hat are powered by pulses 11 ms in length, the middle of the dynamic range of the hi-hat striker. The open-loop set of measurements are carried out with all of the automatic calibration functionality disabled. Following the open-loop measurements, the automatic calibration functionality is enabled and a series of low, medium and high MIDI CC values are sent to the hi-hat, each followed by a strike. This process simulates a sound check or performance rehearsal, and during this time the closed-loop automatic calibration is taking place in the background. The calibration is then frozen in order to capture a snapshot of the system's response. Each measurement is taken ten times in order to provide standard deviation values, as shown in the error bars of the graphs in Figure 7.

1) Open Loop Measurements: The left-side graph of Figure 7 shows the envelope lengths of strikes to the hi-hat, with varying degrees of power supplied to the soft-shift solenoids in the hi-hat movement system. The lowest power value shows the envelope length of strikes to the top cymbal when it is making no contact with the bottom cymbal. The 400 and 800 values show the result of very slight and occasional contact between the top and bottom cymbals after a strike. In hi-hat performance practice, it is relatively rare to strike the top hat in a position of zero contact with the bottom cymbal, as the particular strength of the instrument is the adjustable contact between the two cymbals.

After these values that show the hi-hat entirely and almost entirely open, there is a significant decrease in envelope length as the cymbals come into further contact. A higher standard deviation is also observed, pointing to some inconsistencies in the response of the soft-shift solenoids in their very low power state. By the 1600 mark, the cymbals are making more stable contact, and the solenoids are in a more consistent area of their power curve. This is indicated by the much lower standard deviation values observed. The 1600 to 3600 area of the graph shows a relatively linear decrease of envelope length to solenoid power, and this is where the majority of the musical expression of the instrument lies. At the 3600 point, the cymbals are tightly closed, and additional power does little to affect the sound of strikes and length of envelopes.

2) Closed Loop Results: The right-side graph of Figure 7 presents a graph which shows the resultant envelope lengths of strikes to the hi-hat that have been mapped to MIDI CC values via the closed-loop automatic calibration system. Before calibration, CC values are mapped over the full 12 bit solenoid power range from 0 to 4095, and the ideal envelope length range is specified as the MIDI CC value multiplied by five. The graph shows that the system was successfully able to scale the solenoid power levels to the MIDI CC values to resemble the specified ideal graph relatively closely.

The flatness at the bottom of the graph is a result of reaching the hard limit of tightness described in the open-loop trial, and similarly the lower values show very low standard deviations with the higher values exhibiting somewhat less consistency. Given that many widely used General MIDI (GM) systems utilize just closed, open, and 'chick' hi-hat sounds, and the designers of many high-end virtual drumming software systems may sample up to 6 or 7 different hi-hat positions, these results show that this new DSP board coupled with these closed-loop algorithms and the robotic hi-hat system provides a higher level of expression than such solutions.

VII. Conclusions

This paper has presented a multi-channel embedded DSP system for the express purpose of providing musical robots with the ability to 'hear' the results of their own actuations as they are conducted, and continuously calibrate their performance. It provides practical benefits over previous systems as it enables these functions without requiring an external laptop computer, conducts audio analyses at high sample rates and bit depths with low latency, and is implemented with a modular architecture that provides this functionality with up to 32 input and 32 output channels per board, with multiple boards able to daisy-chain communication for even higher channel counts. Real-world performance experience with the system integrated within a robotic percussion control unit, coupled with quantitative evaluations, show promising results that indicate significant improvements over traditional musical robot control systems.

A. Future Work

The example implementations described in this paper are only two of many potential possibilities of the presented system. The large numbers of audio channels mean that the system would work equally well in polyphonic percussion situations with one channel per note such as in robotic vibraphones, marimbas or xylophones. The large number of DAC channels also facilitate integration with magnetic resonance instruments such as the Magnetic Resonator Piano [18] that utilizes an audio-rate closed loop for synchronizing the phases of the resonating bodies and the output waveforms used to excite them. These types of instruments that have thus far predominantly relied on attaching laptop computers, and utilizing polyphony-limiting multiplexing techniques, can now dedicate individual channels to notes, and embed the signal processing inside the instrument.

While the example implementation focused on latency, amplitude, onset, and envelope detection algorithms as those are most applicable to percussion contexts, algorithms that apply this kind of audio feedback based closed-loop control to different categories of musical instruments are also possible. Examples include pitch detection algorithms for closed-loop tuning of robotic chordophones, and FFT-based timbre analysis, for recognizing the distortion points of resonating bodies.

