Critical Assessment Document V-IR-Bull: Convolution Reverberation

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***Abstract*— V-IR-Bull currently offers two Convolution Reverberation systems; one specific to audio professionals, and one for the consumer audiophile. These systems work to artificially recreate the reverberant characteristics of the acoustically desirable spaces of the users choosing in three stages of listening or recording applications: Reverberant Capture, Audio Convolution, and Music Playback.**

INTRODUCTION

SOUND IS EVERYTHING. It moves the air we breathe. It shapes the rooms in which we thrive, lending us clues as to their vastness – to quite literally the fabric of their existence. Sound is how we communicate not just laughter and song, but with the very walls that surround us. V-IR-Bull's Convolution Reverb gives you the freedom of sound in any space, allowing you to take the fingerprint of your favorite rooms with you for music playback or recording applications, wherever you may wander.

AUDIO HARDWARE

Professional System

Each audio engineer has their personal preference in gear choices and the professional system is designed to easily integrate the audio professional's arsenal of professional-grade gear used in recording and mixing applications. The following describes the characteristics and I/O requirements to successfully interface one's own gear with our system.

1. Studio-Grade Equipment Requirements

The microphone used in capture is of particular importance; it must be sensitive enough to capture all auditory nuances whilst maintaining near-perfect flatband pick up characteristics. We strongly recommend using a DPA4011 condenser microphone, as it was originally designed for use in laboratory setting and consequently exhibits a near-perfect flatband pickup characteristic, with 3dB corners at 20-20kHz (the audible range) [1]. We also selected an ElectroVoice ZLX-12P 1000W self-powered speaker for our capture stage, for its ability to reproduce the audio signal sent to it with relatively equivalent energy output across 20-20kHz. And, finally a mixer that hosts, at minimum, 2 channels with gain control, each featuring an option to host phantom power to the mic-in connections as well as read line-in and send line-out to each channel, an alt line-out L/R, and a main mic-out with independent gain control to send to the PA speaker. We utilized a Mackie 1202VLZ.

2. Prototype Configuration

This prototype is featured in its current stage in Figure 1, and is nearing completion proportionate to funding available. The front panel features cut outs for the pushbutton and LCD control networks, and laser etched labeling. Also prepared for installation are slots for USB and HDMI access into the Raspberry Pi network. The 3.5mm holes for audio inputs and output connectors are also drilled and labeled.

The rear panel features placement for a power converter and master power switch. Also ready to install is a cutout for the fan with independent power control for deactivation during capture.

Consumer System

The consumer system is geared primarily toward playback, allowing the user to listen to their own music within our built-in presets (pre-captured impulse responses ready to use in convolution), or even capture their own rooms.

1. Consumer-Grade Equipment

A cubby slot has been built into the box for the integration of an AKG 120 Perception condenser microphone, or a condenser mic of equivalent flatband pick up and size. Due to their capsule design, condenser mics feature consistency in energy capture across audible frequencies, using the movement of a diaphragm in response to a sound pressure waves to fluctuate the capacitance between it and a back plate, generating an analog electrical representation of the frequencies 'heard' in real time. However, condenser mics need 48v of "phantom" power to be applied across the capacitor terminals to properly operate, and so a small 48v power supply is incorporated into this unit; an ST-MPA48 Dual Microphone Phantom Adaptor [2], which supplies this potential difference.

The output of the microphone is pulled from the ST-MPA48 and fed into an LM4562 Differential Input Microphone Preamplifier (see Figures 2 and 3), which we have bread boarded, etched and soldered. The resistor ladders in this Preamp design provide a gain of 91.09v/v, or 39.2dB, which is sufficiently close to the standard 40dB preamp gain used before mixing. To power the op amp chips used, a 15v Power supply has been also integrated into this design.

For capture and playback purposes, a Bose Speaker is housed into the front panel of this system, and offers easy access to control the gain of the output signal. This is especially essential in the capture stage, as between 85dB and 90dB must be present at the microphone during capture (as determined by Lucas Film, Ltd. and the industry standards in evaluating speaker performance). A cubby for SPL Meter has also been incorporated into the prototype to be clipped to the mic stand for monitoring the sound pressure level present at the microphone.

2. Prototype Configuration

This prototype is featured in its current stage in Figure 4, and is nearing completion proportionate to funding available. The front panel features cutouts for the pushbutton and LCD control networks, and laser etched labeling. The Bose speaker is situated into this panel as well, with its gain knob readily accessible. The holes 3.5mm audio output connectors are also drilled and labeled.

The rear panel features placement for power converter and master power switch. Also ready to install is a cutout for the fan, with independent power control for deactivation during capture. Also prepared for install are slots for USB and HDMI access into the Raspberry Pi network.

