Ultra-Low EMI and THD+N Multi-Level Class-D Audio Amplifier

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***Abstract* - This paper is about a multi-level PWM class-D audio amplifier. The recent trend in audio amplifiers is increasing output power for louder sound and more channels for better sound effects. This typically results in EMI problems and lower efficiency. Instead of a simple two-level buck power topology, a multi-level topology was proposed to solve both the EMI and low efficiency problems. However, the linearity of the amplifier is reduced by adopting a multi-level topology rather than a two-level topology. Level shifted-PWM was utilized in a prior Maxim integrated structure to overcome the low linearity problem. In this paper, a very effective and simple Folded-PWM architecture is proposed, and the new modulation scheme shows better linearity, THD+N, and performance than LSPWM. The FPWM is expected to be used commercially, since it has super performance.**

***Keywords*—Audio amplifier, Class-D, EMI, Multi-level**

I. INTRODUCTION

Audio systems are very familiar and have been widely used in applications including mobile devices, TV speakers and public announcement systems. Sound is an important component of the five senses of the human body and can transfer information, even through a blocked room or wall. Furthermore, the information in sound can be easily adapted, and there are many devices, such as radios, to transfer information without using a display or visual information.

The process of audio medium development has evolved from a first structure based on a cylinder with beeswax to high density recovery media, and improved audio systems have been developed with advances in electrical engineering.

Audio systems are frequently used for home, vehicle and theatre performance. A conventional circuit for the aforementioned audio systems consists of discrete components on a printed circuit board. The conventional circuit is classified as one of three types, depending on the audio output topology. The first, Class-A, a linear amplifier, is one of the longest amplifiers and shows good performance with respect to linearity and noise. However, its efficiency is very low, so a large heat sink is required which significantly increases volume and weight. The second, Class-AB has been widely adopted because it has no electromagnetic interference (EMI), or complex design circuit, and a variety of analysis and solving methods have already been presented. The third, Class-D, is a switching amplifier based on a switching mode power supply. This structure shows higher efficiency than Class-A, and as a result it has mainly been used for high power car audio and stage performances because heat generation is low. However, the circuit design is complex, because the noise generated during the switching operation produces distortion at the output as well as electromagnetic interference waves. Also, as mobile devices have improved, they require audio systems with low weight, small size and high efficiency.

Fig. 1. Development of mobile devices from simple communication device to complex multi-media

In mobile devices, a switching amplifier, Class-D, is used to drive the speaker. The speaker needs high power for sufficient sound, and efficiency is very important because it is powered by battery. Additionally, EMI has to be managed because other circuits in the chips of mobile devices, such as the RF circuit for communication, are sensitive to noise.

Class-AB linear amplifiers are still used in headphones or earphones, because these are connected with long cables of about 1m which limits noise. Furthermore, the power used in such devices is very low, about 1mW, therefore, low efficiency is not a big problem.

Audio systems have become more important in mobile devices because they have more functions, and have evolved with large displays into multi-media devices, as shown in Fig. 1.

Mobile devices require an audio system that is capable of transferring more power to the speaker, and integrating more channels into the system to drive more speakers. To transfer more power to a speaker, the speaker is driven by high

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Manuscript Received Jan. 07, 2022, Revised Mar. 20, 2022, Accepted Mar. 25, 2022

voltage as shown in Fig. 2, however, there is a problem. First, the EMI problem becomes worse and affects the performance of the system, because the high frequency signal generated by high voltage is much larger, which is transferred to the load of the speaker. Second, efficiency is degraded because a boost DC-DC converter has to be used for voltage higher than battery voltage.

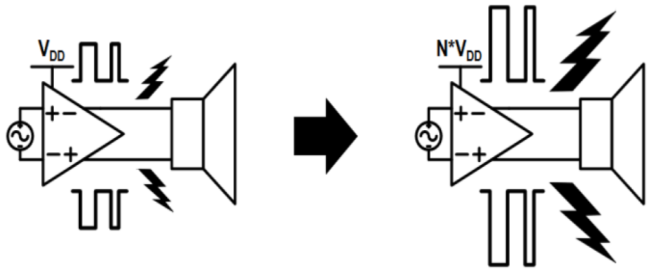


Fig. 2. Problem with transferring more power to a speaker

Another problem is generated with multi-channels. To acquire the effect of a sound field with multi-channels, the speakers have to be as far away from each other as possible, therefore, the distance from chip to speakers must be very far as shown in Fig. 3.

