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A Novel Universal-Serial-Bus-Powered Digitally Driven Speaker System with Low Power Dissipation and High Fidelity

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ABSTRACT

We propose a novel digitally driven speaker system in which a newly devised mismatch shaper method, multilevel noise shaping dynamic element matching, is used to realize high fidelity, high sound power level, and low power dissipation. The unit used for the mismatch shaper method can easily increase the number of sound pressure levels with the aid of an H-bridge circuit, even when the number of sub-speakers is fixed. Further, it reduces the noise caused by quantization and speaker mismatches and decreases the switching loss.

The output sound power level equipped with six voice coils is 94 dB/m when a 3.3-V universal-serial-bus power supply is used exclusively. The power efficiency is 95% at 0 dBFS and 75% at -10 dBFS.

1. INTRODUCTION

At present, a sound signal is recorded as digital data, as in the case of CDs and MP3 files, while in digital audio systems, speakers are still driven by analog signals.

A block diagram of a conventional audio system is shown in Fig.1. Digital audio data is converted to analog signals by using a digital-to-analog converter (DAC). These analog signals are then amplified by an analog amplifier. The output of the amplifier is passed through an analog filter, and the output of the filter is outputted from the analog speaker.

Compared to the components of a digital circuit, these analog devices are usually heavy and also large at the circuit scale. In addition, high-efficiency is required in most analog circuits. Therefore, a class-D amplifier is becoming popular because of its ability to reduce power dissipation and amplifier weight. However, a digital class-D amplifier requires a high-frequency clock signal in the region of 100 MHz Further, a class-D amplifier using a digital delta-sigma modulator (DSM) requires a high oversampling ratio and its out-of-band noise is large.

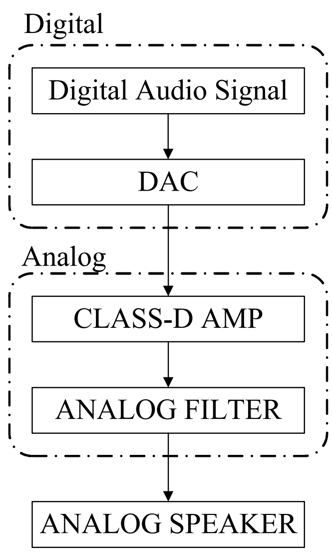

Figure 1: Block diagram of a conventional audio system.

If the speaker system can be directly driven by a digital signal, the DAC and the heavy analog amplifier can be removed from the audio system. We have previously proposed a digitally driven speaker system (DDSP) [1]–[3]. The advantages of the DDSP are as follows.

- All the electrical systems in the speaker employ digital processing (analog circuits are not necessary).
- The speaker units are driven by an ON-OFF signal.
- The sound pressure level (SPL) can be increased by using several speakers.
- The out-of-band noise generated by the DSM can be reduced by using a multibit DSM.

In this paper, we propose a novel DDSP in which a newly devised mismatch shaper method, multilevel noise shaping dynamic element matching (ML-NSDEM), is used so as to realize high fidelity, high SPL, and low power dissipation.

2. CONVENTIONAL DIGITALLY DRIVEN SPEAKER SYSTEM

A block diagram of the conventional DDSP is shown in Fig.2. The digital data are converted into multibit signals by the multibit DSM. The quantization noise resulting from the reduction in the bit length is shifted towards higher frequencies. The output data from the DSM is converted to equal-weighted signals by a binary-to-thermometer code converter. The signals pass through the H-bridge driver and then drive the speakers. The output signals from each speaker are mixed in air and the original sound is reproduced.

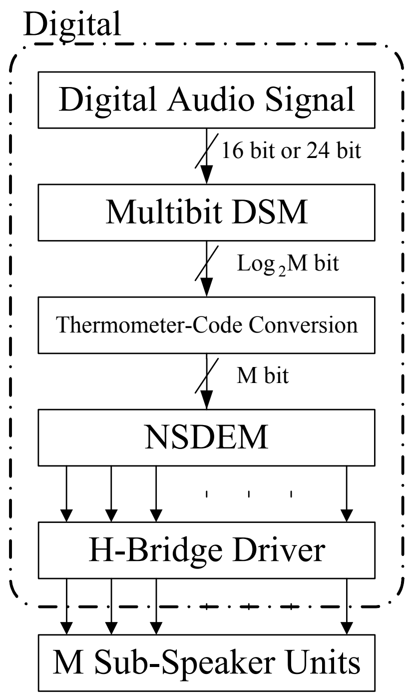

Figure 2: Block diagram of the conventional DDSP.