Also incorporated into the prototype is an XLR Female throughput connector on a side panel (which is connected to the rails on the phantom power box, supplying 48v across the capacitive plates in the condenser mic, and outputs zero bias to the LM4562 Mic preamplifier) as well as the cubbies for the placement of the microphone and SPL meter on the top panel.

Controls Hardware and Functions

Wolfson Network Sub-System

1. Wolfson Audio Card

This card (see Figure 5) hosts high fidelity I/O functions to and from both analog and digital outputs and inputs. This is due to the ADAC (analog to digital conversions and vice versa) function it performs, which is easily programmed for recording and playback in a variety of resolution bit sizes and sampling frequencies. For this stage of prototyping, we have programmed the card to run I/O at CD fidelity: 16bit resolution and 44.1kHz sampling rate. The architecture of the main audio functions can be seen in Figure 6, supplied by the Wolfson user manual [3], featuring the signal flow and pin outs of the WM5102 and WM8804 chips.

The Wolfson Audio card communicates to the Raspberry Pi through p5 pads and GPIO pins. The Raspberry Pi is where the Wolfson modified Raspian Linux operating system is run to maximize utilization of card's feature, which we imaged to 16GB Class 10 SD card.

2. I/Os and Prototypes

Both prototypes use the analog 3-pole line-level input jack, the analog 3-pole line-level output jack and the small signal 4-pole headset jack on the Wolfson Card. The professional system inputs to the 3-pole line-in directly from the mixer through front panel connectors, using an instrument cable pulled from the alt-out, fixed with a 1/4 inch to 1/8 inch adaptor. The consumer system inputs to the 3-pole line-in internally, pulling the signal direct from internal Mic Preamp (which is fed from the ST-MPA48, which inputs the signal from the female XLR connector on side panel). Both systems output through either the line-out or headphones jacks via the front panels, which are readily accessible for hookup into user's mixer, DAW or stereo system. (Note: the consumer system utilizes a splitter internally, splitting the line-out signal between the Bose speaker and the 3.5mm jack on the front panel.)

3. Capture Signal Flow

1. A 10s 1kHz test tone is used to establish 85-90dB at the microphone capsule, using an SPL meter to measure.
2. A 16 bit 10ms pink noise chirp track is played out of the LX Music player from the Wolfson Raspberry Pi sub-system, and is emitted by the speaker.
3. The microphone captures a 10ms chirp and the reverberated copies and reflections of the chirp from the room, thus providing the frequency and timing information necessary to define the acoustic characteristics of the room.
4. The captured audio signal is sent from the mic through a mic cable with an XLR connector, into a microphone preamp, which amplifies the signal by about 40dB and into the analog line-in jack on the Wolfson audio card.
5. This sound-card uses its onboard ADAC to convert the analog electrical signal to digital data, ready for manipulation by a digital audio workstation (DAW) installed on the RPi (Note: Audacity currently acts as the DAW for the consumer system, for which step by step guide has been written and would be included in the literature with the product).
6. After proper editing is performed in the DAW on the capture, it is then exported as a .wav file.
7. The Wolfson RPi system then sends this capture.wav file to the Convolution Raspberry Pi sub-system, which performs FFT analysis to produce an impulse response map of the room.
8. This is impulse response then convolved with a music track of the users choosing in the modified SciPy program.
9. When the convolution stage is completed, the convolution Pi network sends the convolved .wav file back to the Wolfson Pi, which can then either be played back through the line-out or headphone out as-is in .wav format using LX music, or can be run on through a conversion program to compress the file into mp3, mp4, or AAC (& etc) file formats size-friendly to personal listening devices.

Note: Also an option, the Convolution Pi offers USB access to the SD memory, such that the user can save their new music tracks onto their USB flash drive for use in other applications, such as mixing.

Convolution Pi Network Sub-System

1. LCD Display and Pushbutton Controls

As has been mentioned in the Prototype Configurations, the Convolution Pi sub-system has integrated a 16x2 LCD display and pushbutton controls (See Figures 6 and 7). These are all interfaced from the RPi's GPIO pins, and uses pull-down resistors and clever coding to give them function and purpose.

Currently the Preset pushbuttons are programmed to call upon specific 'presets' to utilize in the convolution stage with the user's music files. When any button is pressed the LCD display will show the name of the preset called upon. The LCD screen is also currently accepting messages instantaneously from a HTML user interface, and will soon be used as the visual to browse through an independent computer and upload music files without a wired connection to the physical RPi networks, as is described in a later section. The LCD also has been wired to a potentiometer knob for control over the screen brightness.

