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The Solid-State "Rotary" Speaker

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This article presents the proof of concept for a novel solid-state "rotary" speaker. The device uses a circular array of speakers instead of moving parts to achieve sound modulation. For this, the signal is translated across the array electronically rather than mechanically. Inspired by the original Leslie speaker, this novel concept represents a new paradigm in instrument amplification and sound effects, capable of modulating physical sound waves outside the electronic and digital signal chains and without any moving parts. The authors present the working principle of their device and its technical implementation and demonstrate functionality using test and instrument signals.

Keywords: Array Speakers, Leslie Speaker, Sound Effects

1 INTRODUCTION

In 1935, the electric organ was first commercialized by Hammond Organ Company. It was introduced into the market as an alternative to the more expensive pipe organs used in churches at the time [1]. However, it had a crucial difference: in an electric organ, all notes originated from a single speaker, whereas in a pipe organ, the different notes would be generated on pipes separated from each other. Therefore, the different notes would be slightly out of phase, generating a beating effect [2].

To emulate this effect on the electric organ, Donald Leslie invented the rotary speaker (also commonly known as a Leslie speaker), which modulates a sound source by rotating a horn and baffle around a stationary loudspeaker ([3]). In simple terms, the rotating elements continuously change the path that some of the sound is taking with respect to the directly propagating sound. This, along with a subtle Doppler shift, generates a sound that resembles the beating of a pipe organ. The Leslie was first manufactured commercially in 1941 (with a US patent granted in 1962 [3]) and is still being produced today, with its fundamentals remaining the same for over 80 years.

Although the Hammond organ with a Leslie speaker was successful as an alternative to pipe organs, its impact was most profoundly felt in popular culture. The Hammond and Leslie combo was an integral part of many music genres, including gospel, soul, blues, jazz, and rock throughout the 1950s through the 1970s [1] and is still in use today [4]. A version of rotary speakers was also manufactured for guitars by the Fender company and used in many now-classic pop culture albums [5].

The speaker, however, is not without its challenges: It is expensive compared to a regular instrument amplifier (∼3k GBP [6]), is also significantly heavier and bulkier, and requires significant maintenance due to the electric motors and belts required to drive motion. Moreover, the noise of the motors makes it difficult to play at low volumes. Finally, the speaker itself is not versatile enough to allow for variations beyond limited change to the speed of rotation. These challenges inspired electronic emulations of the rotary speaker sound, which can be found today in sound effects such as "phaser," "chorus," and others [7, 8]. These electronic emulations are widely available today in digital and analogue form and are far more convenient [9].

Although modulation effects have been tremendously successful, they might not replicate the Leslie sound accurately. Efforts for better emulation have been reported on several fronts. For instance, in the use of moving sources [10], the inclusion of Doppler effects [11] or modeling separate baffle and horn dynamics [12]. The contributions of the two moving sources (baffle and horn), the change of direction the listener perceives, and the reverberations of all these elements in the room, make the Leslie a very complex system to emulate. Today, a broad choice of complex emulations is commercially available in the form of plug-ins [13] and digital effects [14].

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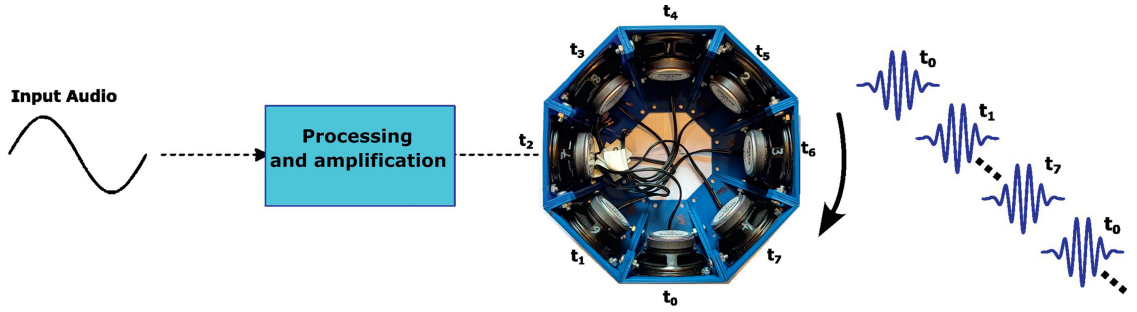


Fig. 1. Concept of the solid-state Leslie speaker. (a) An input signal is split and amplified so it can be cross-faded on a circular array of speakers in sequence (see arrow) to emulate the function of a rotary speaker. (b) The outputs are windowed, providing smooth transitions between the different speakers. The center of the window for each speaker is sequentially delayed to emulate circular movement. The listener would perceive the modulation in phase originating from the interference between directly constant and moving outputs.