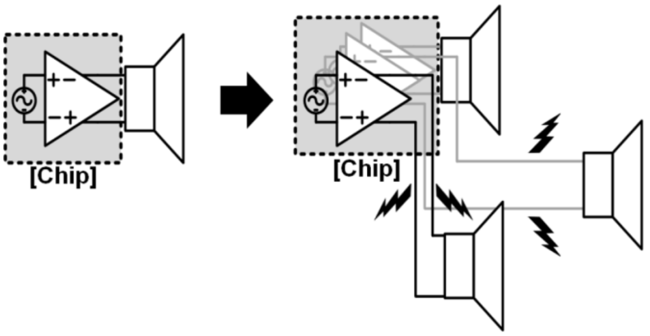


Fig. 3. Problem to use multi-channel

As a result, the EMI problem occurs because area is extended by the long routing of the PCB where the high frequency signal passes.

As noted, audio systems have become more important in mobile devices and require high performance and efficiency. To solve the EMI problems with increasing power, a basic method is presented, as shown in Fig. 4.

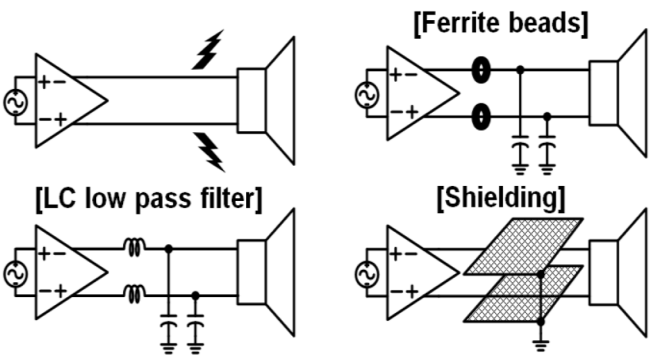


Fig. 4. Conventional solution for EMI

To reduce EMI noise, an additional low pass filter or shield method can be used, however, off-chip components increase cost and occupy larger area and volume on the PCB. [1] used a filter-less Class-D structure with boosted voltage. This structure used a boost converter for high power and a UPWM which adopted a spread spectrum scheme for control to reduce EMI noise. However, there was no method to improve efficiency.

[2] adopted a DPWM for control and a circuit to prevent physical damage was added. Furthermore, to increase efficiency a load adaptive control was used in [2].

However, there was no solution to reduce EMI. The recent research trend in audio systems has concentrated on switching amplifiers rather than linear amplifiers, however no total solution for EMI noise and efficiency has been reported yet.

In this paper we propose a method that deals with both the power efficiency and EMI problems. In the following sections we introduce a folded-PWM (FPWM) scheme for a multi-level Class-D amplifier. The FPWM scheme maintains high linearity at multi-level boundaries by preventing the abrupt DC shifting of error signals, while employing a single carrier triangular wave. The multi-level FPWM amplifier achieves 0.0023% THD+N, transmits 10W output power at THD+N=1% and attains 91% peak efficiency on an 8ohm load. The chip was fabricated in a 0.18um BCD process, and it occupies 6.45mm^2.

II. BASIC SWITCHING AUDIO AMPLIFIER

A. Difference with a linear audio amplifier

The purpose of linear amplifiers and switching amplifiers is to transfer an audio signal to speaker exactly. However, the output topology used to drive the speaker in linear and switching amplifiers are separated, which results in differences. Fig. 5 shows a Class-AB structure with a basic push-pull operation.

