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Multiple-Level Digital Loudspeaker Array

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Abstract

In modern audio systems, the signal path is generally digital in nature, with the notable exception being the analogue loudspeaker. Digital Loudspeaker Arrays (DLA) have been proposed as an alternative approach. Generally, they have a relatively poor acoustic output, as high audio quality requires an impractically large number of speaklets (tiny loudspeakers). We address this limitation by proposing an increase in the number of quantizing levels, thus requiring fewer speaklets. A comprehensive system simulation has demonstrated the feasibility of such a system to produce high quality sound, with a hardware implementation currently being investigated.

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Keywords: Digital Loudspaker Array;Digital Transducer Array

1. Introduction

The concept of a digital loudspeaker array (DLA) was first published in 2002 by Busbridge *et al* and Diamond *et al* [1][2]. A digital loudspeaker array consists of a number of speaklets, each of which is driven by rectangular pulses with constant width and amplitude in order to produce sound. It is envisaged that a practical application would use a piezoelectric material as the array element. To allow accurate reconstruction of the analogue signal from the digital speaker inputs, the acoustic output from the speaklets needs to meet three main requirements [1]:

1. The oscillation of the acoustic response of each digital pulse feeding into a speaklet has to cease before the occurrence of the next sampling period. For conventional audio systems, the sampling period is 22.7 µs (44.1 kHz).
2. All speaklets in the array must have a uniform acoustic response and every repeated response must be the same over time.
3. The step increase of the maximum pressure of the acoustic response must be linear.

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A significant problem with the DLA is that the bit quality of the acoustic output is dependent on the number of speaklets. Typically, for conventional audio systems, this is 16-bits. Consequently, a DLA would require 65532 speaklets in order to reproduce the sound at this quality. This requirement makes the practical implementation difficult. Not only is the scale of the array and driving circuits large and complex but also sound-path differences among the speaklets causes a considerable increase in signal distortion.

1.1. The Concept of the Multiple-Level Digital Loudspeaker Array.

We propose a concept of a multiple-level digital loudspeaker (MDLA), which increases the number of levels of sound that a speaklet can emit. The nature of the pulses feeding the MDLA will thus differ from those used with a conventional DLA, where the width and amplitude of the pulses are uniform. An MDLA requires pulses of constant amplitude, but variable width as shown in Figure 1, thereby maintaining the digital nature of the system.

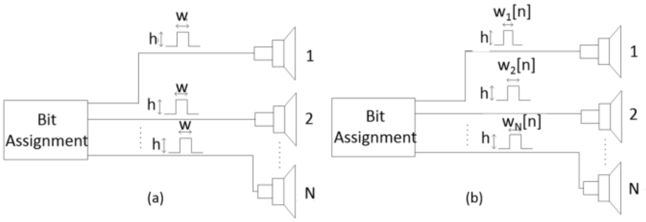

Figure. 1. (a) Traditional DLA and (b) MDLA.

1.2. Mathematical Model of the Acoustic Response.

A Matlab simulation was developed in order to study the acoustic behaviour of the proposed MDLA. The speaklets are mathematically modelled as mass-spring-damper systems [3]. The displacement of mass q(t) can be expressed as Eq.1.

$$\ddot{q}(t) + 2\varphi\omega_n\dot{q}(t) + {\omega_n}^2\, q(t) = f_e(t) \tag{1}$$

For a digital loudspeaker system, $f_e(t)$ is a discrete pulse, which can be expressed as Eq. 2.

$$f_e(t) = \begin{cases} h, & 0 < t < w \\ 0, & otherwise \end{cases} \tag{2}$$

Where h and w are the height and width of the pulse. The equation of motion of the mass $q(t)$ is conveniently solved by Laplace transform methods utilising Eq, 1 and 2, which yields Eq. 3.

$$q(t) = L^{-1}\left\{\frac{h(1-e^{-sw})}{s(s+\sigma_d+\omega_d)(s+\sigma_d-\omega_d)}\right\} \tag{3}$$

where $\sigma_d = \omega_n\varphi$ and $\omega_d = \sqrt{{\sigma_d}^2 - {\omega_n}^2}$

2. Assumptions and Results of Simulation of MDLA

Speaklets within the MDLA produce different acoustic pressures as they are driven by electrical pulses having different widths. In order that the acoustic response meets the requirement for digital reconstruction, the natural frequency and damping ratio of the speaklets are set at 80 kHz and 0.7 respectively. The speed of the clock generator, which is used for generating 1 volt pulses, is set at 200MHz which allows digital pulses with variable pulse widths from a minimum of 5 ns with a resolution of 5 ns.

