THE VELVET CLOUD: INFINITE SUSTAIN AND MODULATED REVERBERATION USING VELVET NOISE

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ABSTRACT

In this project two digital audio effect plugins (VSTs) are proposed to simulate an automatic infinity sustain and reverb respectively. `Matlab` software is used for its development in real-time. Sustain audio effect algorithm is based on convolution with velvet noise. For reverberation, a Modulated Velvet Feedback Delay Networks implementation is developed. The use of velvet noise enhances reverberation with a quick built of early reflections in comparison with standard Feedback Delay Networks. The modulation approach produces interesting deep/psychedelic sounds which model unreal spaces. Both VSTs are thought to work together allowing the user to place them in her/his favourite configuration (e.g. Sustain in series with reverberation). A technical evaluation (impulse response, White Gaussian Noise response and T60 times) is performed regarding the different possible combinations of plugins. Also, a Two Alternative Forced Choice test and a survey about the VST performance and appearance are carried out for a more subjective sound and plugin evaluation. Results are that the developed plugins are unique, achieve a deep, psychedelic reverberation with micro-variations thanks to modulation with a background infinity sustain, over which musicians can play melodies on top. Both VSTs can be found in GitHub [1] and a video example is available on YouTube [1].

1. INTRODUCTION

Reverberation effect can be found in nearly every music or audio production. In contrast, little has been developed regarding digital sustain effect. The available current market VSTs will be consequently studied followed by State-Of-The-Art algorithm implementations regarding both audio effects:

1.1 Regarding available sustain VSTs

Yet, in the market, only a pair of sustain VST [2, 3] can be found. These VSTs are evaluated concluding that [3] "accumulative" infinite sustain saturates the incoming audio and [2] is not a real-time sustain in the sense of needing to load an audio file previously to be sustained.

On the other hand, digital and analog standalone guitar effects that can act also as a sustain [4, 5] can be located, although not many neither.

1.2 Regarding available reverberation VSTs

Due to the popularity of reverberation effect in the audio world, many more VSTs can be spotted.

The sound of the following ones are evaluated: ValhallaSuperMassive [6], NeoVerb [7], DreamScape [8], Eos2 [3], SpatialVerb [9] and FoGConvolver [10].

Best sounding ones (deep and large reverb) and also, the most expensive, are certainly [6], [7]. [6] is probably using Feedback Delay Networks (FDNs) algorithm as mentioned in their website [11].

An informal evaluation and more technical one consisting on measurements of T60 times (full spectrum, low pass filtered and high pass filtered) have influenced the reverberation implementation of this paper towards the election of FDNs as the main algorithm for recreating a wide reverberation.

1.3 Sustain algorithms

Unsurprisingly, not many developed algorithms can be found:

1.3.1 Synthesis by modulation and pitch estimation

In [12], sustain effect is recreated by using a sinusoidal model and partial amplitude modulation for the synthesis. The present latency makes the algorithm unsuitable for real time.

1.3.2 Additive and Granular Synthesis detecting a note onset

In [13], an additive and granular synthesis method when a note or chord is detected to be sustained is used. Unfortunately, the aforementioned effect can not be accessed and evaluated to analyze its performance.

1.3.3 Looping when detecting a note onset

In [14] an onset detection to look for new notes is utilized. Once they decay to a steady state, the note is looped until a new onset comes. Pitch detection is performed to get the period of the signal. The algorithm is often misidentifying the note by few Hz, which is audibly noticeable and unpleasing for a sustain.

[1] `https://github.com/chachipirulin/TheVelvetCloud`.

1.3.4 Convolution with Velvet Noise

In [15] the Overlap-And-Add method is implemented. To freeze the audio signal, an input grain is windowed and convolved with velvet noise accomplishing the illusion of sustain. The algorithm results are very outstanding as found in [16].

1.4 Reverberation algorithms

1.4.1 All-Pass filter and Comb filters

Introduced by Schroeder and Logan [17], is one of the earliest algorithms to tackle reverberation consisting on using all-pass filters. Later, Moorer enhanced this first reverberation stage researching example architectures with delay line and filter parameter values [18]. Although not really recent, some still use it for a real time implementation [19].

