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Advances in Digitization of Microphones and Loudspeakers

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The development of digital microphones and loudspeakers adds new and interesting possibilities of their applications in different fields, extended from industrial, medical to consumer audio markets. One of the rapidly growing field of applications is mobile multimedia, such as mobile phones, digital cameras, laptop and desktop PCs, etc. The advances have also been made in digital audio, particularly in direct digital transduction, so it is now possible to create the all-digital audio recording and reproduction chains potentially having several advantages over existing analog systems.

Keywords: electret condenser microphone, digital microphone, CMOS-MEMS digital microphone, digital microphone array, sigma-delta ADC/DAC, digital loudspeaker, CMOS-MEMS digital loudspeaker, digital loudspeaker array.

1. Introduction

The terms "digital microphone" or "digital loudspeaker" are most often used to define the electroacoustic systems which incorporate conventional analog transducers close connected to on-board electronic circuits, such as preamplifiers and analog-to-digital converters (ADCs) or digital-to-analog converters (DACs). Recent developments in digital audio and microelectronics technology open new possibilities to produce the integrated digital microphones and speakers which directly convert analog acoustic signals to digital electric signals, and vice versa, digital electric signals into analog acoustic signals. So, it is now possible to create acoustic systems-on-chip for applications in such areas as hearing aids, in-ear translators, active noise cancellation, and others.

The first part of this paper provides an overview the EC (electret condenser) digital microphones equipped with on-chip preamplifiers and ADCs, as well as fully integrated CMOS-MEMS (complementary metal oxide – semiconductor-microelectromechanical system) digital microphones which already are commercially available. In the following part, the digital loudspeakers which have been offered since a long time on the market are presented. They contain the digital signal processors (DSPs), DACs and amplifiers in addition to analog transducers. Next, the digitally direct driven sound producing system prototypes using conventional speakers and CMOS-MEMS array microspeakers are also reported.

2. Digital microphones

2.1. Electret condenser microphones

The digital microphone converts an input acoustic signal into an equivalent output digital audio signal to further process the information. First conventional, commercial digital microphones systems were developed over 10 years ago by such mainstream manufactures as Beyerdynamic in 1998 (MDC 100 and MDC 101), and in 1999 (MDC 800 series) as well as Milab in 1999 (DM 1001), and Neumann in 2001 (Solution-D, initially with the D-01 and TLM 103 D large-diaphragm microphones) (Becker–Foss *et al.*, 2010). The DM 1001 digital microphone had a special construction. Its directional characteristic was adjustable through the use of analogue acoustic transducers. Signals from dual-membrane capsule were digitized by a stereo ADC.

No standardized interface was available for connecting conventional digital microphones to other equipment before 2000, the supply and interface units had AES3 outputs and own remote controllable attenuation or amplification systems. Later on, the special interface units that supported newly-defined standard AES42-2001 (AES Publ., 2001) were designed. Then the microphone parameters defined according to with the AES42 could be remote-controlled.

After 1998, innovation in conventional digital microphones has concentrated mainly on high-performance portable applications, e.g. for mobile communication, in which miniaturization, integration and lower cost production were main requirements.

Traditional, digital electret condenser microphone (ECM) configuration, built as a low scale integration system, is shown in Fig. 1. Because the signal produced by an ECM is low, preamplification is necessary to supply a proper signal to an ADC. The digital microphone comprises of the EMC, which serves as an analogue acoustic transducer for generating an analog signal representing an acoustic signal, a buffer/preamplifier built using operational amplifiers with included low-pass filter, a sigma-delta analog-to-digital converter ($\Sigma\Delta$ ADC) chip with included a digital decimation filter, a digital high-pass filter for dc removal and a voltage reference as well as a digital audio transmitter (DAT) chip supporting standard audio data format AES-3.

The electret microphone consists of a membrane, a back plate, and an electret layer. The movable membrane and fixed back plate form a variable capacitor. Movement of the membrane with sound pressure results in a capacitance change.

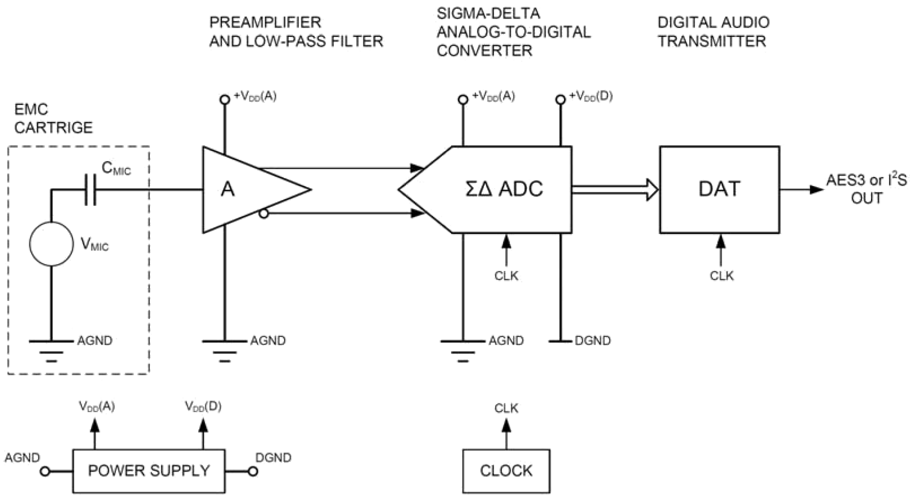

Fig. 1. Block diagram of a traditional digital electret condenser microphone.