From the above assumptions, the acoustic output of a speaklet can be related to the pulse width of the electrical rectangular pulse in Eq. 3, as shown in Figure 2a. The pressure of the acoustic output reduces to 0 (or negligible) voltage in less than 22.3 µs. This means that the response can digitally reconstruct sound at a sampling rate of 44.1 kHz, in accordance with the first requirement mentioned above. In addition, the relationship between maximum pressure and pulse width, and the relationship between the response time and pulse width are linear up to pulse widths of 4.685 µs, with R^2 coefficient of 0.9917 and 0.9967 respectively as shown in Figure 2b (fulfilling the third requirement). Therefore, 937 different pulse widths are available for a speaklet, by varying the pulse width from 0 ns to 4.685 µs in steps of 5 ns. For a 16-bit resolution in a conventional audio system, this requires a speaklet array of 70 elements.

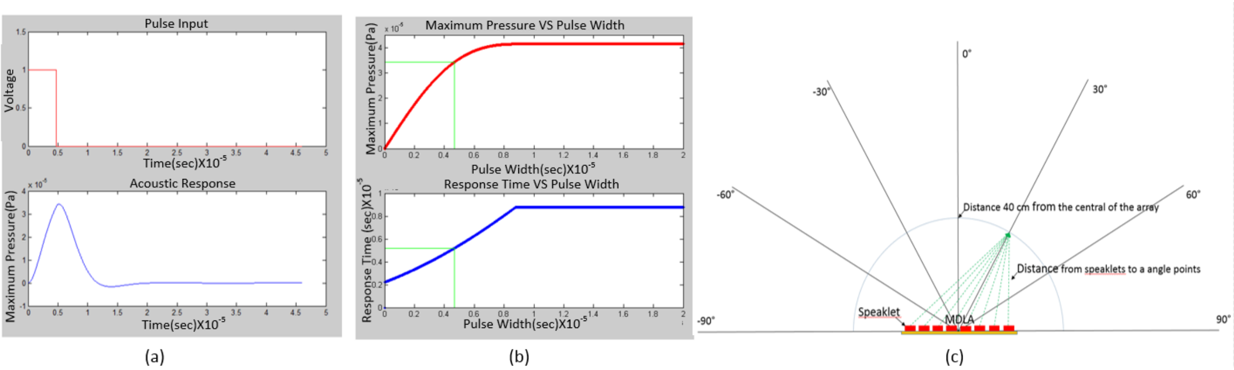

Fig. 2. (a) acoustic output when feeding a pulse with 4.685 µs width, (b) the relationship between maximum pressure and pulse width and the relationship between response time and pulse width, (c) speaklets in the linear array

For sound reconstruction within the MDLA, there are two additional points to consider: Firstly, due to differences in pulse width, the response times of individual speaklets will be different. In order to ensure that the speaklets produce an acoustic response, which reaches the maximum pressure at the same time, the delay of sending electrical pulses needs to be calculated from the relationship between the response time and pulse width. Secondly, due to the acoustic output of MDLA resulting from a superposition of the response of speaklets within the array, there are a variety of combinations of different levels of speaklets allowing the same quantized level of output of the array. Therefore, a specific combination of the levels is assigned for a specific quantizing level of the output. For this simulation, the number of levels of the speaklets is assigned by using all the speaklets within the array and minimising difference in their levels of sound production.

The sound field intensity was also simulated, by assuming that the speaklets are point sources aligning on the *x*-axis and their interspacing is equal to 3.83 mm (a half of sampling distance at 44.1 kHz) as shown in Figure 2c. Figure 3 shows the spectral response from an array at a distance of 40 mm from the centre of the array through different angles and acoustic outputs.

