A New High-Efficiency and Super-Fidelity Analog Audio Amplifier with the aid of Digital Switching Amplifier: Class K_* Amplifier

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***ABSTRACT* - This paper presents an audio amplifier that simultaneously has the advantages of digital and analog amplifier. The high efficiency is achieved by the digital amplifier, which is playing a role of dependent current source. The high fidelity is guaranteed by the analog amplifier playing a role of independent voltage source.**

Experimental results show that the proposed amplifier has the 0.005% total harmonic distortion (THD) and around 90% power efficiency at 50W output.

I. INTRODUCTION

There have been studied many amplifiers for the audio application. Analog amplifiers such as class A, B and AB are based on the liner circuit technology. Although the linear amplifiers have excellent distortion characteristics, they unfortunately have poor efficiency, thus, they are bulky in volume for cooling.

To date with the advent of the Green Round, most of linear DC power supplies have already replaced by the switching circuit technology. New technology for efficiency improvement is also needed in the area of audio amplifier in order to save energy. Switching class D amplifier having pulse width modulation (PWM) control topology can be used to obtain high efficiency for high power audio amplifier. However, there exists a reason that switching topology is hardly used in analog amplification. The distortion introduced by switching amplifier is typically larger than that of their linear counterparts due to their intrinsic nonlinearity. PWM frequency around several hundreds kHz does not really allow high fidelity reproduction of audio bandwidth of 20Hz through 20kHz.

There are so many approaches for the high fidelity amplification. A simple method to think with ease is to increase the switching frequency. But, it is not a novel approach because the switching losses are increased with proportion to the switching frequency. Zero-Current-Switched Quasi-Resonant converters (ZCS-QRCs), in which an auxiliary LC network reduces the switching losses, allow pulsed operation in MHz range [1, 2]. As other methods, Sigma-Delta ($\Sigma\Delta$) technique was introduced in signal processing [3, 4, 5] and multiple loop feedback technique was suggested for the improvement of fidelity [6, 7].

Efforts have been made since early 60's to develop switching power amplifier [8]. In the past decade, the topology of switching converter for power application has been given more attention than the control scheme of switching amplifier [9, 10, 11]. Recently, a few control schemes were introduced in audio applications of nonlinear PWM scheme [12]. As far as authors know, no literatures have addressed the audio amplifier that simultaneously has both of high efficiency and high fidelity except the paper of [13].

A new novel concept of audio amplifier is proposed in this paper. The proposed amplifier [14], named by us *class* K_* amplifier, is composed of two blocks, that is, analog amplifier block and digital amplifier block: the analog amplifier plays a role of independent voltage source and the digital amplifier plays a role of dependent current source controlled by signal current of analog amplifier. Basic concept is realized by combining only the advantages of both of analog and digital amplifier: the advantage of analog amplifier having excellent total harmonic distortion (THD) is mixed with that of digital amplifier that has the excellent power efficiency.

This paper presents the one possible implementation of high-efficiency and high-fidelity analog amplifier having digital amplifier. Section II gives the basic idea extraction of proposed our audio amplifier. The detailed design of proposed audio amplifier is presented in section III. The simulation and experimental results are reported in section IV. Finally, section V deals with the conclusions. Not only the THD comparable to the commercial high-fidelity audio amplifier is achieved but also the power efficiency surpassing the commercial audio amplifier is attained.

