Rotary Circus: A Leslie Speaker Simulator

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***Abstract*—*Rotary Circus* is a Leslie speaker simulator plug-in. It was developed using RackAFX and is VST compatible. The main idea was to make a simplified physical model of a Leslie speaker that captures its most prominent characteristics. The treble horn is simulated by means of a modulated delay line, amplitude modulation, and a modulated low pass filter. The bass speaker is simulated using amplitude modulation. The rotation speed of both treble and bass can be independently controlled with adjustable acceleration and deceleration.**

I. Introduction

One of the most distinctive sounds of popular music in the last 50-plus years has been the sound of the Hammond B3 organ. From lush, atmospheric textures, to screaming roars, the instrument has a wide range of expressive possibilities. Part of what makes the sound so unique, and so expressive, is that it is usually played through a rotating speaker, the most famous being those made by the Leslie company. For convenience we will refer to this speaker simply as ”the Leslie speaker.”

Rotary Circus is a plug-in that simulates the sound of the Leslie Speaker using digital signal processing. While this plug-in is by no means a perfect re-creation of the original, I have tried to capture the most prominent features of the sound while retaining a relatively low complexity.

The great majority of the technical information I needed to create this plug-in was conveniently found in an article by Clifford Henricksen [1]. The information in this article pertains to Leslie Models 122, 145, and 147. These models all share similar features: a single-channel 40-watt tube amplifier, an 800 Hz passive crossover, a rotating treble horn, and a bass speaker that fires downward into a rotating drum-shaped baffle (see Fig. 1).

II. Modeling Techniques

I will not get too far into historical or background information on the Leslie speaker, as this is copiously available on the Web. Instead, I will discuss the major features of the ”Leslie sound” and how I emulated these using DSP techniques.

The first feature to simulate is the tube amplifier which powers the speakers. The saturation of this amplifier is an important part of the sound, so it is simulated using a wave-shaper based on the equation (from [2]):

$$y(n) = \frac{1}{\arctan(k)} \arctan(k \cdot x(n)) \tag{1}$$

In this equation, the amount of distortion can easily be altered by changing the value of k. Since real tube amplifiers exhibit asymmetric clipping, the incoming signal is actually divided into a positive portion and a negative portion. The negative portion of the waveform is processed with a higher value of k than the positive part.

The rotating treble horn is by-and-large the essence of the Leslie sound. The basic idea is that the sound is reproduced from a compression driver connected to a horn that spins in a circle. As the horn moves closer or further away from the microphone (in this case a simulated microphone), there are three things that happen.

1) Due to the motion of the horn, there is a small Doppler shift in the pitch. As the horn is moving towards the microphone, the pitch rises, and as the horn moves away, the pitch falls. In musical terms, this creates frequency modulation, or vibrato.
2) Due to the narrow directivity of the horn, and the changing distance between the horn and the microphone, the loudness of the sound changes with time. This creates an amplitude modulation, or tremolo effect.
3) Also due to the directivity of the horn, the tonal quality, or timbre, changes as the horn alternately points towards or away from the microphone. When the horn is pointing away, the high frequencies will be muffled, as they are much more directional than the low frequencies.

It should be mentioned that while the treble rotor has two horns facing in opposite directions, only one of them produces sound. The other is simply a counter balance to keep the whole assembly from wobbling as it spins.

Fig. 1. The back side of a Leslie Model 147 speaker.