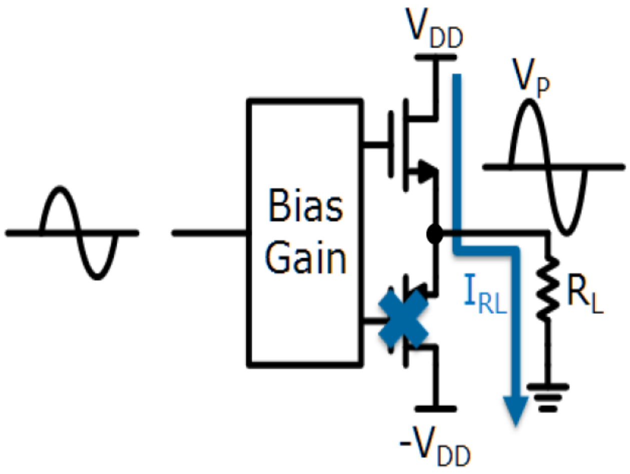


Fig. 5. Output stage of a linear amplifier: for increasing signal.

In the structure, biases have to be well defined because the simultaneous on-operation of two transistors is not allowed.

Generally, there is a distinct difference is efficiency between a linear and switching amplifier.

$$P_{av} = \frac{1}{T}\int_0^{T/2} V_{DS} I_D dt = \frac{1}{T}\int_0^{T/2} (V_{DD} - V_P sin\omega t)\left(\frac{V_P}{R_L} sin\omega t\right) dt = \frac{V_P}{R_L}\left(\frac{V_{DD}}{\pi} - \frac{V_P}{4}\right) \quad (1)$$

$$\eta = \frac{\frac{V_P^2}{2R_L}}{\frac{V_P^2}{2R_L} + \frac{2V_P}{R_L}\left(\frac{V_{DD}}{\pi} - \frac{V_P}{2}\right)} = \frac{\pi}{4}\frac{V_P}{V_{DD}} \quad (2)$$

Equations (1) and (2) mean that the maximum efficiency in a linear amplifier is generated at VP to be equal to VDD. However, this is rare situation, therefore, efficiency is normally very low. On the other hand, power from the source is transfer to load with 100% in a switching amplifier, as shown in Fig. 6.

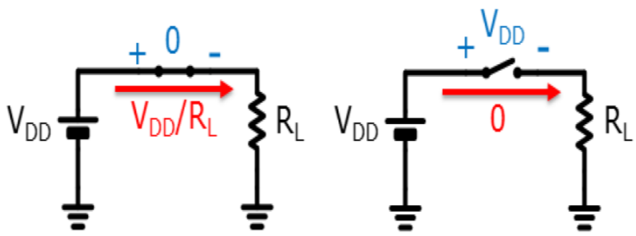


Fig. 6. Ideal efficiency of a switching amplifier.

Therefore, a switching amplifier is employed in mobile devices because it has higher efficiency than a linear amplifier. However, the switching amplifier needs a modulator to generate output, as shown in Fig. 7. A low pass filter (LPF) eliminates high frequency signals in sound.

In a Class-D system, the output of the analog amplifier is converted by modulator, and the operation is similar to an open-loop, from the modulator to the output switch, which means that this does not affect an analog amplifier.

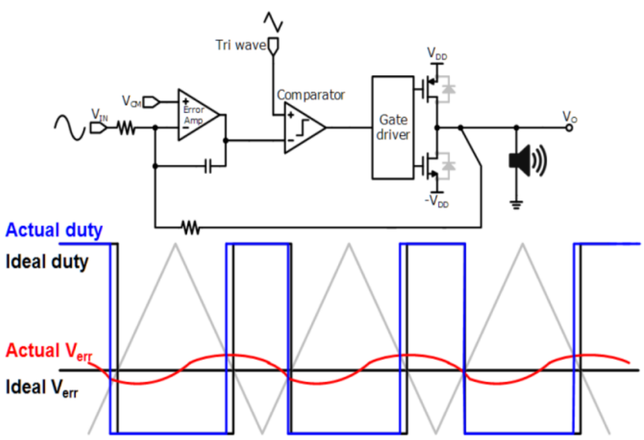


Fig. 7. Block diagram of a Class-D audio amplifier and duty signal

B. Difference with buck DC-DC converter

The structure of a Class-D switching audio amplifier based on electronics is similar to a buck DC-DC converter. Fig. 8 shows a Class-D amplifier and a buck DC-DC converter. The major difference between the buck and Class-D is the reference voltage. The reference voltage of the Class-D is an AC signal, while for the buck converter the reference voltage is DC voltage. This is because the purpose of a buck converter is to generate DC output, while the Class-D is generating AC output with exact gain.