$*$ **The acronym *K* of *class K* amplifier means the initial character of KAIST.**

II. BASIC IDEA EXTRACTION

In the audio amplifier, the upper most important things are fidelity (or THD) and power efficiency. Although so many amplifiers such as class A, B, AB, and D so on have been developed for the audio application till now, there is no amplifier that simultaneously has the excellent fidelity and outstanding power efficiency. Table 1. shows the comparison of advantages and disadvantages for the amplifiers. Analog audio amplifiers such as class A, B and AB have high

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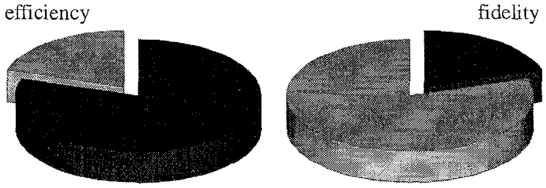
fidelity. However, they show considerably poor efficiency and bulky volume. On the other hand, the switching class D amplifier has the excellent efficiency and small volume, but it shows relatively bad fidelity characteristics. If we represent the major performances for the amplifiers with graph, the performance-index of amplifiers can be simply graphed as shown in Fig. 1.

Table 1. performance comparison of amplifier

amplifier	class	fidelity	efficiency	volume
analog	A	excellent	poor	x-large
	AB	good	fair	large
	B	fair	good	fair
digital	D	bad	excellent	small



a) analog amplifier (b) digital amplifier
Fig. 1 comparison of performance index

Authors have tried to find the way to take only the merits of the analog and digital amplifiers: the high fidelity of analog amplifier and the high efficiency of digital amplifier. Fig. 2 shows the basic circuit that can perform the above two merits. The basic circuit is composed of two sources: one is the independent voltage source V_a that stands for analog amplifier, and the other is the dependent current source i_d that represents the digital amplifier. In Fig. 2, R_{sense} and R_{load} show the current-sensing resistance and speaker impedance, respectively. The current i_a, i_d, and i_o mean the current supplied by the analog amplifier, the current supplied by the digital amplifier, and the load current of speaker, respectively.

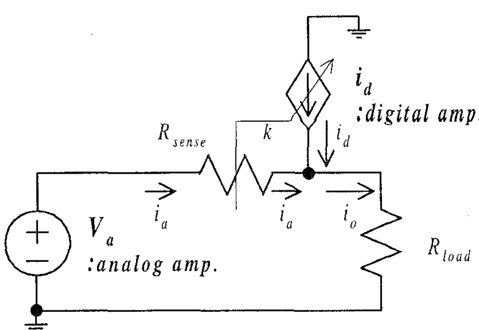


Fig. 2 conceptual basic circuit

The current equation at node N can be written as

$$i_o = i_a + i_d \,. \tag{1}$$

If we let i_d be proportional to i_a, the i_d can be written as

$$i_d = k\, i_a \tag{2}$$

,where k is the proportional coefficient, that is, the current gain. From (1) and (2), it follows that

$$i_o = (1 + k) i_a \,. \tag{3}$$

If we make k become sufficiently larger than 1, the current i_o is approximately the same as i_d and can be written as

$$i_o \cong k\, i_a = i_d \,. \tag{4}$$

This means that the sufficiently large k can make i_a sufficiently small. Since the current i_a is supplied by the analog amplifier that has poor efficiency, it is necessary to either minimize i_a or maximize i_d by increasing the k in order to get high power efficiency. This is because the digital amplifier has much better power efficiency than the analog amplifier. We have to make i_d sufficiently larger than i_a to achieve high efficiency.

Now, let us consider the more detailed configuration of the proposed scheme shown in Fig. 3. The voltage V_i and V_o are input and output voltage signals. A_a, g_d and A_R indicate the analog amplifier block, digital amplifier block and sensing unit, respectively. The feedback network has a transfer function f and feeds back a signal V_f to the input. The signal V_f is subtracted from the input signal V_i at the input differential node. Error signal V_e is the difference between V_i and V_f, and V_f is fed back to the analog amplifier. The i_a, i_d and i_o are the current supplied by A_a, the current supplied by g_d and load current of speaker, respectively. Note that the incoming and outgoing current are the same in the sensing unit. The sensing unit plays a role of generating the sensed voltage V_s proportional to i_a. In Fig. 3, for easy analyses, let us assume that the voltage drop across sensing unit A_R is very small and can be neglected. Then, we can write