1.4.2 Convolution with Impulse Response

One of the most straightforward ways to implement reverberation is to convolve the audio signal with a specific impulse response. The operation is efficiently performed in the frequency domain, although, regarding its implementation in real-time, convolution works on big blocks of audio data which causes some undesirable latency [19].

Although, nowadays computer power capable of running without much artifacts long filters up to several seconds [20]. Some hybrids solutions [21, 22] have been found to enhance performance.

1.4.3 Ray-tracing technique

Also, find reverb effects whose impulse response is calculated as a result of using a ray-tracing technique [9] can be found. In the ray-tracing method, a fixed number of rays is emitted in various directions with equal angles spherically from a given source point. Useful method for for high-frequency sound propagation in complex architectures [23] rather than creating a profound reverb.

1.4.4 Feedback Delay Networks

Proposed in 1982 by Stautner and Puckette [24], Feedback Delay Networks (FDNs) is still one of most efficient reverb implementations. FDNs can be regarded as a series of delay lines interconnected in a feedback loop linked by a feedback matrix [19].

Variations to the original concept can be found, such as increasing the FDN order and therefore the number of delay lines [25], using a different reverberation time for each frequency band [26], among others. In [27], the user can control the FDN in real-time by adjusting its reverberation time in octave bands, the feedback matrix type and delay-line lengths. [28] proposes having control over an accurate graphic equalizer as the attenuation filter of the FDN to reproduce a reverb behaviour in a room. To understand and analyze more accurate how FDN models artificial reverberation, a modal decomposition is resolved in [29].

Circulant and Elliptic FDNs [30] have also been proposed, achieving more computational efficiency and versatility. Some of the drawbacks that FDNs present is the initial slow build-up of echoes. This is tackled in [31] by convolution with velvet-noise.

1.4.5 WaveGuide Networks

First introduced by J.O Smith [32], Wave Guide Networks (WGNs) model reverberation from a physical modeling point of view [25], following the scheme of interconnecting acoustic tubes. One remarkable point of WGNs is its correspondence to FDNs, as it can be used to obtain the FDN parameters based on the physics and geometry of a real acoustic space [33].

1.4.6 Other approaches

Really simple reverberations methods have been implemented too (but not recently). As an example, in [34], a really simple algorithm generates the reverberated samples by adding together several scaled and delayed input. Plane-wave synthesis method has been used in [35] to modify the spectral energy decay curve.

Other reverberation alternatives consider adaptive reverberation such as in [36], which in essence models a FDN depending on the audio harmonic characteristics. Also, in [37], it can be found algorithms for analog simulation of plate reverberation.

2. DECISION AND IMPLEMENTATION

S. D'Angelo [15] algorithm approach is the one selected as a basis for the sustain implementation based on the positive results it shows for an infinite sustain.

Regarding reverberation, the approach studied in [31] is used as a starting point to build a new variant of FDN; Modulated Velvet Feedback Delay Networks (MVFDN). This decision is based on the audible outcomes of the aforementioned work [38] which show a deep, unique and interesting reverberation compared to the other found reverberation works mentioned above.

In real time audio processing using `Matlab`, the incoming audio is processed in "chunks" of 1024 samples as a standard. Having that said, the algorithms for the digital sound processing of the VSTs are the following:

2.1 Sustain algorithm

The incoming input audio chunk will be used to fill up a convolving snippet buffer (it is the audio snippet to be sustained through time).

After this buffer is filled up and full, it is multiplied by a window and low-pass filtered. Consecutively, a set of positions of this buffer will be summed and multiplied by the correspondent signs (performing the convolution with velvet noise). The positions and sign values are given by the generated velvet noise.

The result is then multiplied by a gain (WET) and summed to the DRY signal. This constitutes the output of the system.

At some point when guitarist may strum the guitar, reaching a level higher than a threshold, a new snippet of audio will be selected and sustained through time.