In the context of instrument amplification, the electronic counterparts to the original Leslie remain unable to faithfully replicate a rotary speaker's sound. In it, the sound modulation occurs physically in the sound waves surrounding the listener rather than electronically [11]. The rotational motion provides directionality of the sound source, which is not achievable in mono or stereo systems [10]. This directionality gives the Leslie speaker a much richer and organic sound, particularly when used in a venue or hall with no additional amplification.

Since the development of the Leslie speaker in the 1940s, electronic components, loudspeakers, and amplifiers have increased in efficiency and reduced in size and cost [15]. Today, an amplified speaker as loud as a Leslie speaker would cost a fraction of the price and fit in a significantly smaller form factor. These advantages have seen the rise of speaker arrays in what is normally referred to as 3D audio [16–18]. This new technology uses several sources to achieve directionality that is not possible with mono or stereo systems [19, 20]. However, such systems have not yet been implemented in musical instrument amplification.

This engineering brief presents a new paradigm for the design of a rotary speaker without moving parts, hence *solid-state*. In the methodology, the equivalent of rotation is achieved through a circular array of speakers, in which the signal is digitally windowed sequentially across the array to provide a smooth transition between sources. This represents the first proof of concept for means of modulating the phase and amplitude of acoustic waves similar to mechanical effects but in a solid-state format compatible with modern instrument amplification.

2 WORKING PRINCIPLE AND IMPLEMENTATION

The solid-state rotary speaker uses an array of eight speakers placed circularly facing outward (see Fig. 1). In this array, each speaker points in a different direction and only produces an output for a fraction of a complete cycle. The array was constructed as eight individual segments, which were bolted together to form a structure with eight faces. Each face of the array is 7.5 cm in length and holds a 2.5-in speaker. The output of each speaker is sequentially windowed to emulate the rotary function. The windowing is set at different widths to emulate different speed settings. The rotating sound sources arrive at the listener with different phases since they follow a different path. These interfere with each other, causing the desired sound effect. The array's directionality (i.e., speakers in different locations pointing in different directions) provides the solid-state rotary speaker with the capacity to emulate a conventional mechanical circular motion in a way that is not possible on a mono or stereo system.

To achieve the operation of the solid-state rotary speaker, a novel topology was developed (see Fig. 2). The authors

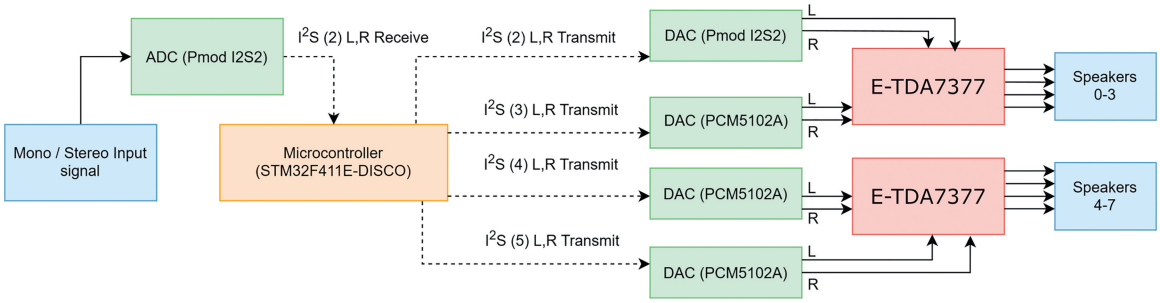


Fig. 2. Block diagram of the implementation of the solid-state rotary speaker. The system uses four I^2S buses to digitalize, split, and amplify the eight speakers on the array. One external ADC is used to digitalize the signal and four DACs to convert it back to analogue. Two four-channel integrated power amplifiers are used then to provide sufficient outputs for eight 2.5-in loudspeakers. An STM32 microcontroller is used for processing the data to activate the multiple speaker outputs at specific times.

