Design and Performance of a Sigma–Delta Digital Loudspeaker Array Prototype*

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Digital loudspeaker arrays (DLAs) are electroacoustic transducers reconstructing acoustic signals out of binary audio streams, usually consisting of a digital signal processing module driving multiple actuators (small loudspeakers). In this work a DLA prototype was implemented using an all-digital audio chain driven by 1-bit sigma–delta signals. The prototype implementation is described and its performance measured and evaluated using frequency-domain and THD analysis. These results also compare the DLA performance when driven by either sigma–delta or PCM input signals.

0 INTRODUCTION

Digital loudspeakers employ conventional transducers for direct digital-to-analog acoustic conversion [1] and may be utilized in an all-digital audio reproduction chain potentially having the advantages of greater flexibility for the user, exact control of operation, greater scale of integration, smaller size, lower cost, and possibly improved electroacoustic efficiency over existing analog systems [2]. In most cases the transducers will form a digital loudspeaker array (DLA), which will allow the direct acoustic emission of the audio bit stream and may also realize controlled directivity [3], [4], as is currently mainly utilized by digital signal processor–controlled analog arrays for surround sound reproduction via a single box [5]. Such digital direct acoustic emission systems can be fully configurable via software, which apart from controlled directivity functions can also realize any desired method of bit allocation, equalization, and so on, implemented via DSPs and field-programmable gate arrays (FPGAs).

Past research on digital loudspeakers was mainly focusing on the direct acoustic transduction of multibit PCM signals [1], [6], [7], where the bit significance of each binary word was translated into the appropriate acoustic gain via the activation of groups of emitting elements (bit grouping). Given the high complexity and the large number of loudspeaker elements needed for mapping 16-bit PCM audio signals to the transducers, requantization, oversampling, and conversion to lower bits are essential steps for any practical bit-grouped PCM-driven DLA realization [1], [7]. Furthermore it has been established that DLA performance is strongly dependent on the listening position having a sweet spot for on-axis radiation where acoustic path lengths from all elements are comparable [7]. Other drawbacks of DLA technology include element and amplifier mismatch, ultrasonic emission due to the absence of low-pass filtering, and insufficient lower frequency reproduction response due to size restrictions for the transducers.

Given that the preliminary simulation results for sigma–delta-driven DLAs compared to PCM-driven systems were promising [9], the present work describes the design, construction, and measurement of a novel digital transducer array implemented as a prototype and tested, driven directly by 1-bit sigma–delta audio streams. The prototype includes a DSP stage implemented via a FPGA, a true digital amplification module, and a two-dimensional array consisting of small electrodynamic loudspeakers. This prototype structure allows flexible adaptation via reprogramming of the FPGA between many digital audio formats, such as the Sigma–Delta Modulation (SDM) format [10], PWM, or low-bit oversampled PCM [11].

This engineering report is organized as follows. In Section 1 a brief technology overview is given as well as the assignment technique used for the current implementation. In Section 2 the DLA implementation presented is explained in detail, and in Section 3 the results from the measurements and the simulations are given. Finally in the last section the results are analyzed and some conclusions are drawn.

1 SIGMA–DELTA DIGITAL LOUDSPEAKER ARRAYS

1.1 Technology Overview

A digital loudspeaker usually consists of digital signal processing and amplification modules driving miniature loudspeaker elements, positioned on a surface to form a DLA. Such DLAs consist of a set of N transducer elements, each with dedicated digital amplification, usually in a two-dimensional circular arrangement. The DLA is fed directly with the digital audio signal, and depending on the audio coding used for the digital data, such systems may be broadly grouped into two categories: binary DLAs (PCM driven) [7] and unary DLAs (Sigma–Delta driven) [9], [12], as realized and studied here.