Fig. 9 shows a typical audio amplifier and buck DC-DC converter. In the audio system, the speaker has varied impedance characteristics with frequency, however, the amplifier has an enough wide bandwidth to cope with variation in speaker impedance. Furthermore, the frequency range is negligible outside the audio range from 20Hz to 200kHz. A full-bridge is used in the Class-D and a half-bridge is used in a buck converter because Class-D transfers AC, while the buck transfers DC.

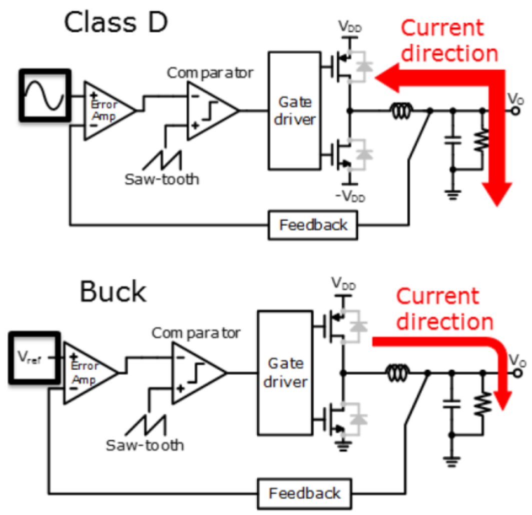
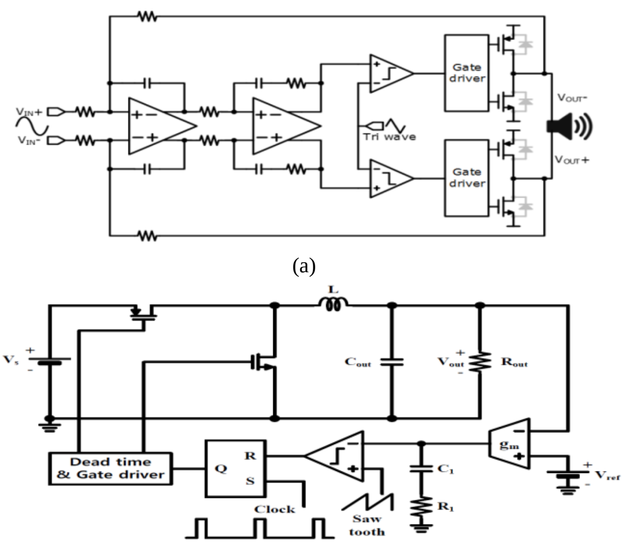


Fig. 8. Comparison between Class-D and buck converter.



(b)

Fig. 9. Typical audio amplifier and buck DC-DC converter; (a) Audio amplifier (b) Buck DC-DC converter

III. PROPOSED MULTI-LEVEL AUDIO AMPLIFIER

Fig. 10 shows the concept for the proposed audio system which uses an integration capacitor. In previous works, when multi-level is used, DC shifting is generated by using only one sawtooth wave. To solve the problem, LSPWM has been suggested, however, it also generates another problem. Therefore, using only one sawtooth wave without the DC shifting problem is the best approach. Here, an integration capacitor flip and signal folding are proposed. The integration capacitor is a feedback capacitor which has information about VIN. Therefore, an opposite voltage is generated at the output from the point of view of the virtual ground of the error amplifier when the capacitor is connected in the opposite direction. At this point, the information in the capacitor has to be maintained, and a non-overlap clock is needed, as shown in Fig. 11.

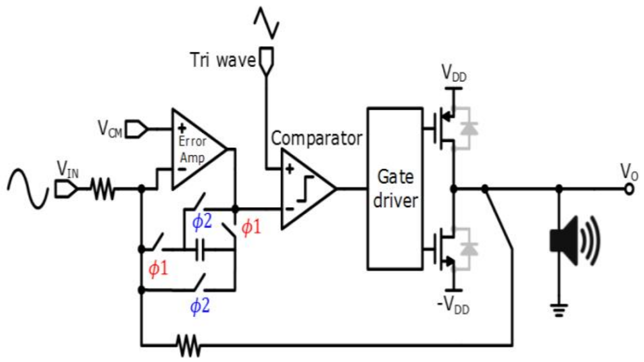


Fig. 10. Proposed audio system with capacitor switching.