There is still much work to be done to realize the full potential of utilizing this hardware to create novel robotic, actuated, and magnetic resonance instruments, along with hybrids of these three. For the applications trialled thus far, the resources of the chosen DSP are sufficient and higher level MIR such as phrase and pattern detection are also entirely possible. Future iterations of the board could also utilize more powerful DSPs in order to conduct more intensive analyses on many channels simultaneously. It is hoped that by developing this dedicated embedded hardware, advances in the field of MIR can be utilized to a greater extent in musical robotics, pushing the capabilities of musical machines into new territory.

References

[1] A. Focke, Ed., *Trimpin: Contraptions for Art and Sound*. Seattle, Washington: Marquand Books, 2011.

[2] L. Maes, G.-W. Raes, and T. Rogers, "The man and machine robot orchestra at logos," *Computer Music Journal*, vol. 35, no. 4, pp. 28–48, 2011.

[3] J. Long, J. Murphy, D. Carnegie, and A. Kapur, "Loudspeakers optional: A history of non-loudspeaker-based electroacoustic music," *Organised Sound*, vol. 22, no. 2, pp. 195—-205, 2017.

[4] J. W. Murphy, D. A. Carnegie, and A. Kapur, "Musical robotics in a loudspeaker world: Developments in alternative approaches to localization and spatialization," *Leonardo Music Journal*, vol. 22, pp. 41–48, December 2012.

[5] Björk, "Biophilia," 2011, one Little Indian Records.

[6] P. Matheny, *Orchestrion*. New York: Nonesuch Records, 2010.

[7] S. Trail, L. Jenkins, D. MacConnell, G. Tzanetakis, M. Cheng, and P. Driessen, "Stari: A self tuning auto-monochord robotic instrument," in *Communications, Computers and Signal Processing (PACRIM), 2013 IEEE Pacific Rim Conference on*. IEEE, 2013, pp. 405–409.

[8] P. Esmonde, "Trimpin - the sound of invention," 2011, microcinema International, DVD.

[9] J. Murphy, P. Mathews, A. Kapur, and D. A. Carnegie, "Robot, tune yourself: Automatic tuning in musical robotics," in *Proc. of the International Conference on New Interfaces for Musical Expression, NIME*. Goldsmiths University of London, 2014.

[10] J. Long, A. Kapur, and D. Carnegie, "The closed-loop robotic glockenspiel: Improving musical robots using embedded musical information retrieval," in *Proc. of the International Conference on New Musical Interfaces for Musical Expression*. NIME, July 2016.

[11] A. McPherson, R. Jack, and G. Moro, "Action-sound latency: Are our tools fast enough?" in *Proc. of the International Conference on New Interfaces for Musical Expression, NIME*. Griffith University, 2016.

[12] J. Long, J. Murphy, D. Carnegie, and A. Kapur, "Improving the accuracy of musical robots with programmable logic circuits," in *Proceedings of the Electronics New Zealand Conference, ENZCon*. Victoria University of Wellington, 2016.

[13] J. Long, J. W. Murphy, A. Kapur, and D. Carnegie, "A comparative evaluation of percussion mechanisms for musical robotics applications," in *6th International Conference on Automation, Robotics and Applications, ICARA*. IEEE, 2015, pp. 173–178.

[14] J. Long, J. Murphy, A. Kapur, and D. Carnegie, "A methodology for evaluating robotic striking mechanisms for musical contexts," in *Proceedings of the International Conference on New Interfaces for Musical Expression, NIME*. Louisiana State University, 2015.

[15] A. Kapur, Trimpin, E. Singer, A. Suleman, and G. Tzanetakis, "A comparison of solenoid-based strategies for robotic drumming," in *Proceedings of the International Computer Music Conference (ICMC)*, 2007.

[16] A. Kapur, J. Hochenbaum, M. Darling, D. Diakopoulos, and J. W. Murphy, "The karmetik notomoton: A new breed of musical robot for teaching and performance," in *Proceedings of the International Conference on New Interfaces for Musical Expression, NIME*. University of Oslo, 2011, pp. 228–31.

[17] J. Long, J. W. Murphy, D. Carnegie, and A. Kapur, "A closed-loop control system for robotic hi-hats," in *Proceedings of the International Conference on New Interfaces for Musical Expression, NIME*. Aalborg University, 2017.

[18] A. McPherson, "The magnetic resonator piano: Electronic augmentation of an acoustic grand piano," *Journal of New Music Research*, vol. 39, pp. 189–202, 2010.