Hence any such system consists of three distinct modules (Fig. 1):

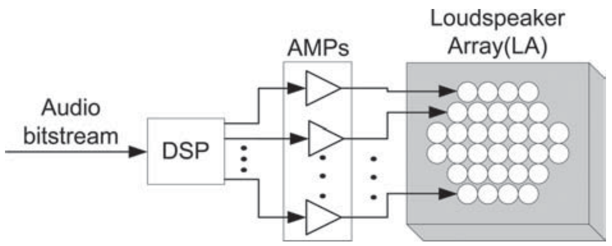


Fig. 1. Typical structure of a DLA system.

- A digital signal processing (DSP) module, which implements bit grouping and assignment
- A digital audio amplification (AMP) module, which amplifies each of the resulting bit streams digitally
- An array of loudspeakers (LA) driven by the amplified bit streams.

These stages can be driven by appropriate preprocessed versions of the incoming PCM signal, either in a low-bit oversampled PCM format, or even converted into sigma–delta bit streams. In this sense a DLA may be considered to have a function comparable to the traditional DACs used in audio applications.

1.2 Sigma–Delta Signal-to-DLA Mapping and Bit Assignment

The mapping of any 1-bit sigma–delta input signal within the DLA as used in the present implementation has been documented in previous work [9]. According to that, the sigma–delta input bit stream is fragmented into frames that consist of the same number of bits as the number of elements on the LA, as shown in Fig. 2. The frames thus produced are digitally amplified and fed directly to the LA. This is a convenient approach allowing the same LA arrangement to be used also for multibit PCM input signals by appropriate signal preprocessing (in the FPGA stage, see next section). Driving the DLA with alternative audio formats allows comparing the signal quality of the different signal encodings.

The DLA implemented here consists of $N = 32$ transducers, arranged in a circular layout on a two-dimensional flat surface. The type of topology examined and tested in the present work employs a unary input signal, thus assuming that all transducers have equal significance and are given an index number k $(1 < k < 32)$, as shown in Fig. 3(a). Hence when the LA is driven by a 32-bit sigma–delta frame, this will be directly mapped to the

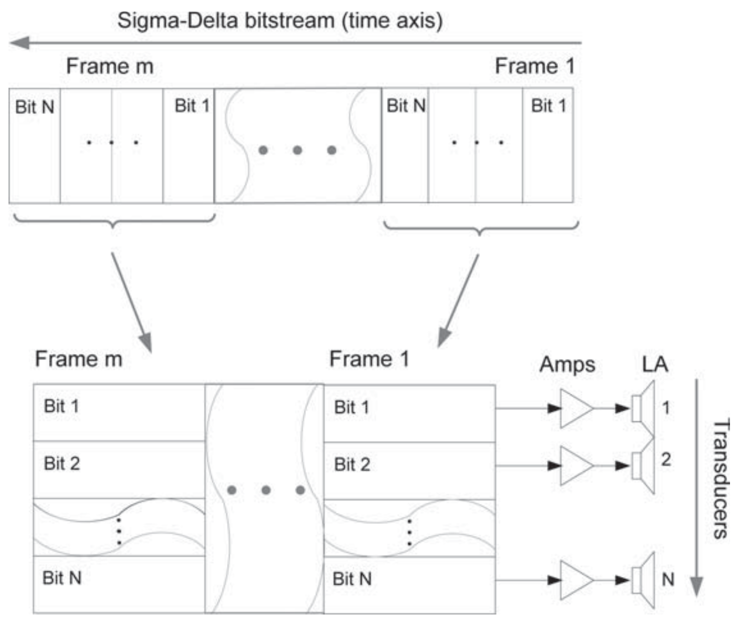


Fig. 2. Typical SDM stream assignment on DLA.

LA elements. As an example, the sigma–delta frame 01000000110011000000000010000010 will generate the LA activation pattern shown in Fig. 3(b).

2 PROTOTYPE IMPLEMENTATION

For the implementation of a sigma–delta DLA working prototype, the system shown in Fig. 4 was realized. It consists of four subsystems.