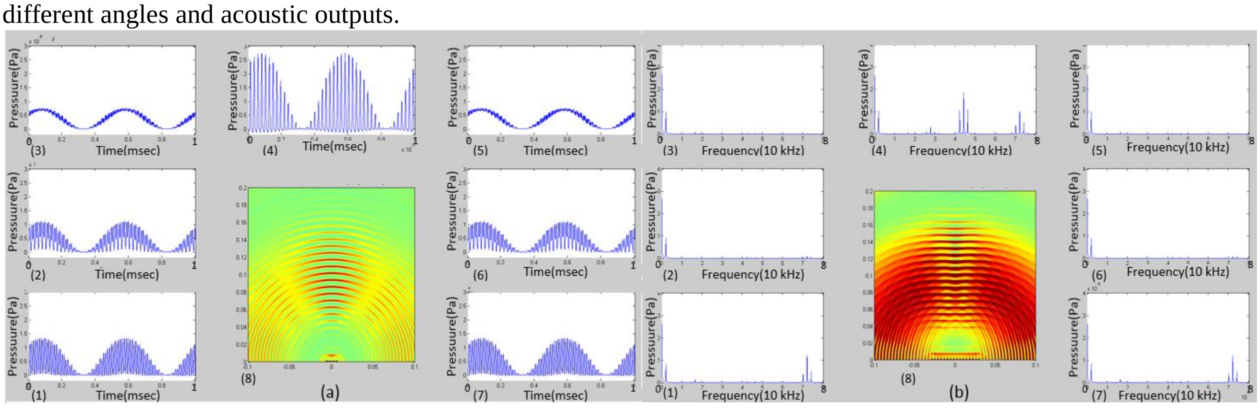

Figure 3: a) 8 Sound field for 2 kHz with 4 speaklets within the area of 20x20 mm and 1-7 are acoustic output at angles of -90, -60, -30, 0, 30, 60 and 90 respectively. b) 8 Sound field for 2 kHz with 16 speaklets within the area of 20x20 mm and 1-7 are output spectrums at angles of -90, -60, -30, 0, 30, 60 and 90 respectively.

Figure 3 shows both the temporal directivity response and the spectral content at that angle with number of speaklets for an audio frequency of 2kHz. It can be seen that the outputs consists of three main components of frequency, especially at an angle of 0 degrees (i.e. directly in front of the element).The first component is the required audible frequency, which is reproduced by digital reconstruction. The second component depends on the natural frequency of the speaklets, which is around 44.1 kHz. The last component is a harmonic frequency of 71.9 kHz. However, the last two components have no effect on hearing because they are beyond the response of the human ear. For these three frequency components, directivity is shown in Figure 4. For a frequency of 2 kHz, sound radiates omnidirectionally. For a frequency of 10 kHz, as the number of speaklets increases, the response is more directional, which is as expected.

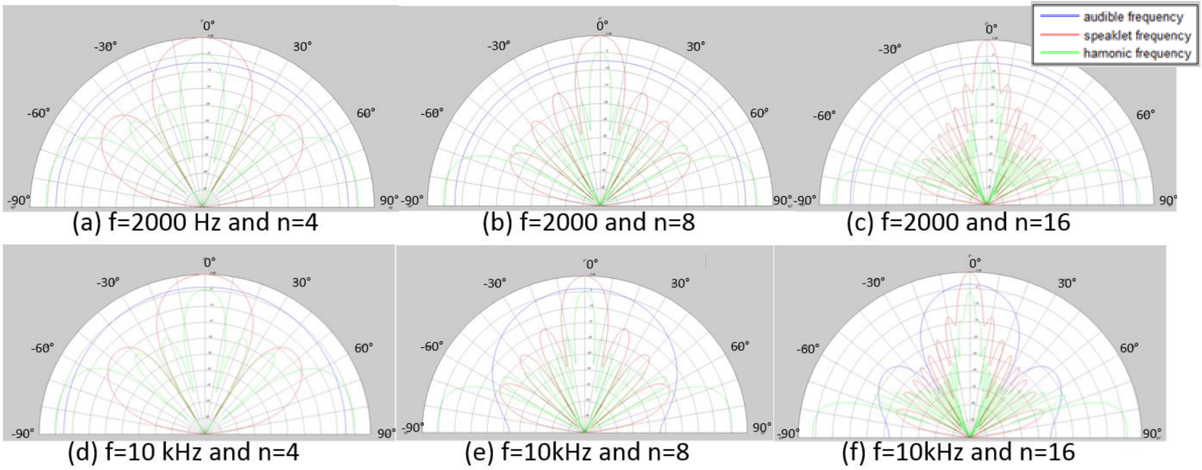

Figure 4: Sound beam directivity a, b and c for pure tone at 2000 Hz and d, e and f for pure tone at 10 kHz with 4, 8 and 16 speaklets respectively. Blue, red and green lines represent audio frequency, speaklet resonant frequency and harmonic frequency respectively.

3. Discussion and Conclusion

The results of the simulation demonstrates that the MDLA can produce audio. However, the simulation has not taken into account some practical issues such as:

- impedance mismatch and efficiency of the piezoelectric element,
- the directivity of an individual speaklet within the array resulting from its shape and size,
- non-uniformity of acoustic outputs of the speaklets due to imperfections in the fabrication process,
- difference in delay time from the pulse generator to the different locations of speaklets and
- non-ideal shape of the rectangular pulse, because of the state transition limitations of transistors.

The speaklets in the array are also linearly aligned in this simulation, while the array will be square in the practical implementation. The concept of the MDLA is amenable to fabrication by thick-film printing using active materials such as PZT and an experimental evaluation of such a system is currently being implemented. The characteristics of thick-film sensors are such that they allow the production of low-cost, robust and miniaturized array elements. Although the study of sound reproduction of MDLA from the simulation is only an ideal case, it indicates that people will be able to hear the sound reproduced by the MDLA, and gives confidence that good performance is possible from the concept, despite the presence of ultrasonic components.

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