$$V_o \cong V_o' \,. \tag{5}$$

From Fig. 3, we obtain

$$V_o = A_a V_e \tag{6}$$

$$V_f = f\, V_o \tag{7}$$

$$V_e = V_i - V_f \,. \tag{8}$$

From these equations, we can obtain

$$\frac{V_o}{V_i} = A = \frac{A_a}{1 + A_a f} \,. \tag{9}$$

In this case, A , so called the closed-loop gain, is the overall voltage gain in the above negative feedback circuit [15], where the gain A is determined only by the analog amplifier irrespective of digital amplifier. The gain A can be rewritten as

$$A = \frac{A_a}{1+T} = \frac{1}{f}\frac{T}{1+T}. \tag{10}$$

,where T is the loop gain given by $A_a f$. Thus, for large value of loop gain T, the overall amplifier voltage gain is purely determined by the feedback transfer function f irrespective of digital amplifier.

Let us now consider the current components supplied by both of the analog amplifier A_a and the digital amplifier g_d . From Fig. 3, we obtain

$$i_o = i_a + i_d \tag{11}$$

$$i_d = g_d V_s \tag{12}$$

$$V_s = A_R i_a . \tag{13}$$

From these equations, we obtain

$$i_o = (1 + g_d A_R) i_a . \tag{14}$$

Comparing between (3) and (14) gives

$$k = g_d A_R . \tag{20}$$

Since the digital amplifier g_d has much more excellent efficiency than the analog amplifier A_a, it is needed to decrease the current i_a supplied by analog amplifier and increase the current i_d supplied by the digital amplifier. From (14), it is apparent that i_a can theoretically be reduced by increasing either g_d or A_R. Since there exist several methods to effectively increase the g_d and A_R, a trade-off study is needed to select a proper method. The circuit configuration, sensing scheme, speed of devices, current supplying capability and slew rate of amplifier, gain of sensing unit and so on should be considered not separately, but as a whole.

III. DESIGN of PROPOSED AUDIO AMPLIFIER

Let us now carry out the details necessary to implement the real circuit. Fig. 4 is the detailed version of Fig. 3. Precisely speaking, the newly proposed audio amplifier is composed of three blocks: analog amplifier, digital amplifier and sensing unit. In order to clarify the operation of proposed scheme, it is necessary for us to consider the roles of each block and point out the advantages of the proposed circuit as follows.

A. Digital amplifier supplies almost all the current needed in the speaker.
B. Analog amplifier merely supplies the ripple current to compensate the distortion caused by digital amplifier.
C. Digital amplifier operates as a dependent current source.
D. Analog amplifier operates as an independent voltage source.
E. The parallel linkage between current source and voltage source at the output gives no problem.
F. High fidelity, i.e., low THD is guaranteed by the analog amplifier.
G. High efficiency is obtained by the digital amplifier.
H. Even if the analog amplifier is class B amplifier, since the current supplied by the class B amplifier is relatively small, the distortion caused by this amplifier is smaller than that of conventional single class B amplifier which is to supply all the current needed in the speaker.
I. There is absolutely no stability-problem in the proposed scheme because the digital amplifier is approximately first order system.
J. It is desirable that the bandwidth of analog amplifier can cover the maximally wide range in order to quickly absorb the ripple current caused by the inductor *L*.
K. It is expected that the fidelity and the power efficiency of the proposed scheme are as good as the fidelity of analog amplifier and the power efficiency of the digital amplifier, respectively.
L. Either large heat sink or noisy fan for cooling is no longer needed in the proposed scheme.

In Fig. 4, analog amplifier is composed of input amplifier, base driver, complementary output stage and feedback network.