However, actual devices exhibit a performance mismatch that is caused by a manufacturing defect. There are mismatches in speakers; the sound quality is degraded due to the nonlinearity caused by the mismatch. In order to improve the sound quality, noise shaping dynamic element matching (NSDEM) is carried out between the thermometer-code conversion and H-bridge driver stages. NSDEM shifts the noise caused by the mismatches to higher frequencies.

2.1 Delta-sigma modulator

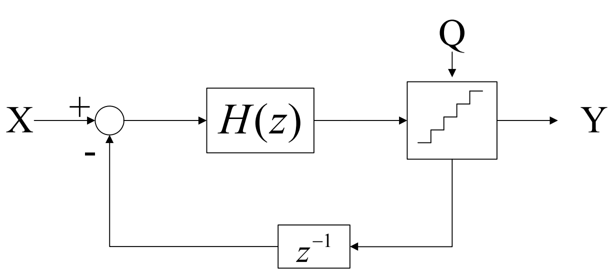

Figure 3: Block diagram of DSM.

The block-diagram of a DSM is shown in Fig.3. The DSM is composed of a loop filter, $H(z)$, and a quantizer. The input signal enters the loop filter and is quantized. The quantized signal passes through the lower part of the loop and is subtracted from the input. Since the difference between the input and the output is the error, only the error is filtered. If $H(z)$ is n-th-order integrator, the transfer function is

$$Y = X + (1 - z^{-1})^n Q \qquad (1)$$

where Y is the output signal, Q is the quantization error, and X is the input signal.

In the equation, X is directly transmitted as the output and the quantization error Q has a noise-shaping effect resulting from its passing through the loop filter. Noise in the audible range is reduced by noise shaping. In addition, a highly precise conversion at less than 100 times the sampling frequency can be realized for the signal bandwidth. In the DDSP, we use a multibit third-order DSM. The noise in the signal band is reduced by the noise-shaping effect.

2.2 NSDEM

If a noise shaping can be performed on the noise caused by the device mismatch, the noise can be decreased further. An NSDEM method [4] is used in the DDSP to reduce the noise caused by the device mismatch.

A block diagram of the NSDEM method is shown in Fig.4. The input signal enters the loop filter, which integrates the number of elements in each choice. The integrated signal is to be sorted in order from the smallest value. The output codes are determined by arranging the values of the loop filter output in increasing order of magnitude, and they are used to maintain a constant difference among the filter outputs.

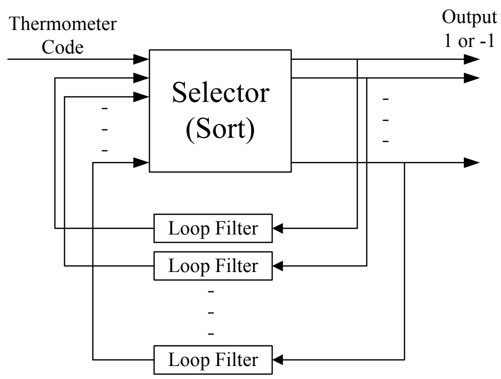

Figure 4: Block diagram of the NSDEM method.

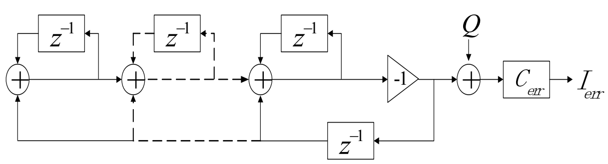

Figure 5: Equivalent circuit of NSDEM.

The equivalent circuit of NSDEM is shown in Fig.5. The noise transfer function is calculated by using equation (2) below, where I_{err} is the output error and C_{err} is the magnitude of mismatching among unit elements.

$$I_{err} = (1 - z^{-1})^n C_{err} Q \qquad (2)$$

This equation shows that the NSDEM method can shape the noise caused by mismatches among the speakers as DSM.

The output of the NSDEM circuit is a 1-bit digital signal for each element. The speakers are driven according to the output signal from the circuit, which is +1 or -1. Therefore, there is no action to stop the speaker; all speakers have to be moved at low output power.

2.3 H-bridge

The driver circuit can be easily realized by using CMOS inverters in the case of single-end implementation. However, because of the supply voltage limit, it is difficult to raise the output sound pressure. A higher output voltage can be obtained by using a H-bridge. The circuit diagram of the H-bridge is shown in Fig.6 and the switch operation is shown Table 1.