1) A sigma–delta modulator
2) A bit-grouping module implemented in FPGA
3) Digital amplifiers (driver and power stage)
4) Acoustic emission elements (LA).

Here, a high-definition analog-to-digital sigma–delta converter supporting a digital output [13] was selected as a viable solution, as shown in Fig. 4. The system is based on an integrated circuit [14], which incorporates a high-precision sigma–delta modulator, an antialiasing linear-phase digital filter, and a high-pass filter to suppress the dc component.

The DLA bit-grouping subsystem based on the algorithm described in Section 1.2 was developed in an HDL language and implemented on an FPGA chip (see Fig. 4) [15]. The digital SDM signal is fed to the FPGA in conjunction with the corresponding clocking signal, and the 32-bit parallel signal is output to drive the digital amplification stage. Any additional algorithms for the optimization of the bit-grouping procedure as well as "smart" directivity control [4] may also be embedded into the FPGA since currently the bit-grouping stage employs few of its resources. Alternative bit-assignment implementations to optimize the array behavior [16] can be tested simply by modifying the algorithm on the FPGA stage, without further alterations to the DLA architecture.

The digital amplifier subsystem consists of 32 separate autonomous digital amplifiers. The subsystem receives the input from a 40-pin header from the FPGA and drives the loudspeakers through 32 discrete outputs. The digital amplifier board topology corresponds directly to the loudspeaker array topology in order to achieve optimal interconnection between the subsystems. Fig. 5 shows the circuit diagram for one digital amplifier. The amplifier is designed with three stages, one for voltage adaptation (based on Q1, a PNP prebiased small signal transistor), followed by the output stage drive (based on NPN/PNP general-purpose transistors Q2 and Q3), leading to the power stage (based on Q5, a dual N&P-channel enhancement field effect transistor), which finally drives the loudspeaker.

Fig. 6 shows a time instance of the input voltage (top) compared to the output voltage (bottom) for one digital amplifier, for a purely resistive output load [Fig. 6(a)] and a loudspeaker load [Fig. 6(b)]. Clearly the inductive nature of the loudspeaker causes overshoots and some oscillation (ringing) at the switching point. These components are expected to have some negative impact on the reproduction quality.

The DLA was developed using 32 electrodynamic miniature drivers, each approximately 15 mm in diameter, similar to those used in portable multimedia applications and laptop computers [17], mounted on an acrylic glass panel in a symmetrical (circular) topology, as shown in Fig. 7. The element selection was carried out taking into account their performance in terms of sensitivity and spectral balance, given the diameter limitation requirement for any DLA in order to reduce the effect of the different acoustic paths for off-axis radiation. As mentioned in [17], the element has a sensitivity of approximately 76 dB/W/m, with the response specified for frequencies from 500 Hz to 40 kHz. Measurements of the loudspeaker have indicated that the frequency

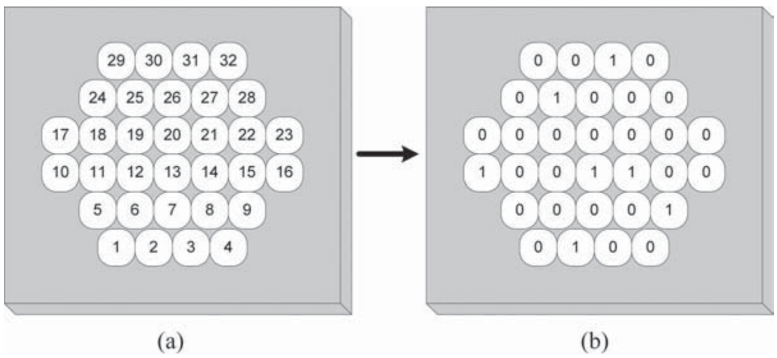


Fig. 3. Typical two-dimensional circular 32-element arrangement and sigma–delta activation pattern. (a) 32-element LA indexing. (b) 32-bit sigma–delta frame activation pattern.