Since charge on the capacitor is constant, the voltage across the capacitor varies with the changing capacitance. The change in capacitance, ΔC, with sound pressure, causes a proportional change in voltage, $\Delta V \approx -V\Delta C/C$. So, an electrical model of an ECM consists of a signal voltage source in series with a source capacitor. The ECMs for mobile applications are rather small and their capacitance is also relatively small (typical values are of the order of 3–5 pF or even less) (Yasuno, Riko, 1999; Rhijn, 2003; Nielsen, Fürst, 2007).

Using higher scale integration technology, the EMC high input impedance preamplifier can be implemented with a simple junction field-effect transistor (JFET). But simple JFET-based preamplifiers have relatively large input capacitance which significantly attenuates the signal from the EMC, poor linearity and low accuracy. In addition, the microphone output signal can easily be corrupted by interfering signals affecting in between the preamplifier and the ADC. The problems can be overcome by replacing JFET-based preamplifiers with CMOS analog and digital circuitry. Preamplifiers implemented in modern submicron CMOS processes have enabled some improvements over conventional JFETs. The CMOS preamplifiers offer lower harmonic distortion, easier gain setting, higher noise immunity, multiple functional modes, including sleep mode for low power consumption and enhanced sound quality. However, the design process of the CMOS preamplifiers must take into account the noise considerations. Three dominant noise sources in CMOS preamplifiers for capacitive microphones are flicker (1/f) noise, wideband white noise from the input transistors, and low-pass filtered white noise from an input bias resistor, needed for setting the preamplifier's dc operating point. Not going into the details we may say that the optimum design usually contain some trade-offs between electrical features and technological constrains.

In digital ECMs built as higher scale integration systems, most ADCs are a low-power sigma-delta ($\Sigma\Delta$) type, and usually contains single-loop, single-bit $\Sigma\Delta$ modulator of order greater than one, whose digital output is a single-bit oversampled signal, typically in a pulse density modulated (PDM) version. Using oversampling and noise-shaping operations the $\Sigma\Delta$ ADCs offer several advantages. Noise shaping shifts the quantization noise upwards, pushing much of it outside of the band of interest. Thus, high resolution and high signal-to-noise ratio in the band of interest can be obtained without imposing severe matching requirements for the circuitry. A single-bit $\Sigma\Delta$ modulator is inherently linear and only one of the loop-filter integrators requires severe design constrains. The inner-loop integrators, which have their outputs noise-shaped, have relaxed design requirements. This leads to the lower power consumption. But a potential problem with higher-order $\Sigma\Delta$ modulators (>2) is that they are prone to instability when the input signal exceeds the MSA (maximum stable amplitude). So such modulators fail to return to stable operations when they become unstable due to overload, even when the input is reduced below the MSA. To counter potential instability, a digitally controlled feedback system alters the $\Sigma\Delta$ noise transfer function, forcing the modulator back into stable operation (Nielsen, Fürst, 2007; Le *et al.*, 2009).

The constructions of the digital ECMs have been changing over the years. The first innovation had a benefis of surface-mount technology which could be incorporated into the microphone body instead of into an external unit. Next step was developing the integrated circuit (IC) preamplifiers which replaced discrete mounted preamplifiers inside the ECMs. Farther innovation was to design complete IC including a preamplifier and an ADC inside the ECM. Using the IC technology it is a natural way to perform the ADC function inside the ECM. Thus, incorporating ADC into the microphone itself provides a digital output that is inherently less prone to corruption by interferers.

Typical cross-section of digital ECM is shown in Fig. 2. It usually has four connection pins which are V_{DD} (power supply), GND (ground), CLK (clock), and

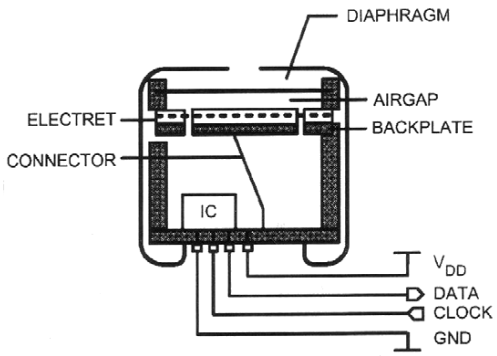

Fig. 2. Cross section of digital ECM (from *National*, 2009).

DATA (bit-stream data). In most cases the digital decimation and other filtering circuits are not built into the microphone capsule. The DSP module which follows the microphone should provide the clock signal and the digital signal (bit-stream data) which forms the input to the DSP module's digital decimation filter (low-pass filtering and down-sampling), and further processing of audio signals. The preamplification and $\Sigma\Delta$ A/D conversion is usually fixed in gain and sample frequency, while the digital filtering is adaptable to enable software customization and real time adaptation. The additional digital microphone functions can include, for example, test mode which enables access to various internal nodes in the circuit, as well as, power-down mode (sleep mode) allowing the system input clock frequency to dropping down, what lowers the current drawn by the system and provides maximum battery life by conserving power when the microphone is not needed.