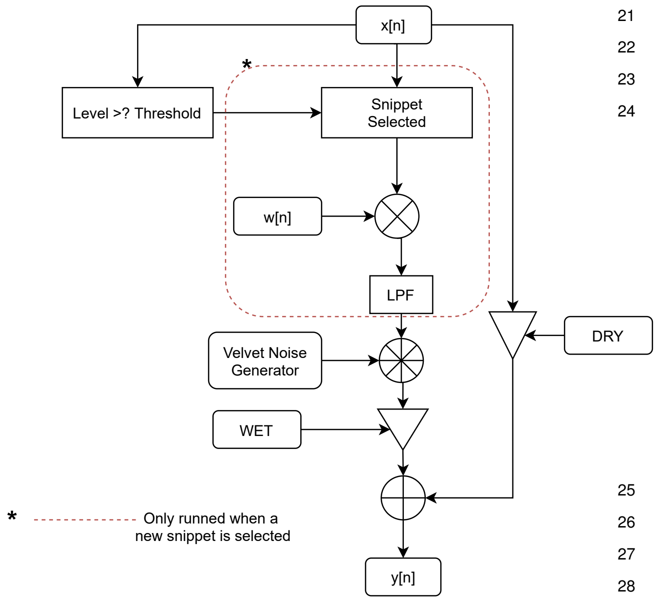

Figure 1. Flow diagram of sustain algorithm.

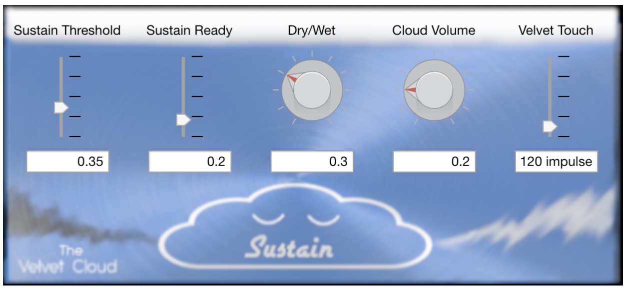

Figure 2. Sustain GUI.

A condition to be held before taking this new snippet of audio to sustain is that all values of an input chunk must had had a lower value than a second different threshold. See Fig. 1 for an overview of the algorithm.

2.1.1 *Graphical User interface*

The designed GUI (see Fig. 2) for the sustain effect has the following turnable parameters:

- **SustainThreshold:** when input signal SURPASSES this limit, input will be sustained.
- **SustainReady:** when input signal is BELOW this limit, ready to sustain again.
- **Cloud Volume:** gain for the sustained audio snippet.
- **Velvet Touch:** the number of impulses per second a sustained grain will have.

2.2 Reverberation effect: Modulated Velvet Feedback Delay Networks (MVFDN)

A standard FDN consists of a set of delay lines interconnected through a feedback matrix ***A*** forming a loop (that is why it is called "feedback" delay network) [26].

The feedback matrix (also called "scattering" matrix) defines the recirculating gains for each delay line. If this matrix is orthogonal, a lossless prototype is obtained, and the output of one delay is redistributed among to the input of all delay lines.

Equation 1 and 2 are of special interest because it models the output *y(n)* of the MVFDN, for an input *x(n)*.

$$y(n) = \sum_{i=1}^{N} c_i(z) s_i(n) \tag{1}$$

$$s_i(n + m_i + amp_i \cdot rate_i(n)) = \sum_{j=1}^{N} A_{ij} g_j s_j(n) + b_i x(n) \tag{2}$$

Where b_i and c_i are the input and output coefficients generated by velvet noise, respectively, A_{ij} is the feedback matrix element, g_i is the attenuation gain, and s_i are the output states of each delay line. Notice that delay lines delays are being modulated by amp_i and $rate_i(n)$.

Modulation of the reverberation is controlled by these both parameters, each per delay line number. See Fig. 5 for the VST GUI where the possible control parameters of the MVFDN can be tuned.