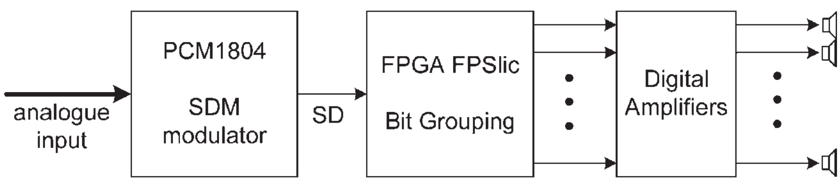


Fig. 4. Architecture of sigma–delta DLA prototype.

response (magnitude) shows a peak of more than 3 dB around 4 kHz. Fig. 8 shows the measured response as well as the one-third-octave smoothed response for the on-axis position.

3 TESTS AND RESULTS

3.1 Measurement and Simulation Parameters

The tests compared measurements on the prototype and results derived via simulations of the identical array topology employing the activation pattern described in Fig. 3. These tests were carried out at the acoustically treated Audio and Acoustic Technology Group lab, following the setup shown in Fig. 9.

A PC with a high-quality sound card and the appropriate software was used to measure the response of the array, along with the THD values for sine-wave signals. The analog signal at discrete sine-wave frequencies of 500 Hz and 1, 2, 4, 8 and 16 kHz was sampled at 48 kHz with 16-bit resolution by the sigma–delta module, sigma–delta modulated using fifth-order NTF and a 64 times oversampling factor, resulting in a sigma–delta 1-bit stream at 3072 MHz. Measurements were obtained via an omnidirectional measuring microphone at a distance of 2 m (far field) and horizontal angles varying from $\varphi = 0°$ (on axis) to $\varphi = 60°$ in steps of 30°. For the simulations the response of the complete array was evaluated by the superposition of each element's frequency response [9]. Measurement and simulation parameters were kept identical, except for some practical simplifications, such as assuming that all elements in the array will have identical responses to the single measured loudspeaker element.

3.2 Measurement and Simulation Results

3.2.1 Frequency Response

The on-axis frequency response of the acoustic pressure generated by the DLA measured at a distance of 2 m with either SDM or PCM (simulated) input is shown in Fig. 8. From the results it is evident that the overall DLA response follows the general response trends of the individual loudspeaker elements. Furthermore the on-axis simulated PCM DLA response seems to be identical to the measured sigma–delta response. These frequency response results are in good agreement with previously published results for PCM-based DLAs [7].

3.2.2 Single-Frequency Output

Typical magnitude spectra derived from waveforms reproduced at a distance of 2 m and at the horizontal angles of 0° and 30° for a 4 kHz input sigma–delta encoded sinusoidal signal are shown in Fig. 10, where a qualitative scale representation is used.

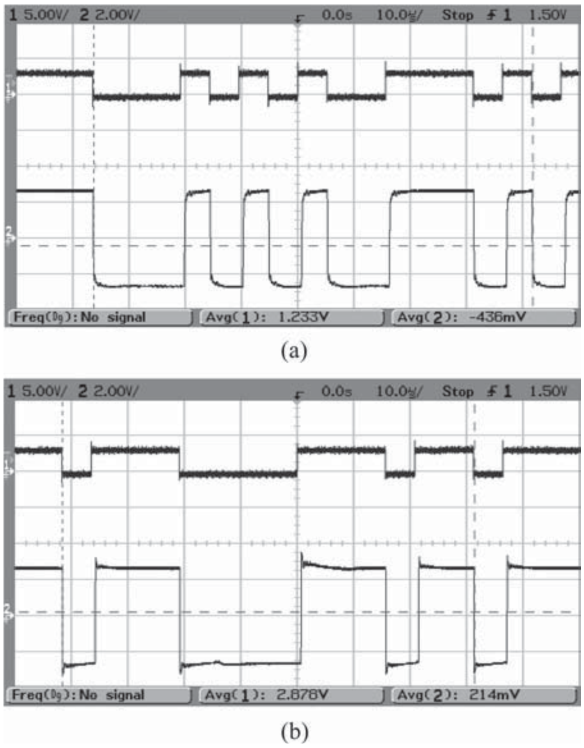


Fig. 6. Input stream and digitally amplified output voltage. (a) Resistive load. (b) Loudspeaker load.

