DIGITAL ROTARY LOUDSPEAKER EFFECTS UNIT

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Rotary Loudspeakers. They have such a distinct sound, due to many complicated effects created by the rotating speakers and the housing they are placed within. They are sought after for many different applications. However there are a few problems. Firstly they are heavy, carrying a large wooden box can be hard and is not always possible. Secondly though people use this type of effect for many different things, from guitar to vocals. They usually only have a jack input capable of handling only one source at a time. So why should a performer have to carry a Leslie cabinet around with them? Why should they be restricted to a single input source?

I am proposing to create a digital DSP based rotary loudspeaker effects unit, that is easy to transport and is able to have many different kinds of instruments run through it, from keyboards to guitars to vocals.

By implementing a real time algorithm through this DSP based rack unit, analogue sound sources will be able to be inserted through the unit and converted into a digital signal. A convolution then takes place and an analogue output will be able to be taken to an amplifier or speaker system.

The unit will need both A/D and D/A conversion within it to be able take an analogue signal, convert it to the digit realm and then once the effect has been added return an analogue signal to the output. As well as the standard effect being sent to the output, creating a blend control the user will be able dial in the percentage of the effect they wish to have blended with their original signal.

The Convolution will involve two feed forward loops running in stereo. Taking a mono audio source splitting it into a stereophonic image, adding the rotating effect to both signals and finally being multiplied with the original signal.

The algorithm used to create this effect must include the following: Delay line modulation and amplitude Modulation to simulate the effect, as shown in figure 1 below.

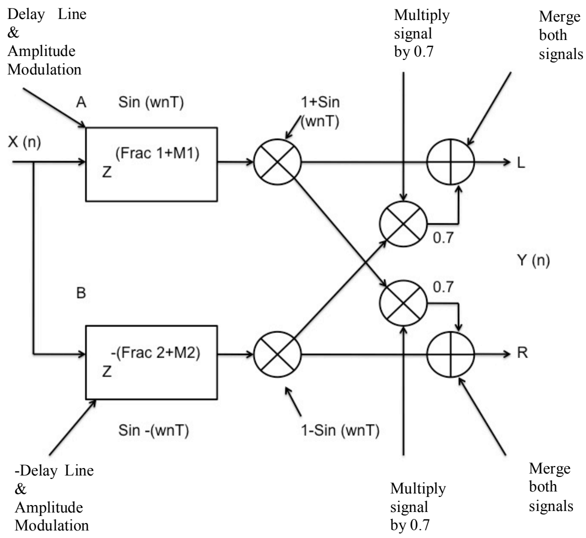

Figure 1.Simulation of rotating speakers [1]

Both the delay line and amplitude modulation are used to simulate the Doppler shift (fig. 2). This effect is carried out from the two rotating horns spinning within the cabinet. By adding a low frequency oscillator the delay line modulation is created, emulating the pitch modifications that the Doppler shift creates. Using the amplitude modification with this controls the intensity modifications (the rate at which the speakers spin).

$$\omega_l = \omega_s \frac{1+\frac{vls}{c}}{1-\frac{vs,l}{c}}$$

Figure 2. Doppler Shift. [2]

Both delay line and amplitude modulation have been added to a left and right channel. These are convoluted with the original signal to create a standard rotary speaker effect, however to make a more realistic effect a Chorus needs to be added.

The reason for chorus is despite the fact both delay line and amplitude modulation create a realistic Doppler shift. A Leslie has a cabinet surrounding the rotating speakers. As the horns spin within the cabinet the sound waves emitted from them reflect of the surface of the cabinet. This means the reflections experience their own Doppler shift. To be able to simulate each Doppler shift for each reflection would be very CPU intensive, so adding a simple chorus to the already created Doppler effect gives the listener the impression of many different reflections.

Once this simulation has been created, the next things to think about are the speeds at which a Leslie is able to rotate.

