# Multiple-Level Digital Loudspeaker Array

**Sangchai Monkronthong*, Neil White and Nick Harris**

**School of Electronics and Computer Science, University of Southampton, Hants. SO17 1BJ, UK**

**Tel: +44 23-8059-3274 Fax: +44 23-8059-2931 *Email: sm12e11@soton.ac.uk**

UNIVERSITY OF Southampton

## Introduction

A digital loudspeaker array (DLA) consists of a number of speaklets, each of which is driven by rectangular pulses with constant width and amplitude in order to produce sound. For accurate reconstruction of the analogue signal from the digital speaker inputs, the acoustic output from the speaklets needs to meet three main requirements: suppression of oscillation of the acoustic response before the next sampling period, uniformity of acoustic response and linearity of increase of the maximum pressure of the acoustic response [1].

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 practical implementation difficult.

## Concept of the Multiple-Level Digital Loudspeaker Array

This work proposes 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.

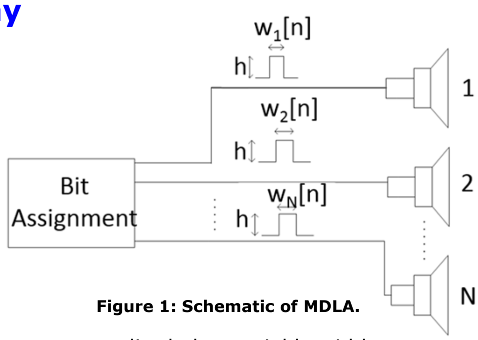


**Figure 1: Schematic of MDLA.**

An MDLA requires pulses of constant amplitude but variable width, as shown in Figure 1, to maintain the digital nature of the system.

## Mathematical Model of the Acoustic Response

The speaklets are mathematically modelled as mass-spring-damper systems. The displacement of mass q(t) can be expressed as Eq.1.

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

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

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

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 utilizing Eqs. 1 and 2, which yields Eq. 3.

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

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

## Assumptions and Results of Simulation - 1

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 200 MHz which allows digital pulses with variable pulse widths from a minimum of 5 ns and with a resolution of 5 ns.

The acoustic output of a speaklet can be related to the pulse width of the electrical rectangular pulse (Eq. 3), as shown in Figure 2. The figure also shows that the relationship between maximum pressure and pulse width, and that between the response time and pulse width, are linear up to pulse widths of 4.685 µs.

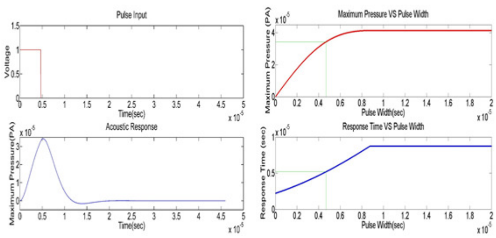


**Figure 2: (Left) acoustic output when feeding a pulse with 4.685 µs width. (Right) relationship between maximum pressure and pulse width and relationship between response time and pulse width.**

[1] B. M. Diamond, J. J. Neumann, and K. J. Gabriel, "Digital Sound Reconstruction Using Arrays of CMOS-MEMS Microspeakers," in Fifteenth IEEE International Conference on Micro Electro Mechanical Systems, 2002, no. 412, pp. 292–295.

## Assumptions and Results of Simulation - 2

The sound field intensity was also simulated by assuming that the speaklets are point sources aligned on the x-axis and their interspacing is equal to 3.83 mm (half wavelength in air at 44.1 kHz). The results are shown at a distance of 40 cm from the centre of the array at different angles. Figures 3 and 4 show both the temporal directivity response and the spectral content versus angle with the number of speaklets for an audio frequency of 2 kHz. The outputs consists of three main components of frequency. 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 since they are beyond the response of the human ear. For these 3 frequency components, directivity is shown in Figure 5. For a frequency of 2 kHz, sound radiates omni-directionally.