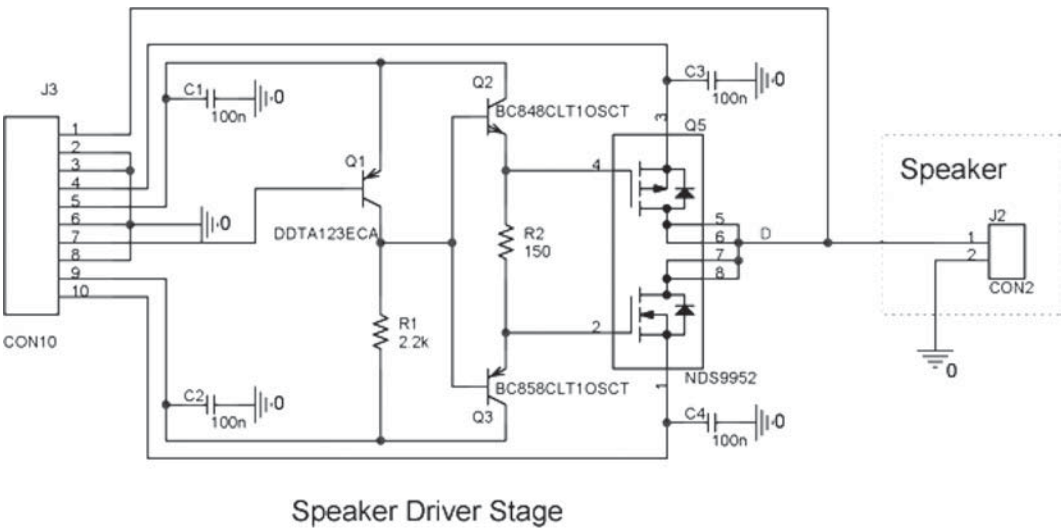


Fig. 5. Circuit diagram of amplification stage.

As is known [1], [7], [9], DLA harmonic distortions occur for off-axis angles due to acoustic path differences from the emitting LA elements and the resulting nonlinear reconstruction of the acoustic signal from the combined bit streams. The results shown in Fig. 10 illustrate off-axis generated harmonic distortions clearly. Such distortions are significantly lower for the on-axis measurements [Fig. 10(a)].

3.2.3 Harmonic Distortion

By extending the preceding measurements it was found that DLA harmonic distortion was sufficiently low for the on-axis position, but increased with angle and frequency, reaching a maximum THD of 60% at around 5 kHz and a 60° angle. However, the measured DLA THD performance appears to be significantly better than comparable results for an identical architecture driven by PCM streams, as was reported in [7]. Furthermore the results of Fig. 11 indicate good agreement for $\varphi = 30°$ sigma–delta DLA simulated and measured results. However, some increase in THD for $\varphi = 0°$ was observed for the prototype when compared to simulations. The prototype also exhibits lower THD for $\varphi = 60°$, possibly due to the less benign nature of the real-life off-axis radiation pattern of the transducer elements. Comparing the sigma–delta results to the performance of the equivalent PCM-based DLA [Fig. 11(c)] it can be observed that in most cases, and especially for $\varphi = 30°$, sigma–delta THD performance is significantly superior.

4 DISCUSSION AND CONCLUSIONS

Digital loudspeakers, and specifically DLAs, may present a promising evolution of traditional loudspeakers. While a number of studies have been presented for multibit DLAs, this work introduces and evaluates a sigma–delta-driven working DLA prototype. This prototype consists of an FPGA connected to 32 autonomous true digital amplifiers driving a corresponding number of miniature moving-coil loudspeakers.