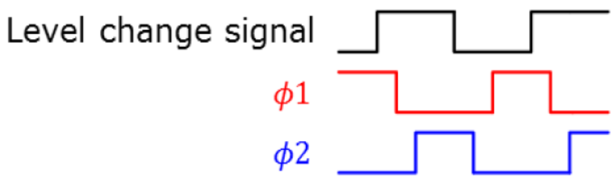


Fig. 11. Non-overlap clock for capacitor flip

Furthermore, by using signal folding, the signal range can be widened as shown in Fig. 12. The amplifier shown in Fig. 12 is the proposed amplifier.

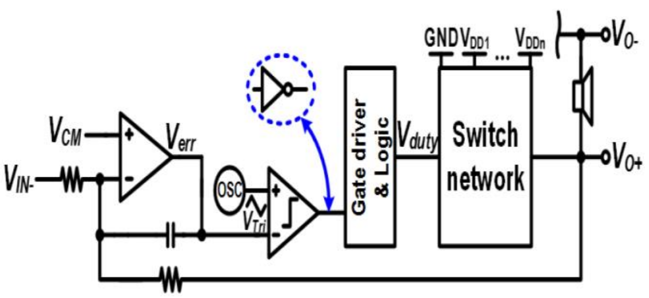


Fig. 12. Multi-level PWM Class-D audio amplifier for signal folding

The problem with a conventional single carrier PWM working with a multi-level power stage is that the duty has to change from near 100% to near 0% and vice versa near level boundaries. This is equivalent to an abrupt DC shifting of the integrator output Verr, which is not possible because an actual circuit needs some converging period. This causes errors and performance degradation.

We propose a new FPWM architecture which is illustrated in Fig. 13, to solve the error signal DC shifting problem. NOT gates and multiplexers are employed at comparator outputs for immediate duty shift while the Verr remains at rest. When a level state change occurs, multiplexers swap working signal paths in between the inverting path and non-inverting direct path so that Vduty changes immediately from ′0′ to ′1′ and vice versa. By doing so, the Verr continues a smooth transition unlike the abrupt change of the conventional PWM. As a result, this new method does not cause severe distortions at transition. Since the fundamental principle of PWM is converting the voltage potential information into timing information, when a specific voltage potential is transformed into a time axis to make a specific duty ratio through the comparator, the duty ratio does not lose its own information even if digital logic blocks are attached to the signal path.

Moreover, the only difference between two changeable signal paths is just one NOT gate. The NOT gate has a negligible effect on the timing information, but unfortunately it is equal to ′-1′ as an analog loop component. Accordingly, the path exchanger is inserted and modifies the loop to maintain negative feedback.

Finally, the level selection block monitors both edges of the triangular wave threshold. When it detects reaching Verr, it provides synchronization in the appropriate order to switch networks, the path exchanger and multiplexers to avoid output errors and keep negative feedback. So, the Verr becomes continuous without any DC shifting, even if level change occurs. This is identical to signal folding at certain boundaries, and this is why the modulation scheme is called a folded PWM (FPWM).

As a result, the FPWM successfully reduces the number of carrier waves to one and enables audio applications to adopt a multi-level topology with little performance loss.

IV. EXPERIMENTS

The proposed chip was fabricated with a 0.18um BDC process, and occupied a 1.5mm×2.35mm area as shown in Fig. 13.

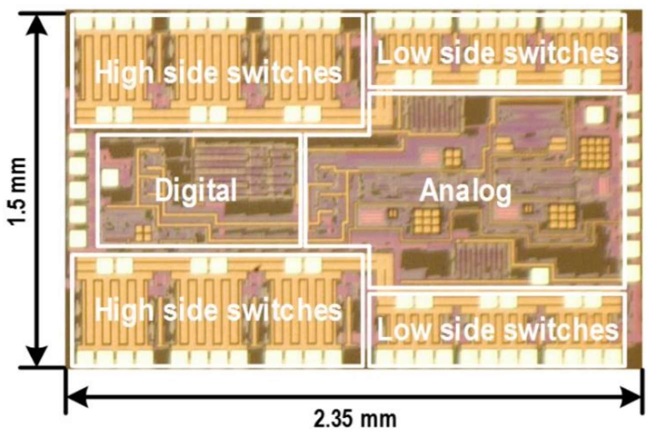


Fig. 13. Chip micrograph

The measurement was performed using Audio Precision's devices as shown in Fig. 14. The Audio Precision APX-555 analyzer and AUX-0025 LPF were utilized.