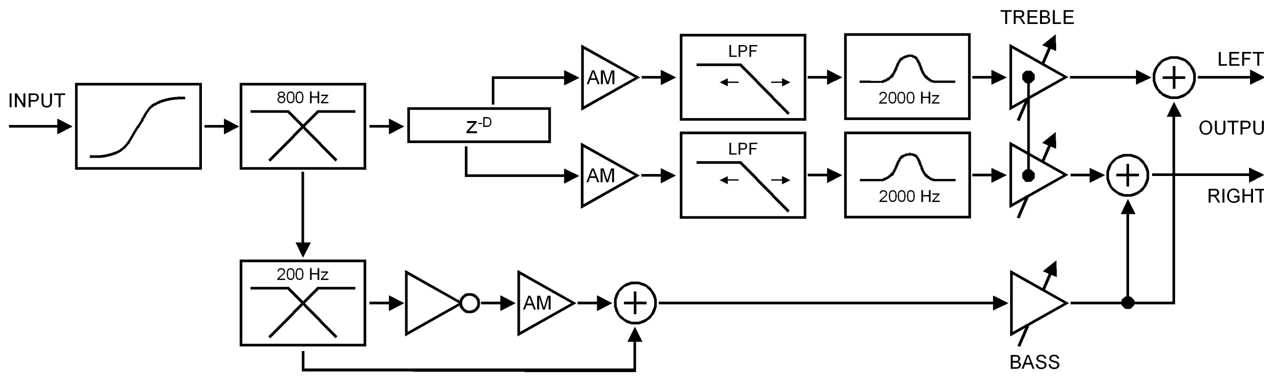

Fig. 2. Block diagram of the Rotary Circus algorithm.

A complete physical model of the speaker and all possible microphone positions would have been difficult to design, and computationally expensive to implement, so I made a few simplifications. The pick-up of sound is assumed to be two virtual microphones placed at the corners of a square which inscribes the path of the rotating horn (as shown in Fig. 3). These microphones are assumed to be omnidirectional with flat frequency response and are placed in anechoic surroundings (not very realistic, but it saves quite a bit of additional DSP). The three features described above were simulated, respectively, by the following methods:

1) A circular delay buffer is used to simulate the propagation time from the mouth of the horn to the left and right microphones. As the delay time shortens and lengthens, the Doppler effect will happen automatically. Only one delay line is needed since there is only one source. The left and right signals each have their own read index which moves independently through the buffer.
2) The relative distance between the horn and microphones is used as the basis for the amplitude (tremolo) effect. This will be applied as a simple modulated gain.
3) A second order Butterworth low-pass filter with changing cutoff frequency is used to simulate the tonal modulation effect.

In addition to these dynamic elements, it is also important to emulate the static frequency response of the horn. The real Leslie speaker exhibits a strong band-pass characteristic centered at 2 kHz [1]. I chose to simulate this using a peaking filter with a center frequency of 2 kHz and gain of 10 dB.

The crossover between the treble and bass speakers is a passive circuit with a crossover frequency of 800 Hz. I chose to implement this using a second order Linkwitz-Riley crossover (design equations from [2]).

In a real Leslie speaker, the bass driver fires downward into a scoop-shaped baffle that is mounted inside a rotating drum. When the scoop is facing out toward the open side of the cabinet, the sound is un-inhibited, however when the scoop is facing in toward the closed side the cabinet, the sound is muffled, and reduced in volume. The modulation effect of the rotating baffle is simulated using a tremolo effect with an envelope that has a sinusoidal shape in decibels. However, not all frequencies are affected equally by the baffle. Frequencies below 200 Hz are likely unaffected because of their long wavelength and acoustic energy. Thus, there is an additional Linkwitz-Riley crossover to separate the modulated bass from the un-modulated bass. Because of the phase response of the crossover, the higher, modulated signal is inverted to assure the correct frequency response when the signals are re-combined.

One of the advantages of having a simulated device is that you can adjust parameters that are fixed on the real thing. A

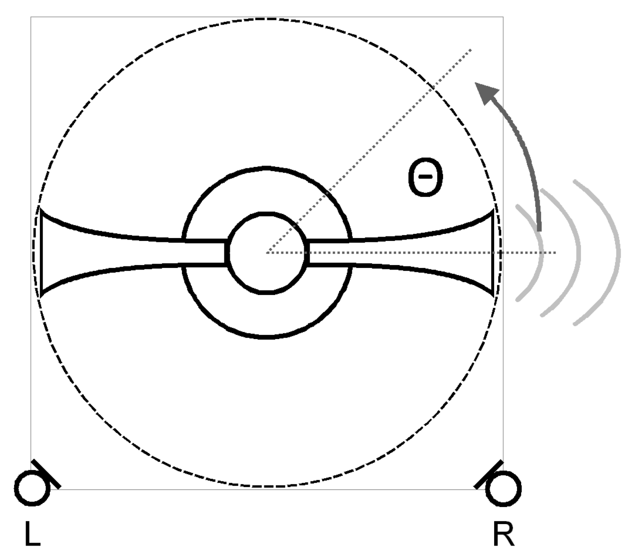

Fig. 3. Geometry of simulated rotating horn and microphones. Zero-degree axis points to the right.

convenience feature I have included in the plug-in is the ability to balance the overall levels of the bass and treble signals before they are combined into the final stereo output. Refer to the block diagram in Fig. 2 to see the algorithm as a whole.