It has been well documented that the main source of distortion for the DLA acoustic emission comes from the different path lengths between each element on the array and the listening position. On-axis listening positions have the advantage of delay cancellation, particularly when a symmetric topology is chosen, as was the case with the present prototype. It is thus clear that smaller transducer dimensions will eventually lead to better DLA performance in terms of harmonic distortion, because in such a case arrays may approach the acoustic emission of a point source. However, given that the DLA size will be

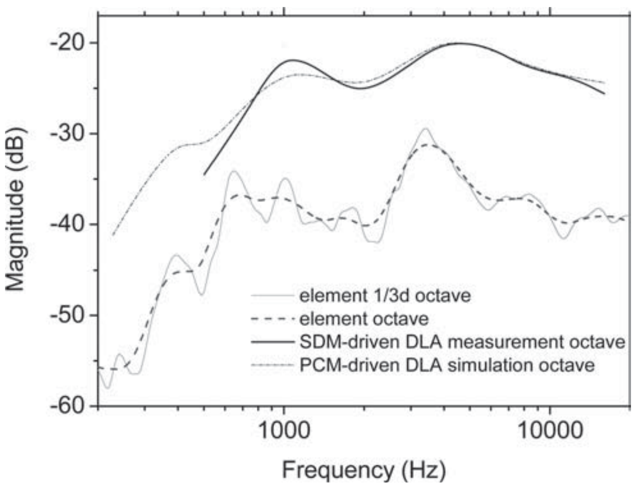


Fig. 8. On-axis frequency responses for one loudspeaker driven by analog excitation (one-third-octave and octave smoothing) and DLA output driven by SDM and PCM streams (octave smoothed). Top curves are shifted vertically for clarity.

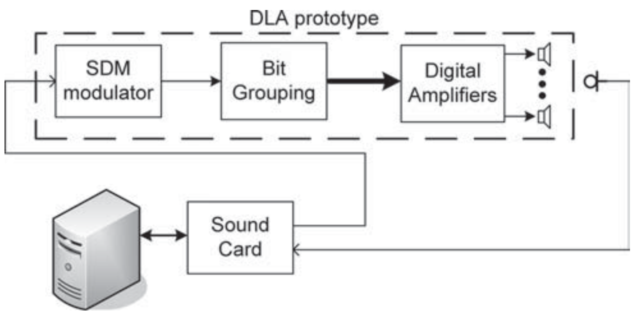


Fig. 9. Diagram of sigma–delta DLA measurement setup.

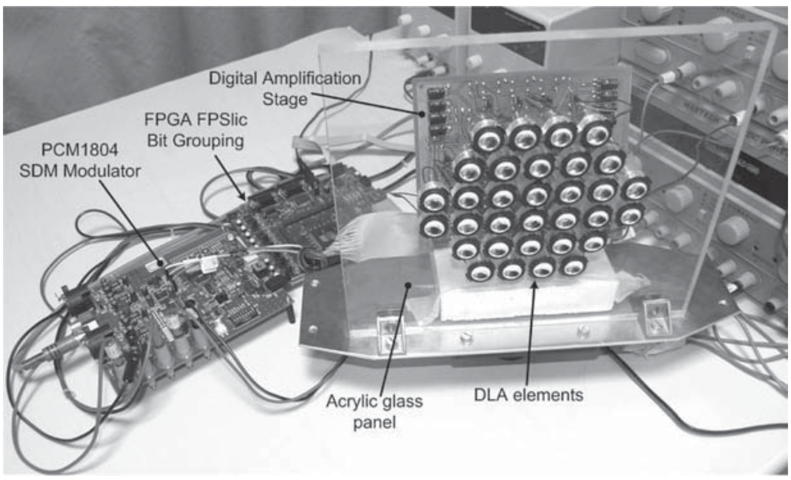
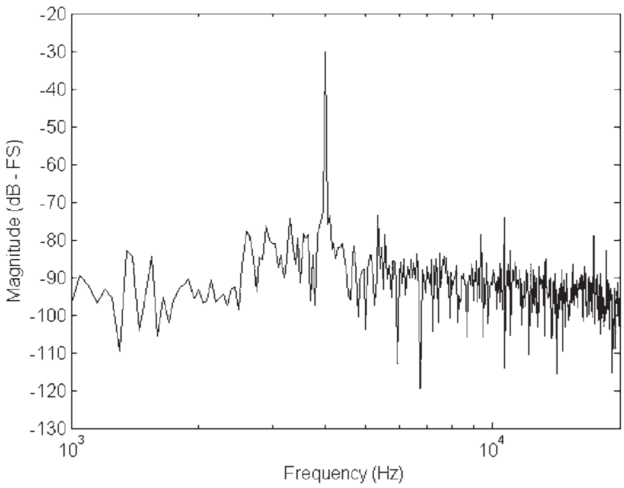