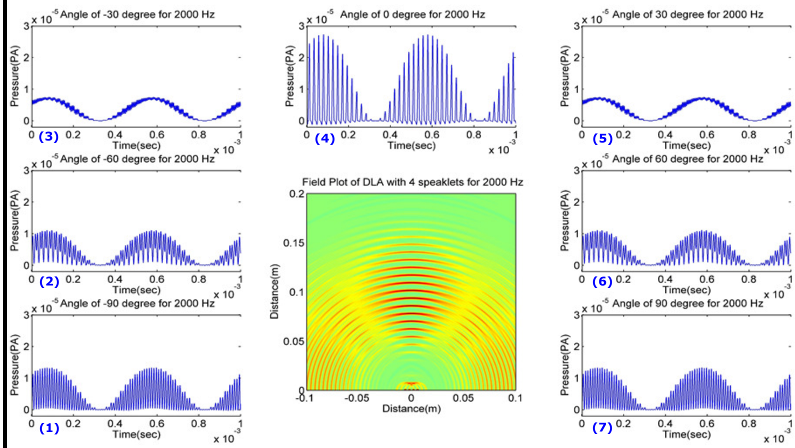


**Figure 3: Central image shows sound field at 2 kHz with 4 speaklets within an area of 20 x 20 mm. Satellite images (1 to 7) show acoustic output at angles of -90, -60, -30, 0, 30, 60 and 90 degrees respectively.**

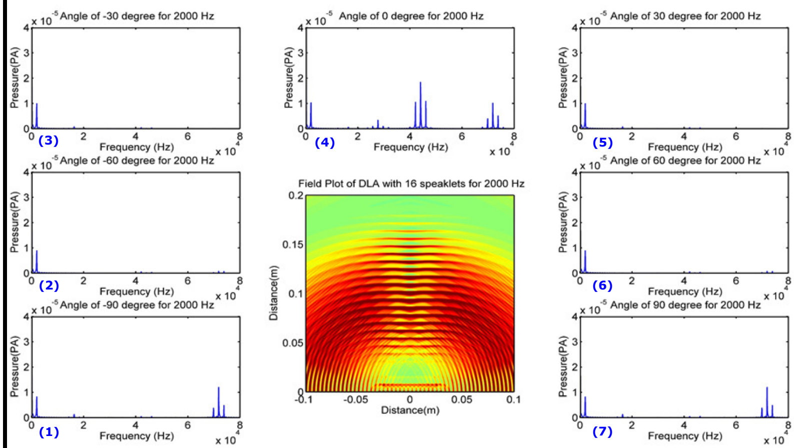


**Figure 4: Central image shows sound field at 2 kHz with 16 speaklets within an area of 20 x 20 mm. Satellite images (1 to 7) show output spectrums at angles of -90, -60, -30, 0, 30, 60 and 90 degrees respectively.**

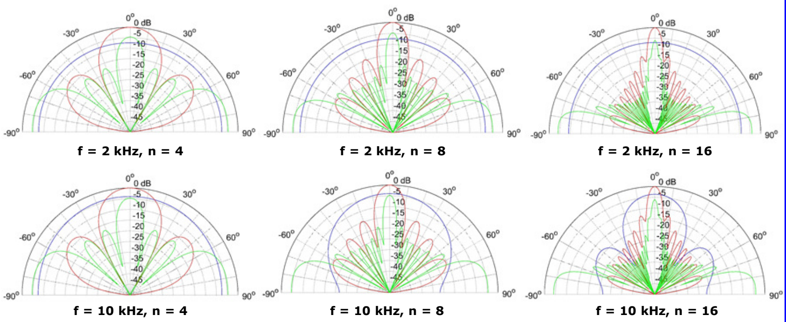


**Figure 5: Sound beam directivity as a function of pure tone frequency and speaklet number (n). Blue, red and green lines represent audio frequency, speaklet resonant frequency and harmonic frequency respectively.**

## Conclusions

Simulations indicate that MDLA can produce audio. The concept of the MDLA is amenable to fabrication by thick-film printing using piezoelectric materials such as PZT and an experimental evaluation of such a system is currently being implemented.

**Eurosensors XXVIII, 7-10 September 2014, Brescia, Italy**