III. Geometry Calculation

At any instance in time, there are two values that must be computed for each virtual microphone: the distance between the mouth of the horn and the microphone, and the angle between the same. Fig. 3 shows the geometry used to solve for the distance and angle, the 0 degree axis points to the right. This distance can be calculated by the following equations:

$$x_L = r_h + r_h \cdot \cos\theta \tag{2}$$

$$x_R = r_h - r_h \cdot \cos\theta \tag{3}$$

$$y = r_h + r_h \cdot \sin\theta \tag{4}$$

$$d_L = \sqrt{x_L^2 + y^2} \tag{5}$$

$$d_R = \sqrt{x_R^2 + y^2} \tag{6}$$

Where θ is the rotation angle, x_L and x_R represent the x distance to each microphone, r_h is the radius of the horn, and d_L and d_R represent the total distance to each microphone. The distance is the basis of the frequency and amplitude modulation (see Fig. 4).

When the horn is pointed directly at the microphone the angle between them is 0 degrees, and when pointed directly away, this angle is 180 degrees. The left microphone has a ”rotation angle” of -135 degrees, and the right microphone has a ”rotation angle” of -45 degrees. The angle between horn and microphone is the basis of the tonal modulation (see Fig. 5).

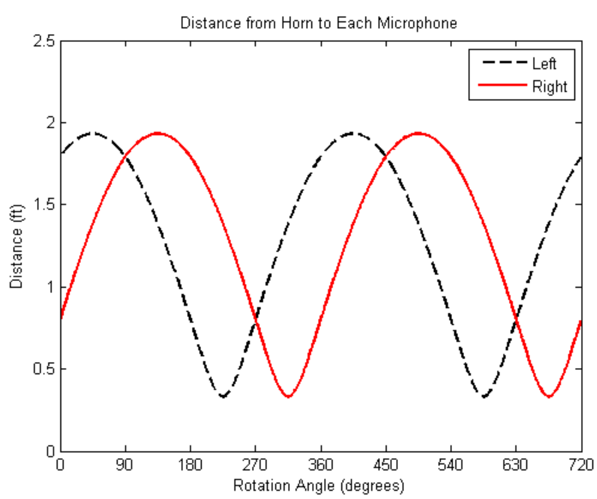

Fig. 4. Distance between the treble horn and each microphone, with respect to rotation angle.

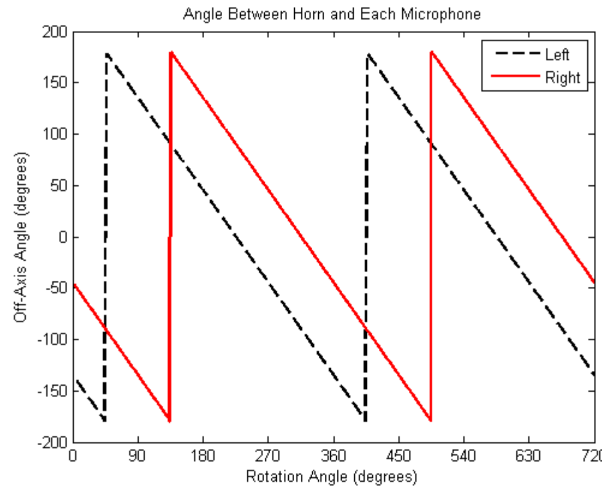

Fig. 5. Angle between treble horn and each microphone, with respect to rotation angle.

IV. Motor Control

In addition, one of the distinctive characteristics of the Leslie speaker is that it has both a fast and a slow speed. The change in sound as it accelerates or decelerates is a much-loved expressive feature. The plug-in has an auto speed control mode that allows the user to choose between ”Off”, ”Slow”, and ”Fast” for the desired speed. The plug-in will accelerate and decelerate the rotation according to parameters set by the user. Once the target rotation speed is reached, it will remain constant until another speed is selected. The treble and bass rotor are controlled separately so that they can have separate acceleration/deceleration speeds and are not synchronized.

V. Development Process

Rotary Circus began as an idea of wanting to use the geometry of a rotating horn as the basis for an effect. Some of the initial concepts were developed in Matlab, but otherwise everything was written in C++. The plug-in was developed using the RackAFX Plug-In Design Lab [2] which simplifies most of the logistical programming and allows the developer to concentrate on the algorithm.