Fig. 7. DLA prototype.

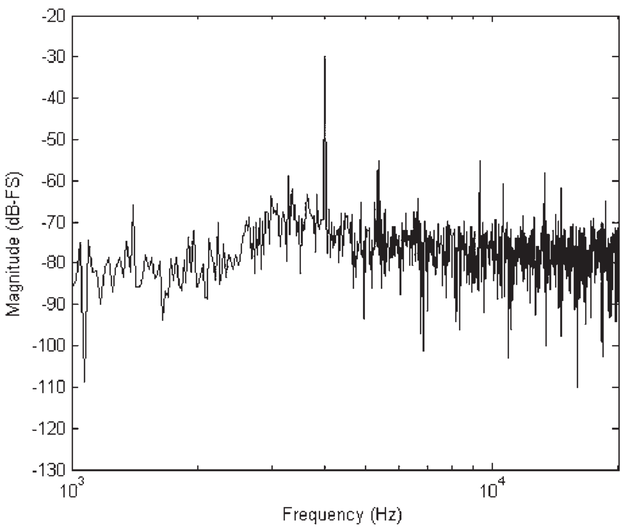
practically finite, a number of other factors, such as the input signal encoding investigated here, may provide significant improvements in DLA performance.

The measurements obtained in this work, namely, single-frequency output, THD, and frequency response, are in accordance with simulation results published previously by the authors. However, the overall performance is slightly inferior to these simulations, possibly because of a number of practical limitations such as the nonideal response and possible mismatch between amplifiers and loudspeakers as well as jitter caused by the FPGA outputs. On the other hand, compared to results obtained by a PCM (multibit) DLA prototype [7], the SDM DLA is shown to achieve lower harmonic distortion, especially for practical off-axis angles and midrange frequencies.

The prototype developed here is modular enough to provide easy reconfiguration. Apart from the amplifier–loudspeaker pairs that are hardwired, the FPGA supports different bit-assignment techniques, which will be implemented in the future, while this system presents no restrictions on the use of different input signals, such as PCM or PWM. Finally the possibility of employing equalization techniques, such as in the 1-bit domain using FPGA

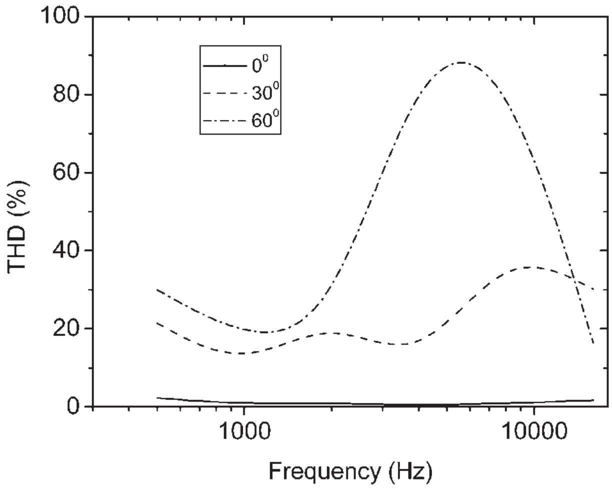


(a)