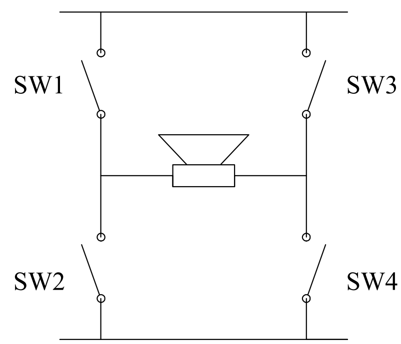

Figure 6: Circuit diagram of H-bridge.

	SW1	SW2	SW3	SW4
1	ON	OFF	OFF	ON
0	OFF	ON	OFF	OFF
-1	OFF	ON	ON	ON

Table 1 Switch operation

2.4 Disadvantages of DDSP

The disadvantages of the DDSP are listed below.

- Similar to digital class-D amplifiers, the speakers must be switched at low output power. The power efficiency when music is reproduced is the same as that of a digital class-D amplifier.
- To reduce the quantization noise, it is necessary to increase the number of speakers. However, because of the size of the speaker and hardware, there is a limit on the number of speakers in practice.

In this paper, to overcome these problems, we propose a novel DDSP in which a newly devised mismatches shaper method, ML-NSDEM, is used.

3. PROPOSED DDSP

A block diagram of the proposed DDSP is shown in Fig.7. The system shorts the speakers in order to represent zero level. By using the ML-NSDEM, the output sound pressure can be controlled such that it belongs to one of three levels: +1, -1, or 0. Therefore, the total number of levels of the DSM can be increased.

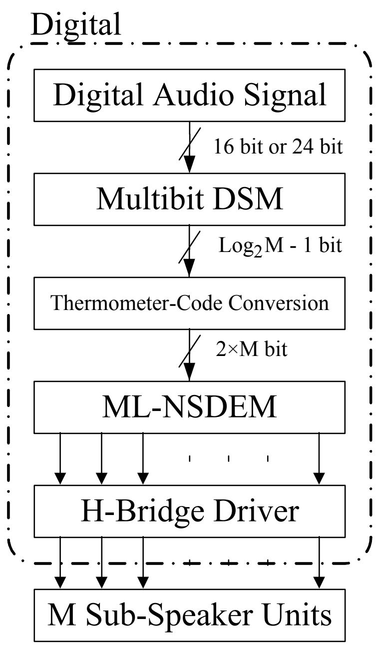

Figure 7: Block diagram of the proposed DDSP.

3.1. ML-NSDEM

A block diagram of the ML-NSDEM is shown in Fig.8. Although the ML-NSDEM circuit is obtained by a slight modification of the NSDEM circuit and three-level DWA [5], it can drive the speaker at the zero level.

There are several advantages in using ML-NSDEM:

- The power efficiency can be improved when the output power level is lower than the full-scale level. This is because the ML-NSDEM unit drives only one sub-speaker at a time when the amplitude of the input signal is small.
- The total number of levels of the DSM increases. This reduces the quantization noise by 6 dB. Moreover, it relaxes the stability condition of the delta-sigma loop. Since a loop filter with a high gain can be used, the noise shaping characteristics are greatly improved.

Thus, by using the three-level driving technique in combination with a high-gain loop filter and ML-NSDEM unit, the sound quality of the DDSP can be improved.

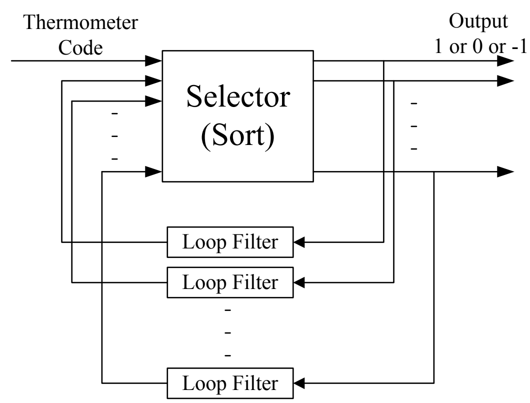

Figure 8: A block diagram of the ML-NSDEM method.

3.2. Low-power Operation

If a universal-serial-bus (USB) power supply is used, high sound pressure cannot be obtained because of the low-voltage power supply. In addition, the USB has a maximum current limit of 500 mA. The proposed DDSP has extremely high efficiency compared to the conventional class-D amplifier at low output power. Thus, the average current is reduced during the playing of music. A high SPL can be realized by USB-powered operation since the DDSP uses several sub-speakers or a dynamic speaker with multiple voice coils. The output power increases in proportion to the number of sub-speakers and voice coils.

4. MEASUREMENT RESULT

The prototype system is implemented using an FPGA and H-bridge circuits on a printed circuit board. A block diagram of the prototype is shown in Fig.9. The digital audio signal from the USB is converted into 16-bit LR channel PCM data, which are inputted to an interpolation digital filter in order to increase the sampling frequency of the data and attenuate an image signal. The up-sampled signal is inputted to a DSM so that the output bit length driving the speakers is reduced to a few bits.