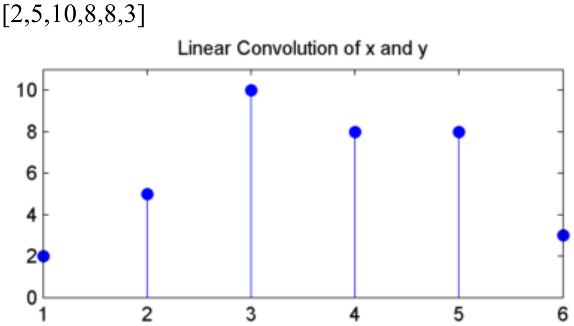
As we continue to gain prowess in programming skills, more buttons will act as controls for 'automatic' functions triggering the capture, upload, convolve and export functions. Currently, the control pushbuttons successfully activate a Raspberry Pi's built-in record function.

2. Convolution Code

<u>Section 1: Linear Convolution Vector Calculation</u>
"For linear convolution of an N-point vector, x, and a L-point vector, y, the convolved signal has length N+L-1"
n = number of samples in vector
Output[n] = dry[n] + IR[n] – 1
Using MATLAB's wavfinfo function to get # of channels and # of samples in the file:
>> [x,n]=wavfinfo('Gtr2.wav')
x =Sound (WAV) file
n =Sound (WAV) file containing: 376126 samples in 2 channel(s)
>> [x,n]=wavfinfo('Stairwell.wav')
x =Sound (WAV) file
n =Sound (WAV) file containing: 95482 samples in 1 channel(s)
>> [x,n]=wavfinfo('STgtr.wav')
x =Sound (WAV) file
n =Sound (WAV) file containing: 471607 samples in 1 channel(s)
So: Gtr.wav had 376,126 samples per channel for a 2 channel file.
Stairwell.wav had 95,482 samples per channel for a 1 channel file.
Gtr convolved with Stairwell
= 376,126 + 95,482 -1 = **471,607**
ST.wav was the convolved signal. It contained **471,607** in 1 channel. *Verified!!*

Actual convolved signal using Raspberry Pi FFT Convolution:
All files outputted to 16 bit, 44.1kHz sampling rate (CD quality fidelity).
Dry voice file = 35,584 samples
Vocalbooth IR = 4208 samples
Convolved = 35,584 + 4,208 -1 = **39,791 samples**
Wavfinfo function MATLAB
>> [x,n] = wavfinfo('dry.wav')
x =Sound (WAV) file
n =Sound (WAV) file containing: 35584 samples in 1 channel(s)
>> [x,n] = wavfinfo('MorsVoxbooth10.wav')
x =Sound (WAV) file
n =Sound (WAV) file containing: 4208 samples in 1 channel(s)
>> [x,n] = wavfinfo('MATLAB LinConv.wav')
x =Sound (WAV) file
n =Sound (WAV) file containing: 39791 samples in 1 channel(s)
>> [x,n] = wavfinfo('Rasp Pi LinConv.wav')
x =Sound (WAV) file
n =Sound (WAV) file containing: 39791 samples in 1 channel(s)
>> [x,n] = wavfinfo('Rasp Pi FFTConv.wav')
x =Sound (WAV) file
n =Sound (WAV) file containing**: 39,791** samples in 1 channel(s)

Linear convolution function (convolve(x,y))
In practice the dry input signal is x[n] and the impulse response is h[n] or y[n] in this example. The linear convolution of the two signals can be applied as:
Ex:
x[n] = [2,1,2,1]
y[n] = [1,2,3]
x[n] convolved with y[n] = [(2*1), (2*2 + 1*1),(2*3+1*2+2*1), (1*3+2*2+1*1),(2*3+1*2),1*3)] = [2,5,10,8,8,3]



Our .wav files have values between -1 and 1 and have over 50,000 samples in each file so by hand calculations are deemed too time intensive. MATLAB has done the job for us.

<u>Section 2: Output Bit Rates of Convolved File</u>
Output bit rate (kbits/second) = (Sampling rate)*(Bit depth)*(Number of channels)
Nyquist Theorem states that the sampling rate has to be more than twice the maximum frequency in the signal under analysis otherwise aliasing may occur. If aliasing occurs, then frequencies that were not originally present in the signal could show up and create noise in the signal. The highest frequency in our sound signals is 20kHz, so the CD quality 44.1kHz is more than enough to cover the spectrum of human hearing.

Quantization requires that the number of levels that a signal is divided up into vertically can be expressed by (2^(n-1)). If the bit depth is very small, then the signal could be misrepresented by the analog to digital convertor built into the sound card on the computer.