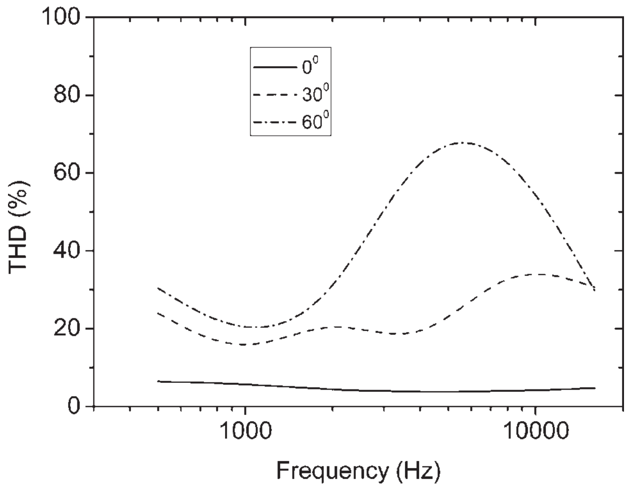


(b)

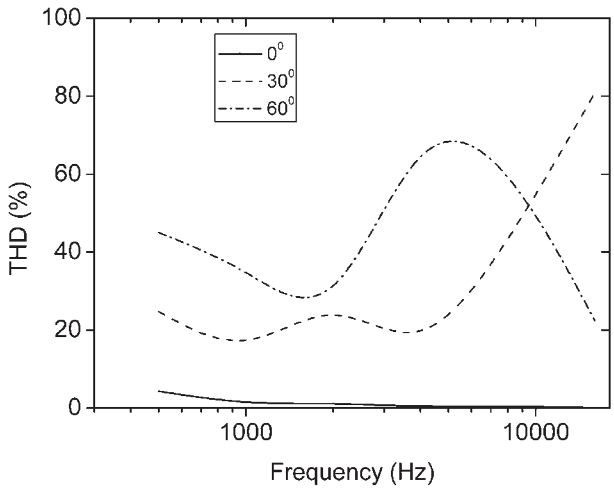
Fig. 10. Sigma–delta pressure spectrum for $f_{in} = 4$ kHz, $r = 2$ m audio band. (a) DLA prototype on-axis measurement $\varphi = 0°$. (b) Off-axis DLA prototype measurement, $\varphi = 30°$.



(a)



(b)



(c)

Fig. 11. THD versus frequency for angles of observation $\varphi = 0°$, 30°, and 60°. (a) Simulated DLA output for sigma–delta input. (b) Measured DLA prototype output for sigma–delta input. (c) Simulated DLA output for 5-bit PCM input (bit-grouped assignment [16]).

resources, should be investigated in order to optimize the LA module reproduction.

5 ACKNOWLEDGMENT

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His research relates mainly to digital processing of audio and acoustic signals, especially for the equalization of room acoustics. He has also worked on perceptually motivated models for such applications, as well as for speech and audio signal enhancement. His recent research interests also relate to aspects of an all-digital audio chain, including WLAN audio, amplification, and direct acoustic transduction of digital audio streams. He has also been active as musician and composer, participating in concerts and recordings.

Dr. Mourjopoulos has authored and presented more than 100 papers in scientific journals and at international conferences, many sponsored by the Audio Engineering Society, where occasionally he was a session chair. For his work he was awarded an AES Fellowship in 2006. He was chair of the AES Greek Section for the period of 1993–1995 and is currently acting as its vice chair, and he is a member of the IEEE and the Hellenic Institute of Acoustics.