Equation 3 is of special interest because it models the transfer function of the MVFDN where *b* and *c* input and output gains vectors are replaced by $b_i(z)$ and $c_i(z)$ sparse Velvet Noise Sequence filters.

$$H(z) = \frac{Y(z)}{X(z)} = \mathbf{c}(z)^T(\mathbf{D}_m(z)^{-1} - \mathbf{A})^{-1}\mathbf{b}(z) \tag{3}$$

Where ***b*** and ***c*** are vectors containing the input and output gains, $\mathbf{D}m(z)$ = diag($\mathrm{G}_1(z)z^{m1}, G2(z)z^{m2}, ..., G_N(z)z^{m_N}$ are the feedback gains and $\mathbf{A}$ is the feedback delay matrix.

See Fig. 3 for an overview of the algorithm.

2.2.1 *Graphical User interface*

The designed simplified GUI (see Fig. 4) for the reverberation effect has the following turnable parameters:

- **Cloud Height:** toggles between high feedback gains values (heaven) and standard values (sky).
- **Cloud Brightness:** selects a modulation preset. There are five different options of modulation (no modulation, light, medium, high, ultrahigh and extreme).
- **Cloud Density:** standard dry/wet knob.
- **Output Gain:** to amplify output.

3. DESIGN CONSIDERATIONS

There are some design decisions specified previously without further explanation. Thereupon, some clarifications follow:

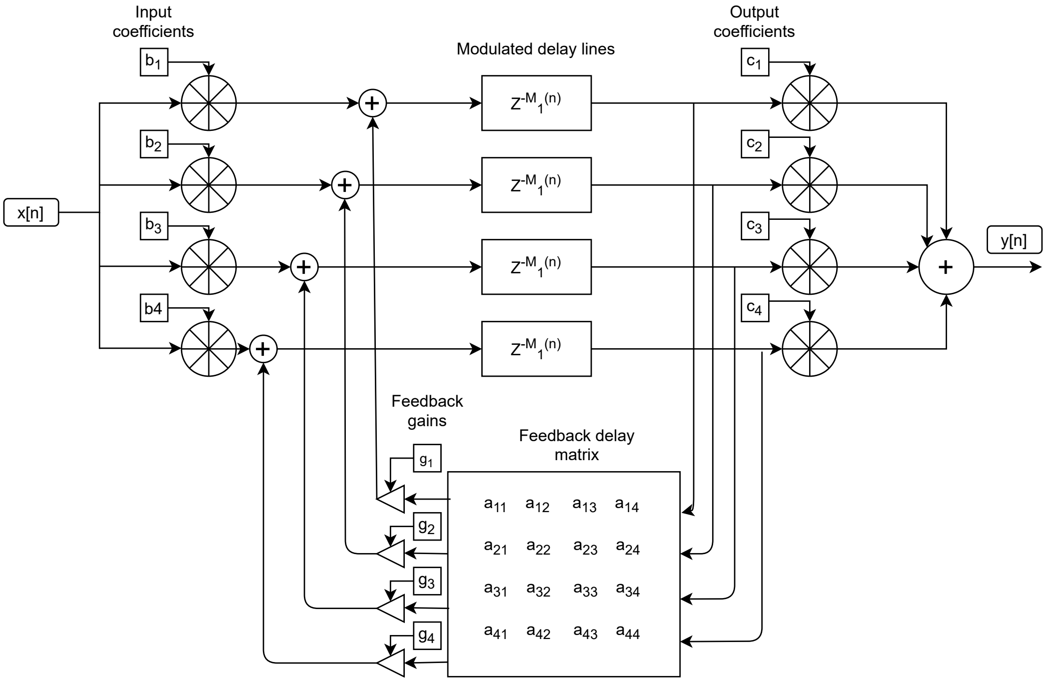

Figure 3. Flow diagram of MVFDNs of order 4 algorithm.

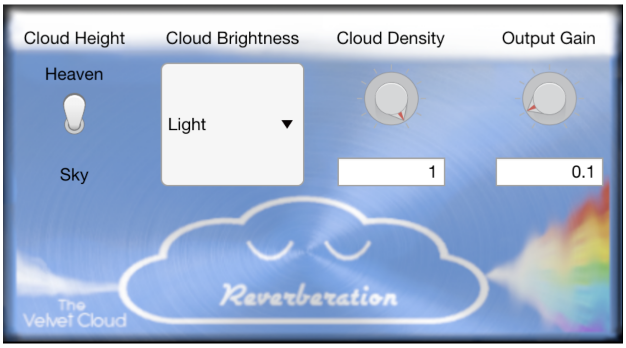

Figure 4. Reverberation GUI (simplified).