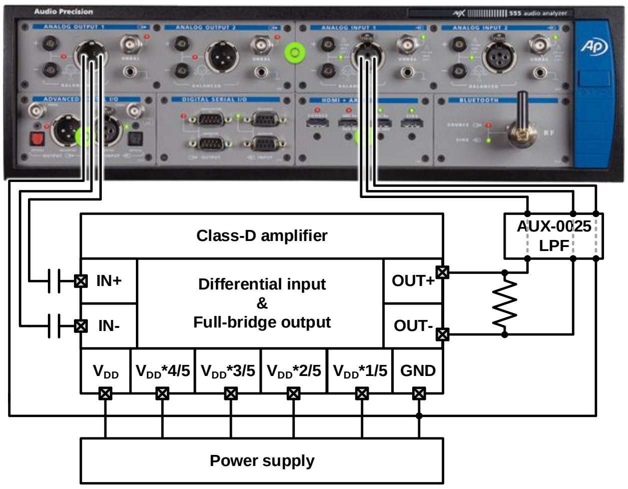


Fig. 14. Measurements setup.

Fig. 15 shows the output waveform when the load resistance was 4Ω. 2-level, 4-level and 6-level are operating well, respectively. Furthermore, duty is well generated with level. Fig. 16 shows the SNR from 20Hz to 20kHz which is 99dB. And Figs. 17 and 18 show THD+N with various conditions, which is under 0.04% for all conditions. Fig. 19 shows the PSRR which results in 86dB with 217Hz 100mVrms. The low frequency signal used in the measurement was generated by an AC generator.

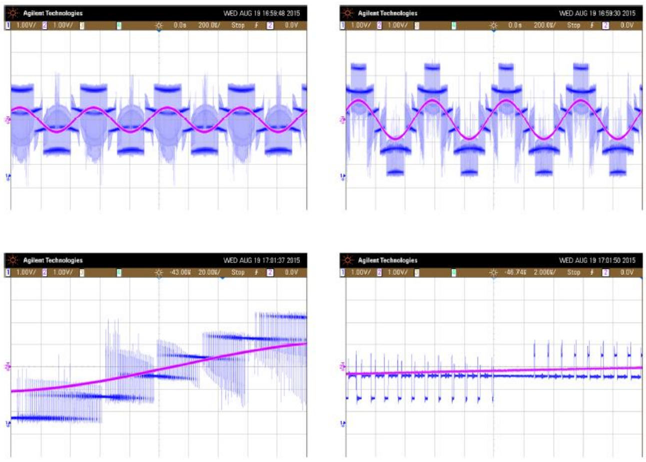


Fig. 15. Measured output waveform.