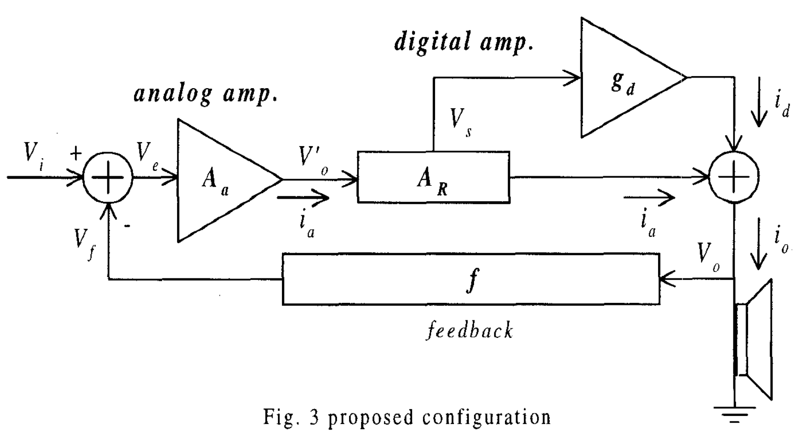


Fig. 3 proposed configuration

Digital amplifier is composed of comparator, gate driver, a pair of power switch of PMOS and NMOS, and inductor that gives a function of current source. Sensing unit can simply be implemented with operational amplifier.

Now, let us consider the operation of the proposed amplifier. The principle of operation is very simple and clear. There exist two switching states in digital amplifier. The first switching state is M1 (PMOS) turn-on and M2 (NMOS) turn-off state. The second switching state is vice versa. Firstly, assume that the input signal V_i of analog amplifier is sinusoidal waveform and, now, is positively growing. When V_i increases, the current i_a also increases with proportion to V_i. Increasing of i_a gives the voltage drop across R_{sense}. Voltage across R_{sense} is amplified to V_s by A_2 and V_s is fed to the inverting input of comparator C_m. When V_s reaches upper threshold voltage V_T^u, the output of C_m becomes $-V_{dd}$ and the noninverting input of C_m is changed into lower threshold voltage V_T^l. Thus, M1 turns on, but, M2 keeps off-state. This is the first switching state. The threshold voltage V_T^u and V_T^l are written as

$$V_T^u = \frac{R_2(+V_{dd})}{R_1 + R_2} \tag{16}$$

$$V_T^l = \frac{R_2(-V_{dd})}{R_1 + R_2}. \tag{17}$$

During the first switching state, the inductor current i_d is increased with the approximate slope of $(+V_{dd} - V_o(t))/L$, and i_a is decreased, where $V_o(t)$ represents the output voltage at time *t*. The current i_d keeps growing and flowing into the load with the same direction with i_o until i_a reaches up to zero. After i_a reaches to zero, the excess current of i_d starts to flow into the analog amplifier. This means that the excess current of i_d is naturally absorbed by the analog amplifier. The amount of excess current of i_d becomes negative i_a. The voltage drop across R_{sense} caused by the negative i_a makes the polarity of V_s negative. When V_s reaches V_T^l, the comparator output is changed to $+V_{dd}$. Thus, M1 is turned off and M2 is turned on. Now, the first switching state is ended and the second switching state is started. During the second switching state, the inductor current i_d is decreased with the slope of $(-V_{dd} - V_o(t))/L$, and i_a is increased to compensate the ripple current . Again, if V_s reaches V_T^u, then the second switching state is changed into the first switching state. The repetition between the above two switching states gives high-efficiency and high-fidelity amplification. In order to accommodate easy understanding of new scheme, simulated waveforms are presented in Fig. 5, which will be dealt in section IV.