Actually, there are several ICs for the ECMs offered by leading chipmakers available on the market. For example, ADAU 1301 microphone preamplifier (Nielsen, Fürst, 2007; *Analog*, n.d.), LMV 1022/23 and LMV1024/26 (Hennink, 2008; *National*, 2009) which consist of the preamplifiers and $\Sigma\Delta$ ADCs for placement inside the ECMs. The digital microphone function can be supported by AIC 33 device (Fang, Toth, 2005). Digital signal from an ECM is fed to AIC 33 where it is filtered and down-sampled by the digital decimation filter, and the digital output is provided to an external processor over the audio data serial bus. For example, each mentioned above LMV1024/26 stereo chip has an integrated 20 kHz preamplifier and a $\Sigma\Delta$ ADC with PDM output. It has the following features: supply voltage from 1.6 V to 3 V (1.8 V typ.), supply current 0.5 mA, SNR A-weighted 59 dB, clock frequency 400 kHz to 2.4 MHz (1.2 MHz typ.), THD 0.03%, power supply rejection ratio (PSRR) 100 dB. The sensitivity of the digital microphone incorporating LMV1024 is the sensitivity of a conventional EMC plus the input to output transfer of the LMV1024. The sensitivity of a typical digital ECM is therefore: $-44 + 15.2$ (gain of preamplifier) $= -28.8$ dB (FS/Pa) (*National*, 2009).

Among more recent developments in the field of professional (studio) digital microphones it is worth to mention small-diaphragm microphones from Neumann and Schoeps. Neumann extended Solution-D system with the KM 183/184/185 D microphones having three different directional characteristics (*Neumann*, n.d.). The DSP functions integrated into the microphones can be configured and controlled remotely via the DMI-2 digital microphone interface and the RCS remote control software. Schoeps, which also has long experience in the development and manufacture of digital microphones, introduced modular “Colette” microphone series with the ability to operate digitally, and the SuperCMIT shotgun type digital microphone, designed for high quality music recording (*Schoeps*, n.d.). These digital microphones support Schoeps digital amplifier CMD 2. Most currently available digital microphones for the professional audio (except of mobile telephones) are based on the new AES 42-2006 standard (AES standard for acoustics

– digital interface for microphones) (*AES Publ.*, 2006). That standard is an extension of the AES3 audio interface standard.

2.2. CMOS-MEMS microphones

The CMOS-MEMS digital microphones are similar to the standard digital ECMs except that the components are built onto a single chip using CMOS technology (material deposition and etching), rather than assembled from discrete parts. The CMOS-MEMS digital microphone is often called as a "microphone on a chip" with the acoustic pressure sensitive membrane being etched directly onto the silicon, and usually accompanied by a matched preamplifier and as well by a $\Sigma\Delta$ ADC on the same chip. Because the audio signal is immediately converted by integrated electronics to the digital output signal, typically in pulse density modulation format (PDM), the system has the advantage to be more robust to noise than its analogue alternative.

Silicon monolithic microphones exhibit improved aspects over ECMs such as compatibility with standard fabrication and assembly procedures, reduced size, low-power consumption, higher immunity to mechanical shocks and low temperature coefficient. In addition, the ECMs cannot withstand high temperatures and therefore cannot go through a standard high volume surface mount manufacturing flow. On the contrary MEMS microphones can withstand temperatures to about 250°C with no degradation in performance (Neuman, Gabriel, 2003).

Many types of small-sized microphones have been constructed during the last twenty years of research and development. Most of them fall into three categories of MEMS microphones as piezoelectric, piezoresistive, and capacitive-type. Traditional "condenser" or capacitive microphones show the highest sensitivity while maintaining a low power consumption. The diaphragms can be made, for example, of metal, p^+ doped silicon, silicon nitride, polyimide, and others. Many successfully microphone designs use polysilicon diaphragm which moves in response to acoustic pressure variations. The use of silicon as the diaphragm material is profitable because of its low intrinsic stress which determines diaphragm sensitivity and its resistance to warpage.

The key part of the MEMS microphone is the variable capacitor formed by a fixed backplane and a flexible diaphragm. Sound pressure deflects the diaphragm, causing a change in capacitance which next causes a change in voltage. The cross-section of the polysilicon diaphragm condenser microphone is shown in Fig. 3. Its use low-stress n^+ polysilicon as the diaphragm electrode and p^+ etch-stop silicon plate as the back plane electrode. The device consists of an n-type silicon substrate, a phosphorus doped polysilicon diaphragm, a p^+ perforated back plane, and the metal contacts. Any bonding techniques are not required. The performance of the microphone depends on the size and stress of the diaphragm. Other parameters, such as air gap distance and the bias voltage, also affect the sensitivity (Hsu *et al.*, 1998).

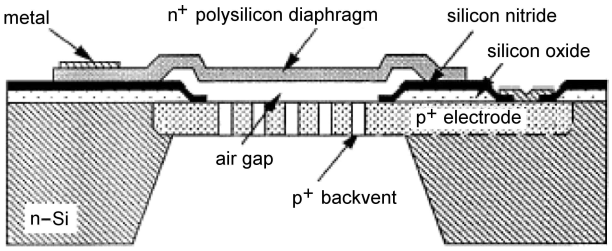

Fig. 3. Cross-section of the polysilicon diaphragm condenser microphone (from Hsu *et al.*, 1998).