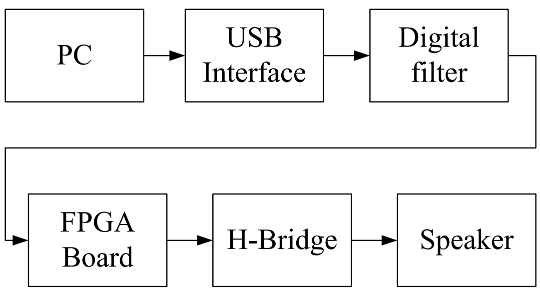

Figure 9: Block diagram of the prototype

The speaker is shown in Fig.10; it is a multiple-voice-coil speaker.

The output power spectrum in the absence of NSDEM is shown in Fig.11. The output power spectrum when ML-NSDEM is carried out is shown in Fig.12. The input signal frequency and the SPL are 1 kHz (sine wave) and 94 dB/m, respectively. When ML-NSDEM is carried out, the noise floor is reduced by about 10 dB. In addition, the harmonic noise is reduced from -20 to -40 dB. The power efficiency against the input level is shown in Fig.13. It can be observed that the power efficiency is not decreased at low input levels. The proposed DDSP improves the efficiency at a low input level and has high power efficiency.

Figure 10: Photographs of the six-coil loudspeaker.

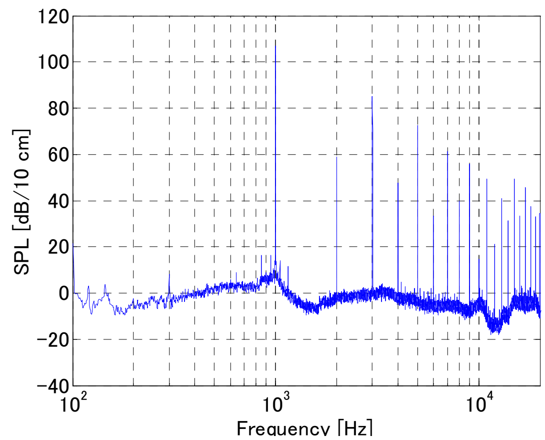

Figure 11: Output power spectrum of the DDSP in the absence of NSDEM.

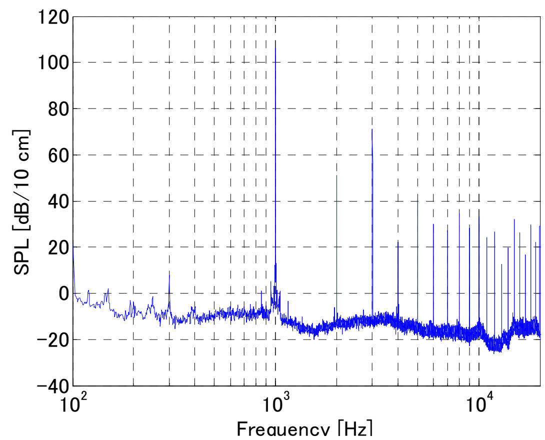

Figure 12: Output power spectrum of the proposed DDSP.

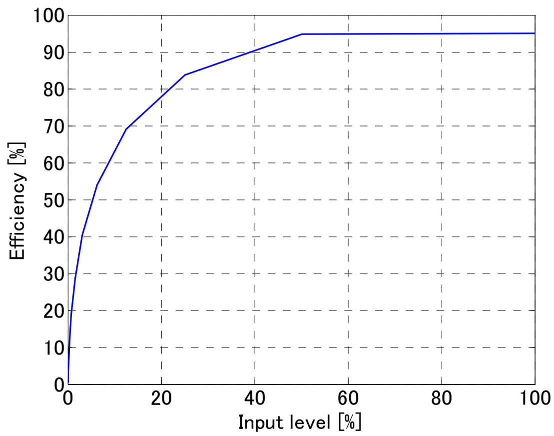

Figure 13: Power efficiency against the input level.

5. CONCLUSION

A novel DDSP involving the use of a newly devised mismatch shaper method, ML-NSDEM, is proposed. The proposed system has several merits:

- The power efficiency can be improved when the output power level is lower than the full-scale level.
- Without increasing the number of speakers, the total number of levels of the DSM can be increased.

The output SPL of the speaker with six voice coils is 94 dB/m when a 3.3-V USB power supply is used exclusively.

6. ACKNOWLEDGEMENTS

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7. REFERENCES

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