Let us consider the current supplying capability of i_d related with the efficiency of the proposed audio system. As mentioned before, the current i_o needed in the speaker is the sum of i_a and i_d. In order to achieve high efficiency, almost all the current i_o should be supplied by i_d instead of i_a. The current supplying capability of i_d is mainly determined by the inductor *L* of the digital amplifier. The current i_d supplied by the digital amplifier during the turn-on time Δt is approximately $(\pm V_{dd} - V_o(t))\Delta t/L$. The current i_d can be adjusted by changing the inductance *L*, Δt or $\pm V_{dd}$. If *L* is increased, then i_d is decreased, and vice versa. The inductance *L* should be selected by considering the supply voltage, switching frequency (turn-on time), speaker current, the ripple absorbing capability of analog amplifier and the audible bandwidth, that is, 20Hz~20kHz. If *L* is too large, problem happens at high frequency since the amount of current i_d built-up during the switch-turn-on time Δt can be too small to supply the almost all the current needed in the speaker. In this case, since almost all the current is inevitably supplied by the current i_a of the analog amplifier, the overall power efficiency becomes bad. The power efficiency of this case is probably close to that of conventional analog amplifier at high frequency side.

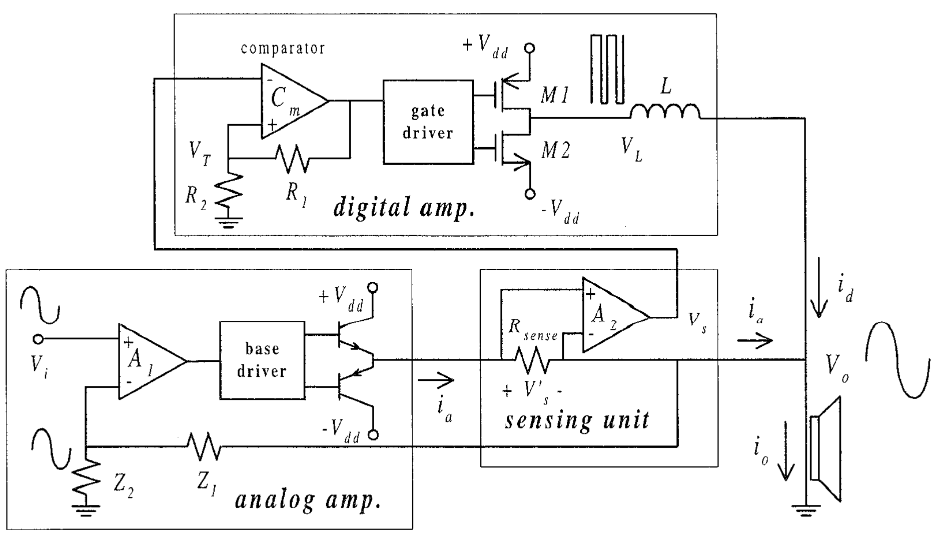


Fig. 4 detailed proposed circuit, i.e., class K amplifier

On the other hand, if L is too small, then the current i_d builds up so fast and the ripple current to be treated by the analog amplifier becomes considerably large at low frequency. As a result, the large ripple current causes some wasteful power consumption.

Next, let us consider the switching frequency. The digital amplifier and the sensing unit form a negative feedback loop. Since this loop is a simple first order system, it is absolutely stable. The switching frequency f_s given by (18) is a function of inductance L, threshold hysterisis voltage V_T of comparator, gain A_2 including R_{sense}, slew rate SR of comparator, gate driver and MOS power switches.

$$f_s \propto \frac{A_2\, SR_{cm}\, SR_{gate}\, SR_{MOS}}{L V_T} \tag{18}$$

Although the switching frequency f_s can be tuned-up with the factors L, V_T, A_2 and SR, it is desirable to choose V_T or A_2 instead of L and SR because of several reasons. The first reason is because the adjustment of V_T or A_2 is more convenient than that of L and SR. The second is because L is severely concerned with the current supplying capability of the digital amplifier. The current slope is limited by V_{dd}/L. Another reason is because all the slew rates except SR_{gate} are naturally determined by the intrinsic slewing performance of the devices. In reality, the slew rate of gate driver is artificially adjusted to protect the shoot-through between the upper and lower switches.