Typical features of the monolithic CMOS-MEMS (mono or stereo) digital microphone are as follow: omnidirectional characteristics, PDM single-bit output, $4 \times 4 \times 1$ mm (typ.) package size, 1.5–4 V supply voltage range, 55–60 dBA signal-to-noise ratio (SNR), -25 to -30 dB full scale (FS) sensitivity, -55 to -65 dB power supply rejection (PSR), 100 Hz to 15 kHz frequency response, 1–4 MHz clock frequencies and very-low power standby (sleep) mode.

The silicon microphones not only can be useful for those applications where digital ECMs are applied (for example, mobile phones, hands-free phones, MP3 players and laptops), but can be considered as candidates in novel applications such as microphone sensor-arrays for enhanced acoustics performance. For example, microphone arrays are used in distant speech recognition in which the speech is captured by multiple distant microphones, typically in an array configuration. The very small size of the MEMS digital microphones allow for placing multiple transducers in a small area to create a microphone array.

The small MEMS digital microphones are now commercially available for high-fidelity audio applications, for example, T 4030 from DK-EPC (group company of TDK Corporation) (*Epcos*, n.d.), WM7210/20 (*Wolfson*, 2010), ADMP421 and ADMP441 (*Analog*, n.d.) and AKU2002C (Fig. 4) and AKU1126 (*Akustica*, 2010).

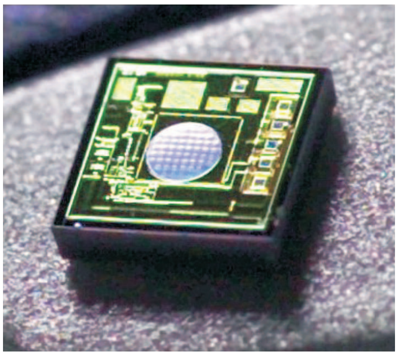

Fig. 4. CMOS-MEMS (1×1 mm die) digital microphone AKU2002C (from *Akustica*, 2010).

3. Digital loudspeakers

3.1. Conventional loudspeakers

The widespread availability of digital audio sources and digital amplifiers provides an opportunity to remove traditional analogue crossover from the signal path and replace it by digital crossover. First loudspeakers named "digital" for marketing purposes which, obviously, could not reproduce sound digitally, have been introduced on the audio market by Meridian Audio Company in 1993. This so-called conventional digital loudspeaker does not look much different from a conventional analogue one, except that inside in addition to analogue drivers contains the digital crossover, DACs, power amplifiers, and power supply. Sometimes, the digital preamplifier can also be included to the speaker's cabinet. Such digital loudspeakers accept a digital input signal and implement the crossover in the digital domain with digital signal processor (DSP). Instead of subjecting the audio signal to resistors, inductors, and capacitors, the DSP crossover separate the frequency spectrum performing mathematical processing on the digital audio data. The DSP crossovers can have a very good time behavior performance, desired slope and frequency without regard for component limitations or tolerances.

The block diagram of a 3-way, conventional digital loudspeaker system is shown in Fig. 5. The speaker accepts a digital input signal from digital source. DSP system inside the speaker split the frequency spectrum into bass, midrange,

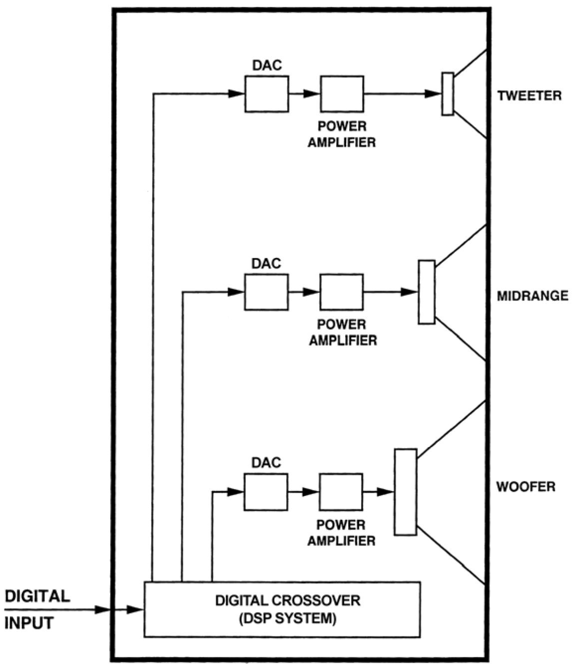

Fig. 5. The 3-way conventional digital loudspeaker system.

and treble. This system can also equalize and delay signals to produce proper acoustic behavior at the analogue driver output. Each of the three digital signals is then passed through a DAC and power amplifier specially designed to power the particular driver. It is a big advantage of digital loudspeaker because the power amplifiers are designed to drive a known load. Also digital loudspeakers eliminate need of the monoblock power amplifiers in a playback music system.

Now there are a lot of different models of the conventional loudspeakers commercially available. Some of them are equipped with remote control systems to adjust chosen parameters.

3.2. Novel loudspeaker prototypes

Most significant developments in loudspeakers and microphone designs were described two years ago in the study by Rumsey (Rumsey, 2009). Considering direct conversion of digital audio data into air movement, all processes (from the input to the output of the loudspeaker system) should be performed digitally. Interesting approaches have been described in two AES Convention papers (Ueno *et al.*, 2006; Saito *et al.*, 2008).