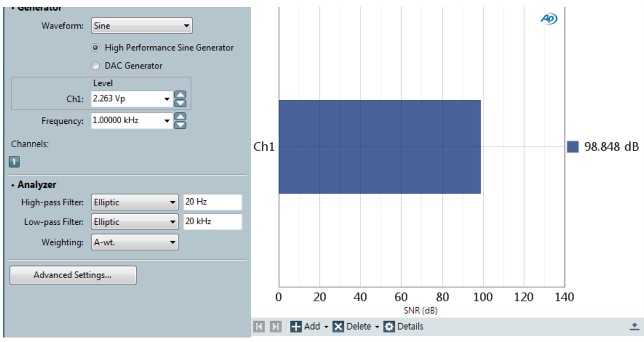


Fig. 16. Measured SNR

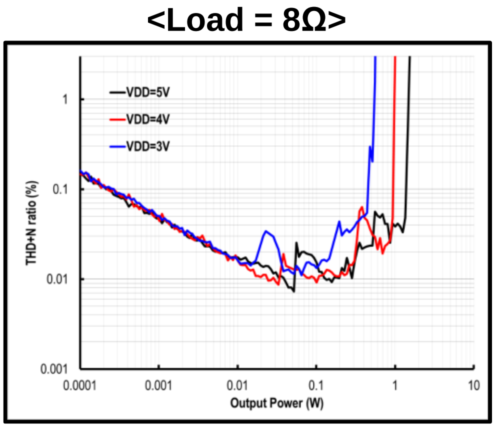


Fig. 17. THD+N with variety voltages

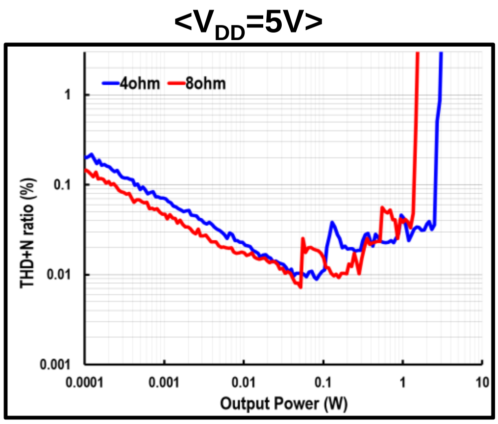


Fig. 18. THD+N with variety load conditions.

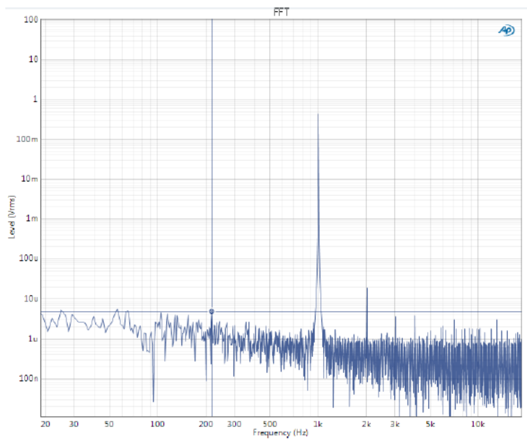


Fig. 19. PSRR

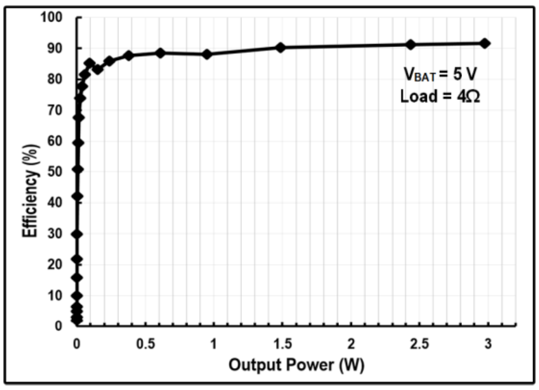


Fig. 20. Measured efficiency

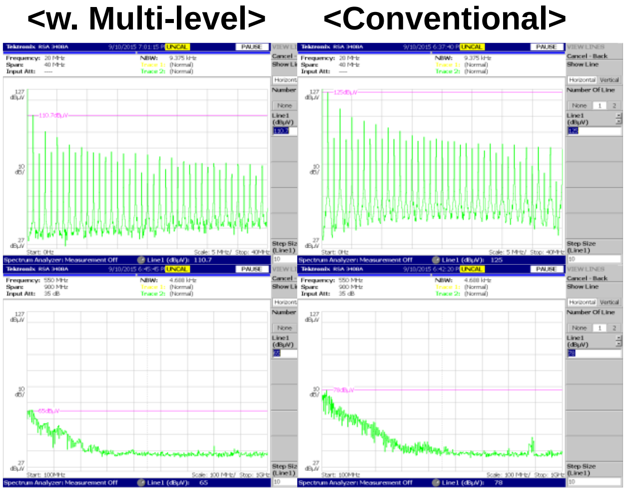


Fig. 21. EMI measurements.

Fig. 20 shows the efficiency graph. Peak efficiency is 91.6% with a maximum power of 3W. Fig. 21 shows the EMI measurements results. A 0~40MHz area -14dB decrease and a 100MHz~1GHz area -13dB decrease were measured.