3.1 Sustain

3.1.1 *Buffer size*

The buffer size for holding the audio snippet to be convolved is 30ms long. Several values have been used for the length of this snippet, resulting in 30ms as a good value to work with low computational demand and nice sound. This is a little smaller value than the recommended value in [15].

3.1.2 *Low pass filtering*

If a guitar is once strummed, if the first audio cues are the ones to be sustained, it will be resulting in a striking sound, as the sustained audio is filled with lots of different frequencies of nearly the whole spectrum. Rather than that, we would like not to sustain the first cue of the strum, but rather, the "tail" or where the sounds tends to fade off, as it is a more pleasant sound to hear through the time, with less amplitude and mainly low frequency content. That is the reasoning by using a low pass filter.

3.2 Reverberation

3.2.1 *Choice of delay lines*

Although there is not much consensus about the delay line values [39], their values, as suggested by Schroeder and in [40, p. 111], are chosen to be be mutually prime. These values are chosen randomly among prime numbers keeping a minimum distance of 400 samples and are uniformly distributed.

3.2.2 *Choice of feedback matrix*

Inside unilossless matrices, we can find different matrix types: Hadamard [41], Householder [41], identity matrices [42] and orthogonal. Hadamard is finally chosen as it turns out to prolong more the reverberation in MVFDN.

3.2.3 *Amplitudes and rates values for modulation*

At some amplitude and rates values, the MVFDN behaves as a flanger or chorus, which is not the expected comportment of a reverb effect. Then, using informal hearing, the highest values (just before turning the MVFDN into a flanger) of amplitude and rates are selected. These highest values conform the "*Extreme*" mode for the reverb plugin while the slightest corresponds to "*Light*" (see Fig. 4 noticing "Cloud Brightness" drop-down list). Convolution with velvet noise is performed in the input and in the output of the MVFDN. This increases echo density in the beginning of the impulse response with low computational cost, with better results (faster echo density growth) than doubling the number of used delay lines [31].

3.2.4 *Using a decaying velvet noise*

Standard velvet noise cause frequency coloration in comparison of using a decaying velvet noise sequence which

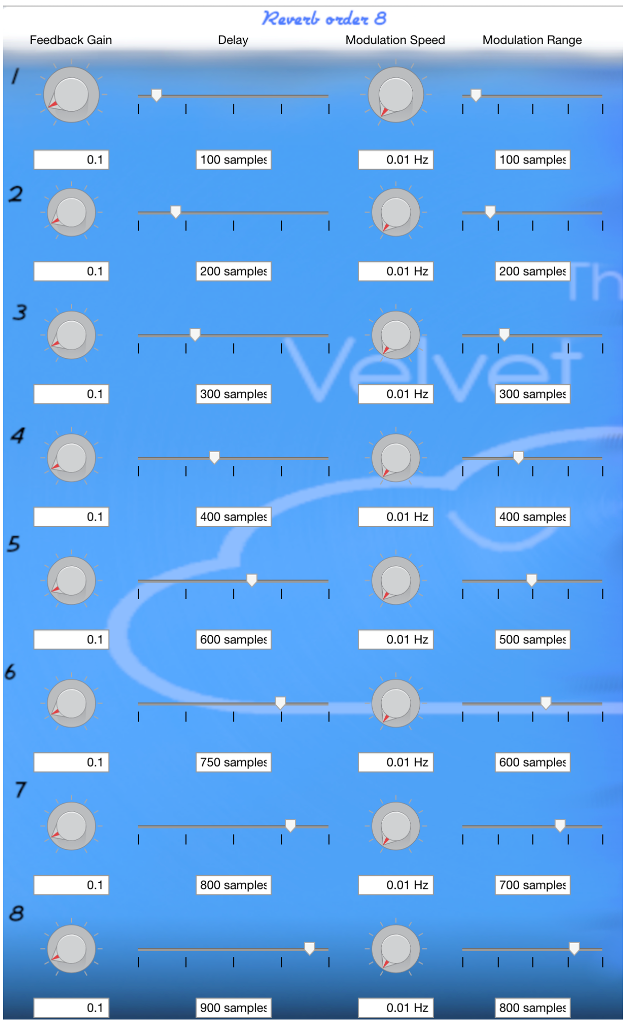

Figure 5. GUI version for full control of the MVFDN of order 8.