In the first paper (Ueno *et al.*, 2006), the proposition of digital-driven piezoelectric speaker system employing multi-bit $\Delta\Sigma$ modulation (DSM), named also as $\Sigma\Delta$ modulation (SDM), is presented. The effectiveness of the proposed speaker system was tested by a prototype system. Its block diagram is shown in Fig. 6.

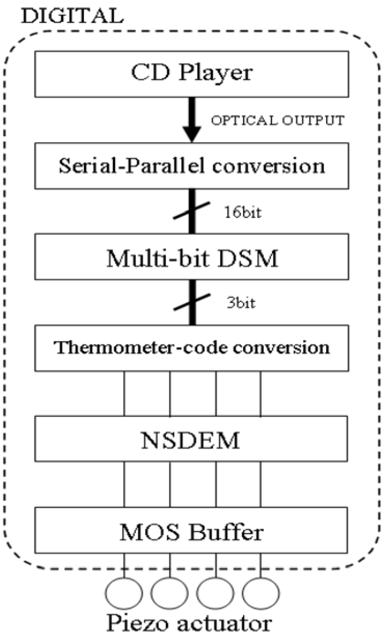

Fig. 6. Block diagram of the proposed digitally driven speaker (from Ueno *et al.*, 2006).

The PCM serial signal from the CD player is converted into a parallel signal. The 16 bits of the input digitized data is reduced to 3 bits by the multi-bit DSM. Next, this output binary signal is converted into an 8-bit thermometer code. Since the thermometer code constitutes equally weighted bit data, which is 1 or 0, several sub-speakers can be driven on-off by this code. To improve the accuracy of the internal DAC, the noise-shaping dynamic element matching (NSDEM) unit has been used. Furthermore, the on-off signal is amplified by passing it through a buffer of MOS transistors in the output stages. Finally, the output signals from each sub-speaker are mixed in the air and music is reproduced. Since the output signal is mixed in the air, each characteristic has sum of the signal value. In this case, eight piezoelectric loudspeaker subunits (Fig. 7) have each differential value, and the loudspeaker unit does not output the original signal. And also, the power gain is quite low. But by the experimentation and simulation it was possible to check that digital sound data was converted into an analog acoustic signal.

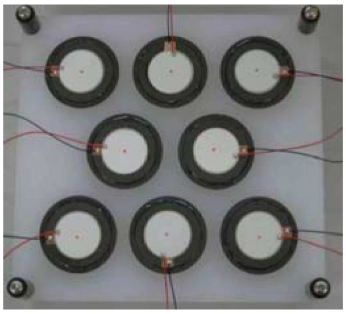

Fig. 7. Photograph of piezoelectric loudspeaker unit (from Ueno *et al.*, 2006; Saito *et al.*, 2008).

In the second paper (Saito *et al.*, 2008), the previous used piezoelectric speaker is replaced by a loudspeaker with multiple voice coils (Fig. 8) and a digital system (Fig. 9). The multiple voice coils in the prototype digital dynamic speaker are directly driven by the digital signals. The voice coils are driven by the mismatch shaper circuit in order to reduce the mismatch effects caused by

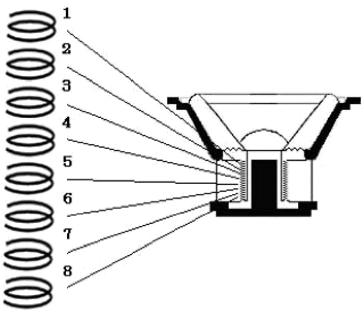

Fig. 8. Structure of dynamic speaker (from Saito *et al.*, 2008).

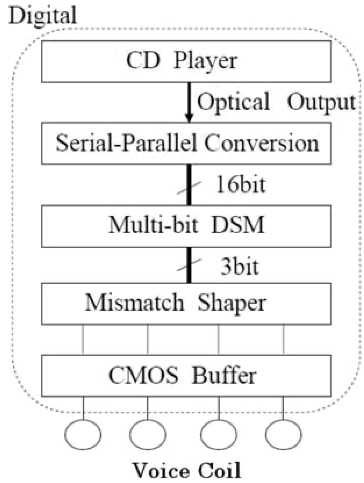

Fig. 9. Block diagram of digital speaker with mismatch shaper (from SAITO *et al.*, 2008).

position of the voice coils. The mechanical displacement of a diaphragm (voice coil) is controlled depending on the net magnetic flux generated by the voice coils. The mismatch in the magnetic flux generated by the voice coils is small and noise caused by the mismatch is reduced by the mismatch shaper circuit.

As compared to the output power obtained with the piezoelectric-type loudspeaker, the output power of the dynamic speaker can be easily increased. Since multiple voice coils are used, the output power is multiplied by a factor equal to the number of coils. The output power of 1 W can be realized with a 1 V power supply by using the proposed digital system (a 25 cm drive unit was used with the impedance of the voice coil of 4 Ω). The measured output spectrum of digital system is shown in Fig. 10a. The THD and SPL of the digital speaker are 0.1% and 104 dB, respectively. The frequency response of digital speaker is shown in Fig. 10b.

In the next paper (HUSNIK, 2008) it is shown that it is possible to create miniature electrostatic or condenser transducers on a silicon chip that use direct digital-to-analog conversion. In conventional condenser transducers, the back plate forms a single electrode that gives rise to an evenly distributed electrostatic force between it and the membrane, as shown in Fig. 11a. In case of digital condenser loudspeaker (transducer with the direct DAC) the back electrode is partitioned into areas corresponding to different bits, and these are distributed over the surface in one of many possible patterns (Fig. 11 b and c).