IV. CONCLUSION

TABLE I shows the proposed multi-level Class-D audio amplifier and previous works. It verifies the FPWM architecture successfully exploiting a multi-level for the audio application.

REFERENCES

[1] A. Nagari, E. Allier, F. Amiard, V. Binet, and C. Fraisse, "An 8Ω 2.5W 1%-THD 104 dB(A)-Dynamic-Range Class-D Audio Amplifier with Ultra-Low EMI System and Current Sensing for Speaker Protection," *IEEE J. Solid-State Circuits*, vol. 47, pp. 3068-3080, 2012.

[2] M. Berkhout, L. Dooper, and B. Krabbenborg, "A 4Ω 2.65W Class-D Audio Amplifier with Embedded DC-DC Boost Converter, Current Sensing ADC and DSP for Adaptive Speaker Protection," *IEEE J. Solid-State Circuits*, vol. 48, pp. 2952-2961, 2013.

[3] Maxim Integrated. (2015, Mar.). “MAX98307/MAX98308 3.3W Mono Class DG Multilevel Audio Amplifier” [Online]. Available: http://datasheets.maximintegrated.com/

[4] M. Xin, C. Zao, Z. Ze-kun, and Z. Bo, "An Advanced Spread Spectrum Architecture Using Pseudorandom Modulation to Improve EMI in Class D Amplifier," *IEEE Trans. Power Electronics*, vol. 26, pp. 638-646, 2011.

[5] F. Guanziroli, R. Bassoli, C. Crippa, D. Devecchi, and G. Nicollini, "A 1 W 104 dB SNR Filter-Less Fully-Digital Open-Loop Class D Audio Amplifier with EMI Reduction," *IEEE J. Solid-State Circuits*, vol. 47, pp. 686-698, 2012.

TABLE I. Performance summary and comparison

	This Work	MAX98308 [3]	Nagari [1]	Berkhout [2]	Xin [4]	Guanziroli [5]
Topology	**6-level**	**3-level**	**2-level**	**2-level**	**2-level**	**2-level**
THD+N @ before start clipping (%)	**0.03**	0.05	0.025	0.015	0.02	0.2
SNR, A-weighted (dB)	**99**	-	104	100	93	104
PSRR @ 217Hz (dB)	**86**	78	93	-	-	-
IQ (mA)	**1**	1.85	2	1.55	-	3.3
Supply (V)	**3 to 5**	2.6 to 5.25	2.5 to 4.8	-	2.4 to 5.5	3.6 to 5
η (%)	**91.6**	84	91	81	90	81
Fs (kHz)	**1000**	340	446	-	300	384
Max Pout @ 1kHz THD+N<1% (W)	**3 (4Ω load)**	2.85 (8Ω load)	2.5 (8Ω load)	2.3 (4Ω load)	1.2 (8Ω load)	1 (8Ω load)
Architecture	**FPWM**	LSPWM	UPWM	DPWM	PWM	UPWM
Process	**0.18 um**	-	0.13 um	-	0.5 um	0.13 um
Area(mm2)	**3.125**	-	2.19	-	2.25	0.94

Ji Hun Lee received the B.S., M.S., and Ph.D. degrees in electrical engineering from Korea Advanced Institute of Science and Technology (KAIST), Daejeon, South Korea, in 2014, 2016, and 2021, respectively. Since 2021, he has been with System LSI Business, Samsung Electronics Company Ltd., Hwaseong, South Korea, where he is currently involved in the design of audio amplifier chips. His research interests are in the field of circuits and systems for power converters and audio power amplifiers.

Gyu Hyeong Cho received the B.S. degree from Hanyang University, Seoul, South Korea, and the M.S. and Ph.D. degrees from the Korea Advanced Institute of Science and Technology (KAIST), Daejeon, South Korea, in 1975, 1977, and 1981, respectively, all in electrical engineering.

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Dr. Cho received the Outstanding Teaching Award from KAIST. He served as a member of the ISSCC International Technical Program Committee, and is currently an Associate Editor for the IEEE JOURNAL OF SOLID-STATE CIRCUITS. At the ISSCC 60th Anniversary in 2013, he was awarded the ISSCC Author-Recognition Award as one of the top 16 contributors of the conference during last 60 years in ISSCC.