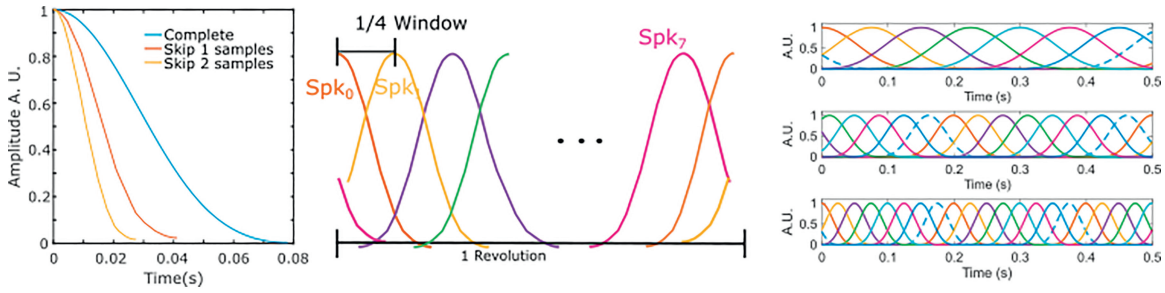


Fig. 3. Output envelopes for the solid-state rotary speaker. With only one half Blackman function (a) (continuous) uploaded to the STM32 memory, the resulting envelopes for speakers 0–7 (W_0–W_7) can be generated by using the same window where two speakers are functioning at any given time: one ascending and one descending until the whole speaker array wraps around one revolution (b). Envelopes for each speaker at (c) 100, (d) 200, and (e) 300 RPM.

have used an STM32 microcontroller (STM32F411VET6) that contains enough I^{2}S [21] buses to drive eight individual outputs. The buses interact with the inputs and outputs with a stereo analog-to-digital converter (ADC; Digilent Pmod I^{2}S2) [22–24] and four stereo digital-to-analog converters (DACs; Texas Instruments PCM5102A [25]), which require four I^{2}S (2–5) interfaces. Each output is amplified separately using two integrated four-channel power amplifiers (STM E-TDA7377 [26]). The speakers (2.5 in) were mounted angled by 45^o along a radius of approximately 19 cm.

The STM32 chip was configured using four I^{2}S buses in a standard real-time configuration [27] (see Fig. 2). The buses were set up using direct memory access (DMA) [28], continuously operating (circular mode) with very high priority and at half-word (24-bit on 32-bit frame). One bus was configured in full duplex mode (for transmitting and receiving data), and the remaining three buses operated in half duplex (transmit only). Each I^{2}S channel (L and R) was stored on 16-bit unsigned integer buffers. A read buffer and four write buffers (four L and four R channels) were created to store the samples read from the ADC and to be written to each of the DACs. A short buffer with only eight memory allocations allowed for low latency (approximately 0.95 ms at 100 RPM) [29].

With the I^{2}S buses set up as described above, the STM32 received and transmitted the samples in half words with interrupts, which were called once the buffer was half filled and once again when it was filled. When each interrupt was received, a corresponding callback function was executed. The sampling rate was set to 62.5 kHz for all the buffers. The I^{2}S interfaces used standard input/output pins.

A function has been developed that uses a half Blackman window [see Fig. 3(a)] to provide the momentary operation of each speaker through envelopes. The half Blackman window is used to calculate the ascending and descending parts of the each speaker's amplitude modulation to create symmetric envelopes. In this function, counters keep track of the index of the half Blackman window and the speaker number, such that every time the function is called, at the DMAs half-complete and complete callback functions, it can return the appropriate multiplication factor for each speaker. The envelopes of the speakers overlap one quarter of the length of the Blackman window, with four

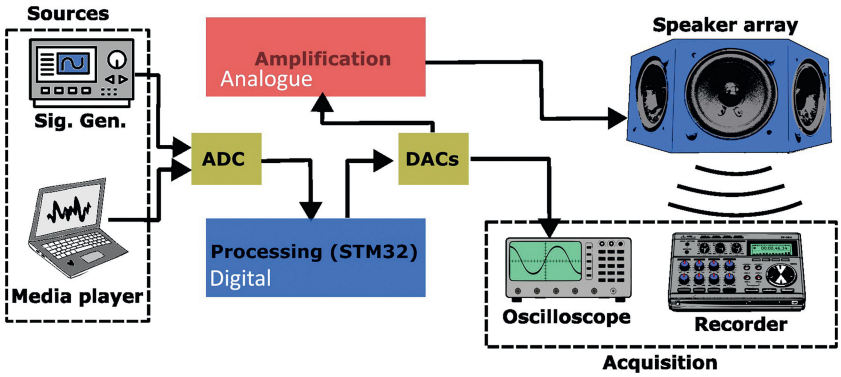


Fig. 4. Experimental setup. To test proof of concept, the authors used two different sources including a signal generator (Sig. Gen.) with test tones of 300 and 500 Hz and a computer source (Media player) playing a pre-recorded piano piece through the computer's sound device output. These signals served as input to our device. Output signals were acquired on an oscilloscope directly from the DACs' output, ensuring correct timing was maintained. Sound emanating from the speaker from test tones and media were captured using the built-in condenser microphones from a Tascam DP-006 recorder.