The force acting on the membrane is composed of contributions from each of these areas. The areas in the center will have a larger effect than those near the edges where the membrane is clamped. To improve the function of the transducer the correction factors can be used to account for the different contribution weights of electrodes at different points across the back plane. It is proposed that when

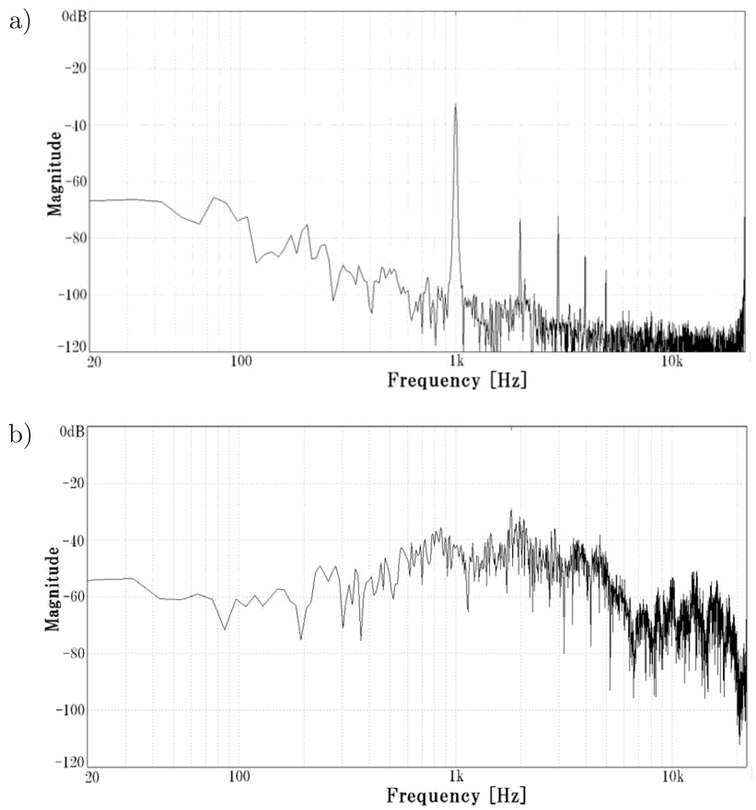

Fig. 10. Output spectrum of digital system with 1 kHz sine wave at the input (a); output spectrum of digital speaker with white noise at the input (b) (from Saito *et al.*, 2008).

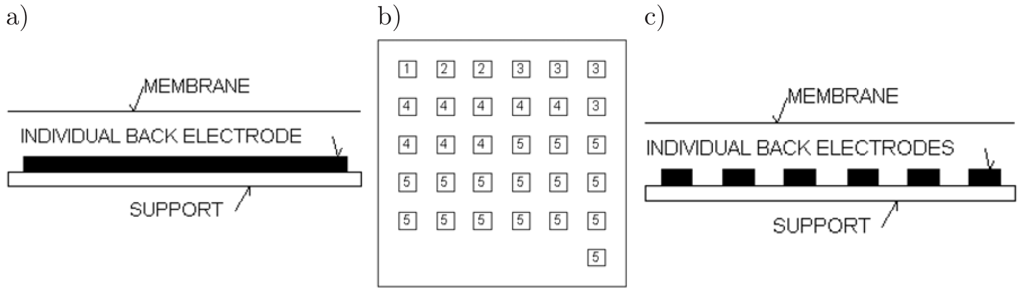

Fig. 11. a) Arrangement of electrodes in a conventional condenser single-acting transducer; b) example of partitioning the back electrode in a 5-bit digital transducer (with direct digital-to-analog conversion); c) partitioned back electrode in a 5-bit digital transducer (from Husnik, 2008).

there are multiple electrodes responsible for a particular digital bit, the correction factor should be applied to the whole group so the effect is the same as if they were evenly distributed around the center of the back plate.

The direct-converting, digitally driven loudspeaker array prototypes with moving-coil transducers have been developed, as reported by Mendoza–Lopez *et al.* (2007). Direct conversion is achieved by superposition of the acoustic waves which represent the different bits of the input signal. In the paper, the theoretical evaluation of the radiated sound-field quality is presented. The effect of key experimental factors such as transducer mismatches, array size, transducer interspacing, baffle diffraction, and transducer arrangement on the radiated sound-field properties was isolated and quantified experimentally. Three types moving-coil tweeters characterized by different electromechanical parameters were used for the prototype implementation, and two different digital transducer array (DTA) topologies were built. In the first one, one transducer was assigned to each bit of the PCM signal. Binary weighting was carried out in the electronic domain. Both line and circular array prototypes were built with one transducer per bit, both with tree types transducers (Fig. 12a and b). The second array topology constructed was the concentric-circle bit-grouped. A total of 63 transducers were set on semicircular baffle forming concentric rings with binary-weighted transducer numbers for each ring (Fig. 12c).

a) b) c)