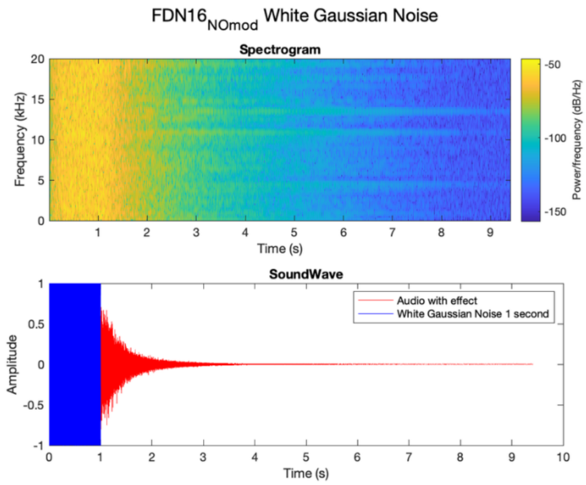

Figure 6. FDN16 WGN response without modulation.

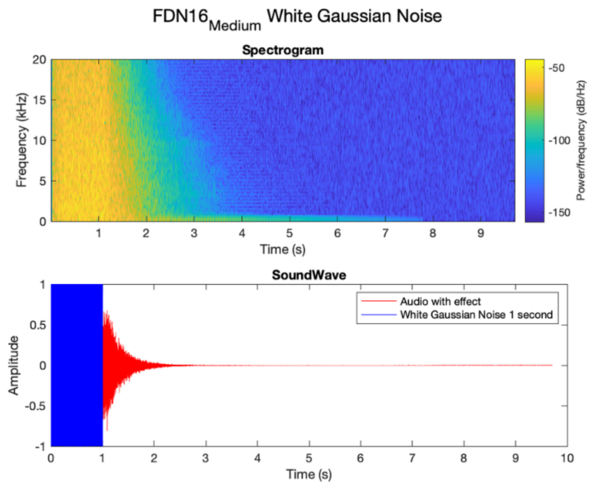

Figure 7. FDN16 WGN response for high modulation.

have a more flat spectrum [31]. Also, output is more pleasant confirmed by an informal listening by the author.

4. EVALUATIONS AND RESULTS

Three different evaluations are carried to assess this work:

4.1 Technical evaluation

Impulse response and White Gaussian Noise response are calculated over sustain and reverberation (for MVFDNs of order 4, 8 and 16). Additionally, T60 times (full spectra, filtered and HP filtered) are measured.

Regarding the sustain effect, its impulse response is silence as expected. The WGN response is the presumed one; a reduction of the amplitude filtered by the low pass filter and a sustained sound in time indefinitely.

Respecting the reverberation effect, T60 times are not great compared to what achieved by [6] or [7] measured in 1.2. Order 8 gets higher T60 times obtained in order 4, and order 16 get briefly greater order 8 T60 times.

When it comes to WGN responses, versus a WGN response without any modulation (see Fig. 6), the more modulation the reverberation has, the more scattering of the frequencies in the spectra is produced. Frequencies tend to accumulated in the low range (see Fig. 7).

4.2 Two-alternative forced choice (2AFC) test

This test is carried to measure the subjective experience of a person of how pleasant or how she/he likes/prefers a sound (some guitar notes) in comparison with another. The question asked is: *"Which sound do you prefer/like the most?"*. 7 people with musical experience (mean of 4 years of musical practice) from 23 to 35 years old participated in the survey.