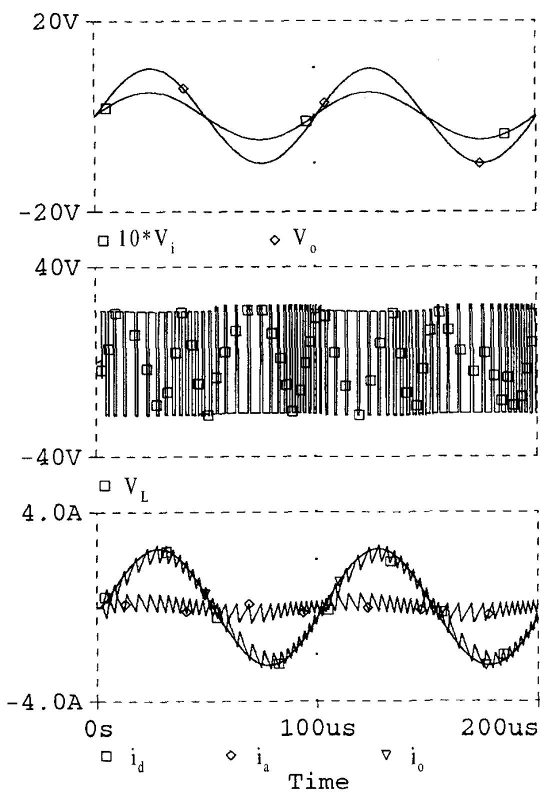


Fig. 5 simulation results at 10kHz

Since the feedback topology of this proposed system is categorized into voltage series feedback, the closed loop voltage gain A can easily be calculated from Z_1 and Z_2 as follows:

$$A = 1 + Z_1/Z_2\,. \tag{19}$$

IV. SIMULATION and EXPERIMENTAL RESULTS

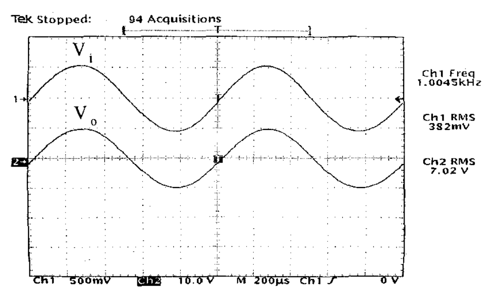
The computer simulation with PSpice and experiments are performed to confirm the operation of the proposed audio amplifier. The supply voltage $\pm V_{dd}$ is set to $\pm$22Vdc and 4 Ω load is used for a typical 50W output. The closed loop gain is adjusted to about 20. The switching frequency in idle condition, in which no signal is applied to the input, is tuned-up to around 300kHz.

A. SIMULATION

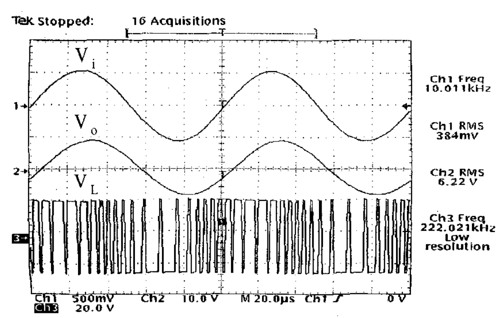
The input signal of $0.5\sin(2\pi\ 10k\ t)$ is applied to the input of analog amplifier. Fig. 5 shows the waveforms: input voltage V_i, output voltage V_o, the current i_a supplied by the analog amplifier, the current i_d supplied by the digital amplifier and the output current i_o, respectively.