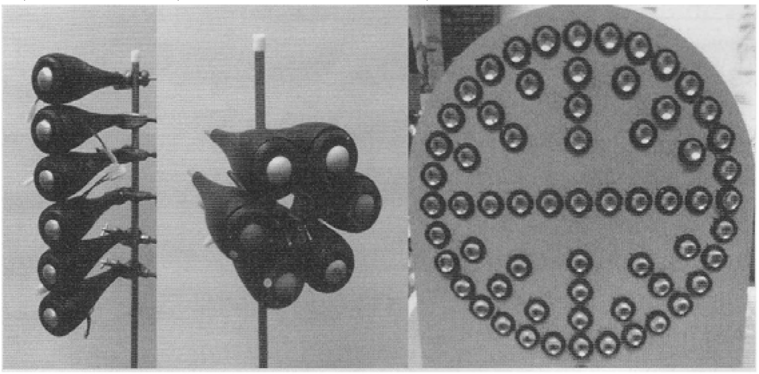

Fig. 12. 6-bit prototype digital transducer arrays (DTAs); a) line array, one transducer per bit; b) circle array, one bit per transducer; c) bit-grouped concentric circle DTA (from Mendoza–Lopez *et al.*, 2007).

The transducers were wired in different series-parallel combinations, depending on number of transducers for each bit. The control PC, a DSP system, the class AB voltage amplifiers, a transducer array under test placed in anechoic chamber as well as measurement microphone, a preamplifier and a PC, were included in the experimental setup. The quality of radiated sound field has been explored through THD, frequency response, directivities, and swept-sine responses, and related to the fundamental properties of the constitutes transducers. Omitting detailed results of measurements, it is worth to cite one of the final conclusions. The implementation of all-digital direct-converting systems currently needs transducer technology to evolve to produce smaller and more efficient transducers if the harmonics of the bit streams are to be radiated.

The research on digital loudspeakers which has been conducted over last years was mainly focusing on the direct acoustic transduction of multibit PCM signals. The bit significance of each binary word was translated into the appropriate acoustic gain via the activation of groups of emitting elements (bit grouping). In most cases the transducers had a form of a digital loudspeaker array (DLA). For mapping 16-bit PCM audio signals to the transducers the large number of loudspeaker are needed. In addition, for a practical bit-grouped PCM-driven DLA realization the sophisticated software implemented in digital signal processors (DSPs) and field-programmable gate arrays (FPGAs) is necessary.

The paper of Tatlas *et al.* (2009) describes design, construction and measurements of a digital transducer array prototype, driven directly by 1-bit sigma-delta modulated (SDM) audio stream. The prototype includes a DSP stage implemented via a FPGA, a digital amplification module, and a two-dimensional array consisting of miniature electrodynamic loudspeakers, positioned on a surface to form a DLA (Fig. 13). 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 divided into two categories: binary DLAs (PCM driven) and unary DLAs (sigma-delta driven), as realized and tested in the Tatlas *et al.* paper.

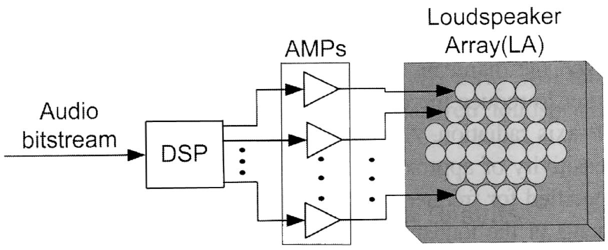

Fig. 13. Typical structure of a DLA (from Tatlas *et al.*, 2009).

The mapping of any 1-bit sigma-delta input signal within the DLA has been explained and documented in previous work of the cited authors (Tatlas, Mourjopoulos, 2004). 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 (array of loudspeakers). The frames thus produced are digitally amplified and fed directly to the LA. The DLA implemented in the describing work consists of 32 transducers, arranged in a circular layout on a two-dimensional flat surface. The unary input signal was used for tests. Thus assuming that all transducers have equal significance and are given an index number k ($1 < k < 32$), as shown in Fig. 14a. Hence when the LA is driven by 32-bit sigma-delta frame, this will be directly mapped to the LA elements. The generated activation pattern example is shown in Fig. 14b. The electronic system of

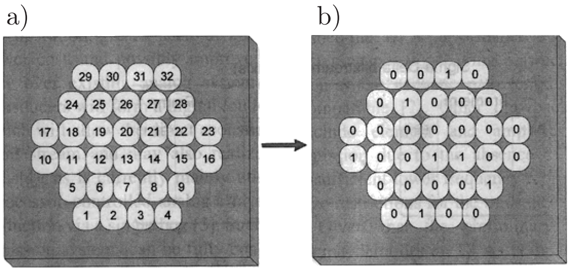

Fig. 14. Typical two-dimensional circular 32-element arrangement and sigma-delta activation pattern; a) 32-element LA indexing; b) 32-bit sigma-delta frame activation pattern (from Tatlas *et al.*, 2009).

the DLA prototype consists of an $\Sigma\Delta$ ADC, an FPGA chip in which the developed algorithm of the DLA bit-grouping system is implemented, and 32 digital amplifiers is shown in Fig. 15. The DLA was developed using 32 miniature moving-coil (electrodynamic) loudspeakers, each approximately 15 mm in diameter.

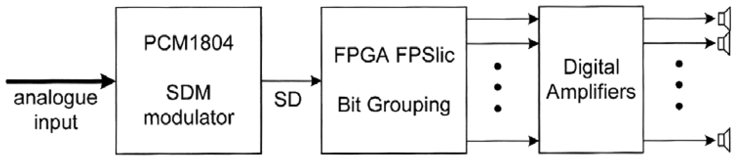

Fig. 15. Architecture of sigma-delta DLA prototype (from Tatlas *et al.*, 2009).