Typically there are two speeds, choral and tremolo. This effect will stay true to the original and apply both of these. A ramp signal is needed and must be applied to a voltage-controlled oscillator (VCO). To simulate this correctly the signal will split and send low frequencies through one VCO and highs through another then merge them back together. This allows the signal to feel like it has both the rotating horns and the bass drum. All this is shown in figure 3, below.

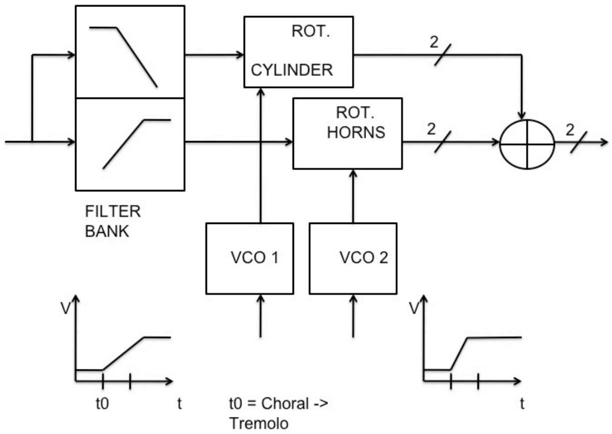

Figure 3. Rotary Speaker Simulation [1]

The next and final step is to implement a blend control, allowing the original sound source to be blended with the rotary speaker effect. The way in which to go about this is by adding the original sound source with the effected signal in steps of ten. Meaning the blend control will have eleven steps on in starting at zero then to ten and going to one hundred. Each step is adding a further ten percent to the original signal, until at hundred percent there is only the modified signal.

This algorithm will be carefully evaluated with tests beginning with non real time code, then implementing it into a real time DSP. To demonstrate the way in which this algorithm can be run a Matlab code has been created and the program has been run. The Matlab code takes everything that has been mentioned in the above proposal, chorus, delay line and amplitude modulation. It can be found at [3].

Attached to the referenced report [3], there are two audio files ‘SmoothPiano’ and ‘rotating_speaker_effect.’ The ‘smoothpiano’ is a dry piano signal and the “rotating_speaker_effect’ is the signal with one hundred percent rotary loudspeaker effect. You can clearly hear the difference and the likeness to an original Leslie cabinet.

So far I have talked about creating a digital DSP effect. I began the proposal by mentioning that the final product would become a rack-mounted unit, suitable for consumer use. I would like to validate my reasons for this. Firstly the two I mentioned earlier, carrying a small rack unit is much easier to transport than a large wooden box.

The fact that the unit will be capable of handling multiple input sources may seem like it will be more memory intensive, however as I mentioned the major convolution will take place as a feed forward loop, meaning that FIR’s are being used. These are simple to implement on most DSP microprocessors, so they will not be taking up large amounts of space.

I believe this product will not only create new and interesting techniques in audio, but will also be viable from a business point of view. This in my opinion is due to the popularity of the original sound in the current market. Creating a DSP based algorithm allows for the transfer of the effect into not only a rack effect but also computer-based systems and anything else capable of handling DSP processors. The versatility of the effect makes it useful and marketable for many different types of audio, and is an effect that has become more and more popular in recent years. In my proposal I have shown the specification of the effect how to implement it, and the reasons behind my choice.

BIBLIOGRAPHY

[1] S. Disch and U. Zolzer, “modulation and delay line based digital audio effects,” in 2nd cost G-6 workshop on digital audio effects, Trondheim, 1999, pp. 3-4.

[2] Julius O. Smith III, Stefania Serafin, Jonathan Abel and Dave Berners, “Doppler Simulation and the Leslie,” Center for Computer Research in Music and Acoustics., Stanford Univ., Stanford, CA, 2006.

[3] R. Brennan, “Lab Report 2 Rotary Loudspeaker,” Faculty of Architecture and Design Science in Digital Audio Systems.. Sydney Univ., Sydney, Australia, 2012.