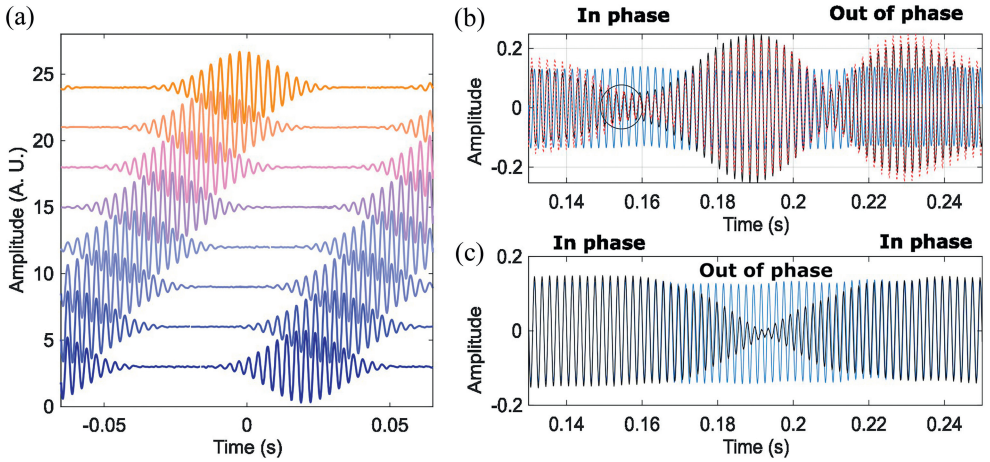


Fig. 5. Experimental traces from the solid-state speaker. (a) Signals obtained at the DACs output of each speaker channel using a 300-Hz tone. The outputs of speakers 0–7 are shown from bottom to top, and the signals are clearly in phase. Speakers are windowed sequentially to emulate movement. The rotation speed is set to approximately 500 RPM. (b) The output of the speaker captured using a recorder with (solid and dashed for L and R, respectively) and without (gray) rotation. There is evident modulation of phase, and this is also slightly different for both microphones (L and R; see circle) of the recorder. Compared to a conventional phaser (c), the solid-state rotary speaker modulation of phase does not match the modulation of amplitude.

speakers remaining sounding at any given time, ascending and descending sequentially as shown in Fig. 3(b). To adjust the speed of the rotation, the corresponding location at the Blackman window is skipped (increase speed) or held [to reduce speed; see Figs. 3(c)–3(e)]. The width of the half-window was selected arbitrarily such that its natural rotational speed was approximately 100 RPM.

3 TEST AND DEMONSTRATION

To demonstrate the functionality of this device, three different tests were performed. Firstly, test signals (300-Hz tone) were collected with an oscilloscope at the individual DAC outputs. Secondly, a 500-Hz tone was used to verify the modulation of phase outside the signal chain using microphones. And finally, a media player reproducing a solo piano recording was used to provide a sound sample of the speaker at different speeds with a musical instrument source. The experimental setup is shown in Fig. 4. A signal generator is used to create the different tones, and a computer is used to reproduce media. The sources connect to the ADC as previously discussed in SEC. 2. The DAC outputs were connected to an oscilloscope to capture the processed waveforms fed to the amplifiers. A Tascam DP-006 recorder was used to collect sound emanated by the speaker array.

The acquisition and windowing functionality was tested by inserting a 300-Hz tone from a signal generator and observing the outputs at all the DACs (see Fig. 4). The frequency of the tone was chosen arbitrarily such that it allowed the signal and envelopes to be shown clearly. The experimental traces are presented in Fig. 5(a). These show that the envelopes of each speaker are behaving as expected: each speaker is modulated by a Blackman window at equally spaced time intervals. Before a rotation is completed, a new cycle starts with speaker 0 rising in amplitude again. This shows that the windowing process does not induce any change in phase at each speaker.

The sound emanating from the speaker was captured using the condenser microphones integrated inside a Tascam DP-006 stereo recorder. The source, a signal generator, provided a 500-Hz tone for testing (see Fig. 4). The tone was selected arbitrarily for display purposes. Fig. 5(b) shows three waveforms, one reference without rotation (gray line) and two with rotation at approximately 500 RPM. The black line is the left channel (L), and the dashed one is the right channel (R).

The rotating signals are modulated both in amplitude and phase compared to the reference one, and there are differences between the channels. The left channel, positioned only a few centimeters away, shows a slightly different modulation of amplitude and phase (see circle). This suggests a dependency on the listener's position. For comparison, the reference signal was also postprocessed with a standard phaser at 4 Hz using Audacity (open source audio processing software). Contrary to this approach, the phaser output modulates amplitude and phase simultaneously as the result of mixing the input signal with a single delayed version of itself.