As an example of the obtained measurement results, 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. 16 (in an arbitrary decibel scale). Other results of measurement, as single-frequency output,

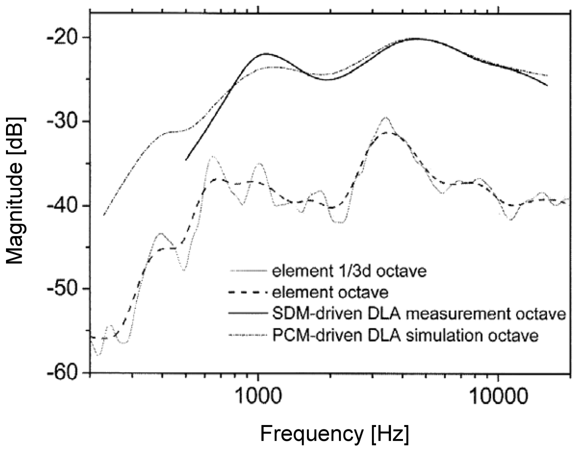

Fig. 16. On-axis frequency response for one loudspeaker driven by analog excitation (one-third-octave and octave smoothing) and DLA output driven by SDM and PCM streams (octave smoothed). Top curves are shifted vertically for clarity (from Tatlas *et al.*, 2009).

THD, and frequency response are well documented. The main source of distortion for the DLA acoustic emission comes from the different path lengths between element on the array and listening position. Compared to results obtained by a PCM (multibit) DLA prototype (Mendoza–Lopez *et al.*, 2007), the SDM DLA achieves lower harmonic distortion, especially for practical off-axis angels and midrange frequencies.

3.3. CMOS-MEMS loudspeakers

The CMOS-MEMS process provides relatively simple, cost effective and scalable means for fabricating monolithic, integrated microelectromechanical microphone systems (as mentioned earlier) and loudspeaker systems. A method of a direct digital sound producing using an array of CMOS-MEMS digital microspeakers has been proposed (Diamond *et al.*, 2003). Digital sound reconstruction (DSR) involves the summation of discrete acoustic pulses of energy to produce time-varying pressure waveform, as shown in Fig. 17.

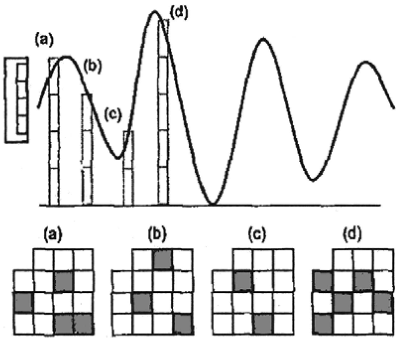

Fig. 17. Illustration of the concept of DSR with the hypothetical 15-speaker (4-bit) chip. An idealized sound pulse is generated from the motion of a single speaker binary motion (shadowed squares). Multiple speakers binary motions at different times create a sound waveform (from Diamond *et al.*, 2003).

The discrete acoustic pulses are based on a digital signal coming from digital audio source in which each signal bit controls a group of speakers. The n-th bit of the incoming digital signal controls 2^n speakers in the array, where the most significant bit (MSB) contains 2^{n-1} speakers and the least significant bit (LSB) contains a single speaker. When the signal for a particular bit is high, all transducers in the group assigned to the bit are activated for that sample interval. The number of speakers in the array and pulse frequency will determine the resolution of the resulting waveform. Through the post-application of an acoustic filter from the human ear, the listener will hear an acoustically smoother signal identical to the original analog waveform.

A digital array of speakers was required to implement in practice direct digital sound reconstruction. The prototype of an 8-bit digital speaker chip with

255 square speakers, each speaker 216 μm on a side, has been designed and fabricated using CMOS-MEMS process. A variety tests of the developed array of microspeakers have been performed showing the feasibility of digital reconstructed sound but also a significant amount of harmonic distortion. In further developments in microspeaker design, the speaker uniformity and linearity improvements, as well as better array drive and reconstruction algorithms are needed to reduce harmonic distortion and to prove that the full potential of direct digital sound reconstruction can be utilized.

4. Conclusion

In this paper the recent advances in the area of digital microphones and digital loudspeakers are briefly reviewed. In case of the digital microphones, the main goal of improved design for mobile communication application is to integrate preamplification with analog-to-digital conversion, increasing dynamic range through improved linearity and lower noise, while retaining very low power consumption. Some commercial applications need inexpensive, low grade microphones where size and performance are not critical parameters. In portable applications, such as mobile phones, headsets, digital cameras, laptops, etc., there is a need of compact, low size, and high quality microphones. Now, when CMOS-MEMS is introduced to the small electroacoustic devices to increase the number of elements, the direct-converting digital microphones, as well as the microphone arrays, become more popular in many applications.

In case of the digital loudspeakers research trends have addressed new digital loudspeaker arrays (DLAs) using piezoelectric and traditional moving-coil transducers which allow the direct acoustic emission of the audio bit stream and may also realize controlled directivity. However, up to now only loudspeaker array prototypes have been developed. It seems that CMOS-MEMS loudspeaker technology may offer the best practical solution in form of microspeaker arrays.

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