Sound with reverberation of order 4 with no modulation is preferred over any order 16 possibility. Comments are that order 16 sound is too wide and bad-defined. It is observed that order 16 is preferred over order 8 with no modulation.

4.3 VST performance and appearance survey

This test is made for people who have had previous experience with VSTs, to evaluate the plugins appearance and performance. In total, 15 people (1 females and 14 males) participated in the survey. Questions are about VSTs installation, GUI, performance, pros and cons, enhancement's, suggestions and sending a screenshot with their favourite plugins configuration.

Some of the comments are: *""I liked the simple design and the reduced amount of controls", "It would be nice to have both effect integrated within the same GUI", "didn't ever "explode" or go into unstable", "had some clicking", "It's definitely crazy, but in a fun way"*. People thought (as mean) TVC is worth price of 40 dollars and all participants chosed to have some modulation (from the different modulation presets 2.2.1 that the reverberation VST allows) in the reverberation configuration.

5. DISCUSSION

The use of modulation in the MVFDN tends to disperse the energy of the sound wave making the amplitude of the reverberated tail smaller, concentrated on the low frequencies 4.1. It act similarly as a low pass filter would perform looking at the spectrogram (see Fig. 7), nevertheless, using informal hearing, it sounds completely different from just a low pass filtering [1]. T60 times are relative attenuated when higher modulated, which holds as modulation is acting as a filter in essence.

Modulation caused pleasant sensations in the hearing as showed in the 2AFC test 4.2. In the test there are comments of type: "*I prefer this because is brighter, reverberating and wider*", versus comments of type: *"The other sound is really bad-defined and too blurry"*. As there is not a clear statement about which criteria to follow as a subjective question is made, results are maybe biased and not confirming clearly which sound sounds with more reverb, but rather which is more pleasant, which means different things for each person.

Having this said, if MVFDN order 4 versus order 8 is compared, in general, a modulated sound is preferred over no modulation at all, which could indicate that modulation adds a more "pleasant" sound to reverberation. Order 16 is less liked in general compared to other orders due to its "too much" reverberation, which can mean reverberation is pleasant until some point, when it gets really "wet" or "crowded".

With respect to the VSTs evaluation 4.3, it can be agreed that modulation is preferred over no modulation and chosen by every participants, which confirms also the comments participants stated about modulation in 4.2. Participants consider plugins attractive with a simple GUI, achieving a deep/psychedelic (thanks to the modulation possibilities) and really wet reverb, which recreates an unreal spaces and over which you can play melodies thanks to the infinity automatic sustain.

It is clear from the aforementioned that modulation achieves a pleasing tone and produces an enjoyable sound within the reverberation effect confirmed by the exposed results.

6. CONCLUSION

In this work, two different real-time plugins are presented. After reviewing the State-Of-The-Art in sustain and reverb audio effects and investigating the different plugins the market offer for sustain and reverb effect 1, a technique (as a starting point) is developed and enhanced for each effect: Convolution with velvet noise 2.1 for sustain and Velvet Feedback Delay Networks 2.2 for reverberation. As for reverberation, a new modality not present in The-State-Of-The-Art is proposed in this project: Modulated Feedback Delay Networks. The aforementioned has been implemented as two different VSTs which are found to have a deep, psychedelic and more interesting sound than usual reverb thanks to modulation with an unique and suggestive GUI 4.3. A technical evaluation 4.1 is performed over the VSTs possibilities as well as a Two Alternative Forced Choice test 4.2 to evaluate how its configuration affects sound subjectively, concluding that modulation adds an interesting enjoyable effect than rather no using modulation at all.

7. FUTURE WORK

A more profound analysis of how modulation configures the output of the system and how it influences people rating reverberated sound is left for future work. Additionally, further possibilities of modulation are to be explored, such as how different modulation wave-forms affect reverb in a technical and a subjective way. Also, suggestions by participants are to be implemented, such as making an all-in-one VST, solving the "crackling" when changing VST parameters or a button to trigger sustain or visual animations of the GUI. Besides, although working properly, an implementation in a lower language level could improve the CPU consumption of the VSTs.

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