The parameters for the plug-in are as follows:

- Input Drive (distortion level)
- Treble Rotational Speed (in RPM)
- Treble Horn Radius (in feet)
- Treble Amplitude Modulation Depth
- LPF Modulation Depth (in octaves)
- LPF Center-Cutoff Frequency
- Treble Horn Resonance Center Frequency
- Treble Horn Resonance Gain
- Bass Rotational Speed (in RPM)
- Bass Amplitude Modulation Depth
- Bass to Treble Crossover Frequency
- Overall Treble Volume
- Overall Bass Volume

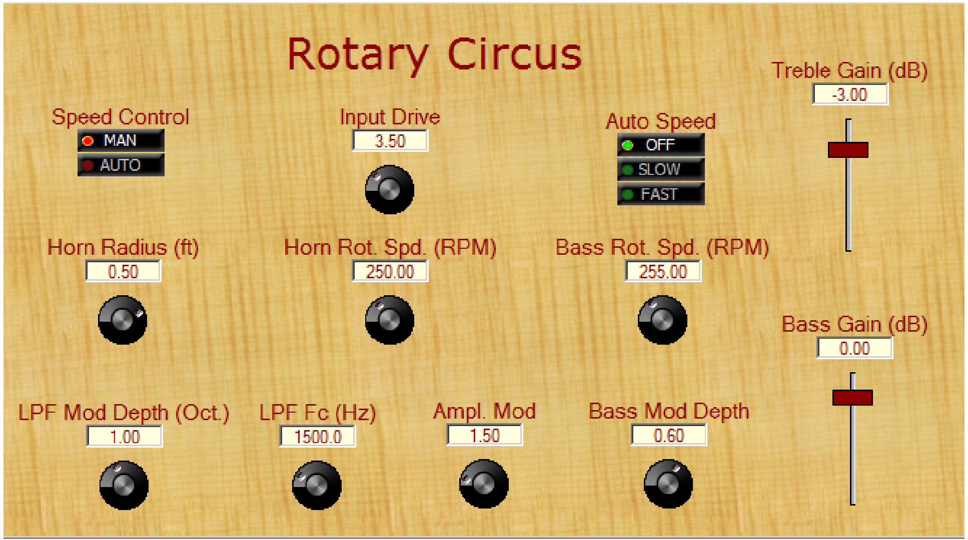

Fig. 6. Screen shot of the user interface.

In addition, if the user selects automatic speed control, the manual rotational speed controls for bass and treble are disabled, and instead the user can adjust the acceleration and deceleration time (in seconds) for the bass and treble rotation.

Several parameters that were available to the user in the prototype environment are left out of the final GUI in the interest of a simpler interface. These include the controls over the acceleration/deceleration time, the cross-over frequency, and the gain and center frequency of the peaking filter for the horn. Fig. 6 shows a screen shot of the final plug-in.

VI. Future Improvements

Unfortunately I did not have access to a real Leslie speaker in developing this plug-in, so that is obviously the biggest issue in making further improvements. Many of the parameters I simply adjusted based on what sounded right to me. It would be invaluable to do a side by side comparison and adjust parameters accordingly.

The distortion produced by the wave shaper is an important part of the sound, but it can also be problematic. Since it is a non-linear process, it adds additional (higher) frequencies to the signal coming in. This has the potential to cause aliasing, which is a very unpleasant sound. Ideally, oversampling should be used to eliminate this possibility. Fortunately, in the intended usage of this plug-in, the problem is not so readily noticeable as tone-wheel organ sounds often lack strong high frequency components and distortion levels are kept to a moderate level.

VII. Conclusion

All things considered, *Rotary Circus* provides a remarkable degree of realism while not placing an undue burden on the host computer. Every step of the process was done with the intent of recreating the character of the original, while at the same time giving the user flexibility to experiment with new sounds. The many parameters available mean that the user can dial in very subtle or very extreme effects. To my ears, the right combination of settings result in a sound that is very close to a well recorded Leslie speaker.

References

[1] C. Henricksen. Unearthing the mysteries of the leslie cabinet. *Recording Engineer/Producer*, April, 1981.

[2] W. Pirkle. *Designing Audio Effect Plug-Ins in C++: With Digital Audio Signal Processing Theory*. Focal Press, 2013.