Finally, to further illustrate proof of concept, the signal has been tested with a prerecorded source (see Fig. 4). The source in this case is a recording of Beethoven's Piano Sonata No. 14 on an acoustic piano. During the testing, the speeds of rotation were approximately 100, 200, and 300 RPM. The resulting sound resembles modulation of phase (phaser-like) and amplitude (tremolo-like) (see Supplementary Data[1]).

[1] https://doi.org/http://doi.org/10.17639/nott.7628.

4 DISCUSSION

This proof of concept demonstrates the functionality of a rotary speaker—modulation of phase and amplitude without moving parts or modulation in the digital or electronic signal chains. To the best of the authors' knowledge, this has not been achieved before. The speaker also emits sounds in multiple directions, and these directions change with time. This directionality is a key missing element of emulations and provides the listener with a more immersive listening experience.

Modulation of phase comes from the different sources interfering at the listener location after following different paths, causing the characteristic whooshing sound. Modulation of amplitude, even though it is used to achieve functionality, also arises from interference. This is therefore not consistent across the signal, like a conventional tremolo, but changes with the environment and the listener's position.

One of the key limitations of this concept is the lack of continuous movement of the sources. As the speed increases, the combination of sources might better approximate continuous movement and Doppler effects. It is unclear at this stage the extent of this compromise. However, this approach does not intend to accurately mimic or emulate the original Leslie's complex physical dynamics. Instead, a new approach is presented for sound modulation inspired by rotary speakers.

Nevertheless, some of the original Leslie speaker emulation elements could be included in this system. For instance, the Leslie has two rotating elements(a horn and baffle), which spin in opposite directions. Such functionality could be mirrored by separating the lower and higher frequencies from the input signal and making them rotate in opposite directions.

This proof of concept has many areas that would benefit from improvement and optimisation. These include, from the software perspective, the window overlap and separation of the low/high frequencies. From the hardware side, optimisation includes signal conditioning before and after the microprocessor, full frequency response, continuous adjustments of speed and gain, and the number, size, and spacing between the speakers. The authors anticipate that larger speakers with full frequency response and a dedicated speaker cabinet will improve sound quality. However, these improvements remain outside the scope of this work.

Increasing the window overlap would minimize the modulation of amplitude experienced between windows, while decreasing it would reduce directionality as more speakers would be operational at one time. Better signal conditioning will reduce noise and improve the frequency response of the system by providing appropriate impedance matching and levels among the different elements (source, microprocessor, and power amplifiers). It is out of the scope of this proof of concept to explore all these design aspects. Even though circular arrays are unusual, it is possible that a similar effect could be achieved with a more common front-facing array of speakers set up in a rectangular grid. Such a speaker cabinet arrangement is already common practice on instrument amplifiers of various sizes. This could allow for different patterns of movement across the array. For instance, an "8" or zig-zag–shaped movement could be experimented with, in addition to circular movements.

The digital signal processing engine can provide further functionality. For instance, the frequency response of the direct and modulated signals can be modified to emulate different baffle or horn characteristics without any modification to the system. The input could be split into three or more bands and rotate at different speeds. Additional delays or modulation of phase could also be used to accentuate directionality, akin to 3D audio. Digital signal chains are common practice on electronic pianos, organs, and synthesizers. However, for analog instruments such as an electric guitar or bass, this might not be as desirable.

The need for multiple analog amplification channels and speakers can imply a cost and weight penalty compared to normal amplifier systems, particularly as the total output power capacity can only be exploited when all speakers are operating continuously. However, the authors are confident that this will still provide some of the directionality of the original Leslie. Finally, the subjective sound tests presented in SEC. 3 show what the proof of concept can sound like. The sample piano sound illustrates the intended application. A characteristic phase modulation sound is clear, along with a modulation of amplitude. However, it is up to the reader to judge its qualities.

5 CONCLUSION

This paper presented the proof of concept for a new type of rotary speaker that does not use moving parts. The device uses a circular array of speakers to emulate the movement of a sound source. This device can modulate sound waves outside the electronic and digital signal chains, similar to the original Leslie speaker and without moving parts. The authors believe that this new hybrid digital/acoustic concept represents a new paradigm in sound effects for instrument amplification. The authors expect hybrid systems, such as the one presented here for the first time, to have a significant impact on the production of new music.

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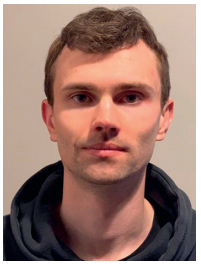
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