For a stereo .wav file at CD quality 16 bit:
(44,100*16*2) = 1411.2 kbps
For a mono .wav file at CD quality 16 bit:
(44,100*16*1) = 705.6 kbps

EDR Plots in Preset Imaging

EDR stands for Energy Decay Relief plot. This method of graphing is a 3-dimensional representation of a signal, and we have generated EDR plots to exemplify the differing characteristics of our preset captures. The Kaiser windowing method is applied, versus the more common Hanning window, because Kaiser makes the Fourier Transform side lobes to be constant (-80dB) across frequencies away from the first harmonic being calculated and the Kaiser window produced more identifiable points in our EDR plot that we could see the varying changes over time much more easily. The spectrogram function in MATLAB computes the Short Time Fourier Transform (STFT) and then the EDR is calculated once that is determined for the .wav file. [4]

Server Interface

Our next upgrade to these systems is the use of local servers as a user interface option to upload and download the music files desired to be used in convolution, as well as for RPi-to-RPi communication (they can currently only send data through GPIO). The current method for user music upload and download is through the use of a USB flash drive connected to the Wolfson Pi, which saves to the SD card.

We have tested several server methods: PHP, FTP, and HTML. The PHP server method was researched, developed and proven effective for basic commands, and was successfully developed to interface from PiBuddy iOS iPhone app to control individual LEDs. The FTP method was researched, developed and was proven effective in browsing and securely transferring files from a computer to the Raspberry Pi, using only the PC, but only after converting said file to binary format.

We chose the HTML method for server-based interface, and would need more funding and a computer programmer to perfect. Currently, as proof of this concept; it communicates to the Convolution Pi system (including the LCD display), and is operating to prompt the user to input a message to send to the RPi and LCD display via a browser portal, which then upon submission displays immediately on the LCD (See Figures 8 and 9). This interface would need to browse files on a computer, and allow the user to upload files and not just text to be fully useful for our designed purpose.

Power Distribution

Each system features an internal power rail activated by a switch built into the three-prong adaptor we selected. When the master power switch is activated, 120VAC potential appears across rail. The devices powered from this are connected to this from one pole, with the other run to switches on the back panel, which are in turn connected to rail for independent control of each system necessitating such.

The power supply running the fans on both prototypes are designed utilizing a step-down transformer from 120VAC to 12VAC (13.4VAC measured), which then rectifies the power to 11VDC using a diode bridge, and then finally employs an LM317 Voltage Regulator to 7.2VDC (see Figure XX).

Each Raspberry Pi is powered by a 5.1VDC, 1Amp iPad USB adaptor, connected in the aforementioned configuration to the rail and switches. In the consumer system, the ST-MPA48 was powered from the same system developed for the fan, but wired for 12VDC instead of 7.2VDC, the 15v+/- supply was run directly to the rail, as was the Bose speaker.

EVALUATION OF RESULTS

Convolution Reverberation's most important spec/requirement was to be able to convolve the impulse response of a room with a music file recorded in another space, and have the result sound as if the music file was being performed in the captured space by taking on its reverberant characteristics. In this mission we were ultimately successful. The Wolfson Pi subsystem inputs and outputs audio through ADAC, and hosts a number of DAWs for capture recording and editing. The Convolution Pi runs the modified SciPy program to generate high fidelity music convolved to express the reverberant characteristics of the selected captured space. All other features were secondary goals, and we met some, and are still developing others.

Primary Testing Goals Met:

- Selection of most flat-band microphone: DPA4011
- Placement of mic relative to point-source speaker; facing and directly inline with speaker cone, but across the room at preferred 'listening' position.
- Signal levels to avoid distortion due to overdrive and to read i/o signal with optimum SNR.
    1. of chirp track sent to Mackie mixer
    2. of level sent from mixer to the Electro-Voice PA
    3. of captured audio coming out of the DPA4011 mic
- File type to send to convolution: .wav
- Test Capture functions with four rooms of vastly differing reverberation characteristics
    1. Large reflective room- USF Concert hall
    2. Large absorptive room- Morrisound Live Room A
    3. Small reflective room- stairwell
    4. Small absorptive room- Morrisound Isolation booth
- Debugging of the code (C++/Python)
    1. ADAC between mic, Pi network and outputs
    2. To run SciPy convolution function in RPi
    3. To export convolved file out of Pi as a wave file
    4. To interface with LCD and pushbuttons
- Testing convolution program for fidelity with multiple music source types (drums, vocal, guitar) and multiple captured impulse responses.

- Safe, in-box power distribution to each system component

Secondary Goals Met:
- Pushbutton trigger of convolution with Preset capture
- Proof-of-concept pushbutton trigger of native 'record'
- HTML Sever proof-of-concept with LCD and Web Browser message upload

Secondary Goals Yet Unmet (due to programming and funding obstacles)
- Pushbutton triggered automation of capture and convolution and playback systems.
    1. Triggering 1kHz and chirp files
    2. Triggering 'record' function in DAW
    3. Sending complete recording from DAW to Pi
- Full Server user for upload/download of music files, and Wolfson Pi to Convolution Pi communication
- Fully functional Prototypes

V-IR-Bull Convolution Reverb Systems gives you the freedom to defy physical limitations and take your sound with you anywhere, anytime. Embrace your freedom today!

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FIGURES