B. EXPERIMENT

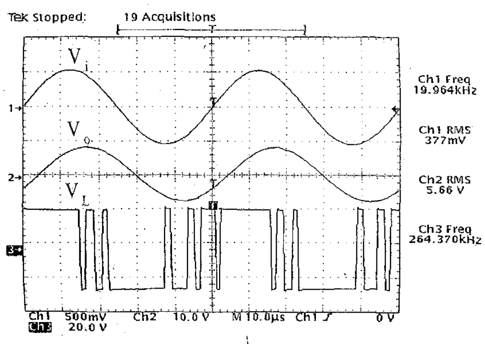
The input and the output oscillograms are shown in Fig. 6 at 1kHz, 10kHz and 20kHz for V_i, V_o and V_L, respectively. From the comparison between Fig. 5 and Fig. 6, we can see that the waveforms of experiments are all the same with the simulations.



a) f=1kHz



b) f=10kHz



c) f=20kHz

Fig. 6 experimental results

C. TOTAL HARMONIC DISTORTION

THDs vs. output powers, and THDs vs. frequencies are measured. Fig. 7 shows the THD vs. output powers at 1 kHz. The shape of the curve looks like an 'U' character. The THD at low-level output is worse than that of at mid level output because the noise severely affects the signal at low-level output. Also, the THD at the end of high level output is poor. If the output signal is too large to cover with the supply voltage $\pm V_{dd}$, the clipping problem of output signal abruptly arises due to $\pm V_{dd}$ limitation. Note that the THDs at mid level power 0.5W through 50W are about 0.01%. Fig. 8 shows the THDs vs. frequencies at 1W, 10W and 50W, respectively.

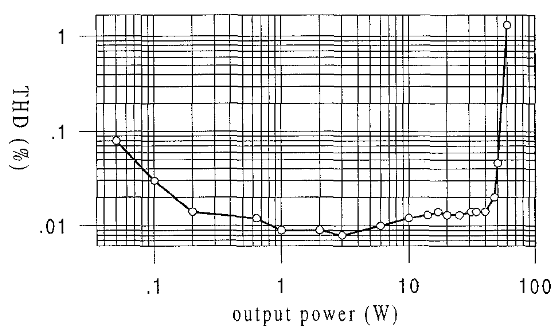


Fig. 7 THD vs. output power at 1kHz

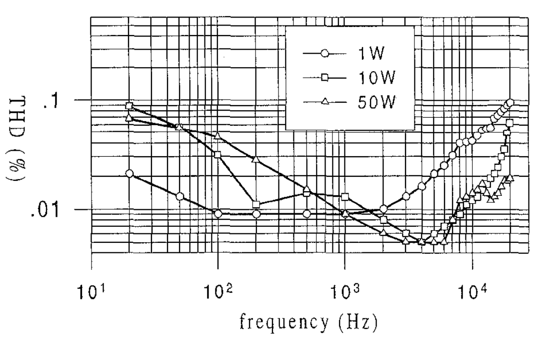


Fig. 8 THDs vs. frequencies

Experimental results show that the proposed audio amplifier has excellent THD performance comparable to commercial high fidelity audio amplifier.

D. POWER EFFICIENCY

Depending on the traditional definition, the total power efficiency [16] is given by

$$\eta(x) = \frac{P_o(x)}{P_{dd}(x)} \tag{20}$$

,where $P_o(x)$, $P_{dd}(x)$ and x are the output power of speaker, total power consumption and arbitrary output level, respectively. Fig. 9 shows total power consumption vs. output power of speaker at 1kHz. Fig. 10 shows total power efficiency vs. output power of speaker at 1kHz. From Fig. 9 and 10, we can find that the newly proposed audio amplifier has the excellent power efficiency of around 90% at 50W, and becomes better for the output power to go higher.

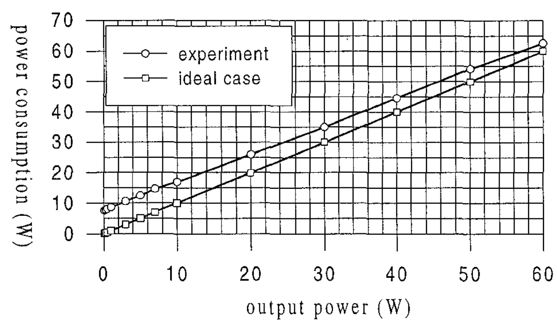


Fig. 9 total power consumption

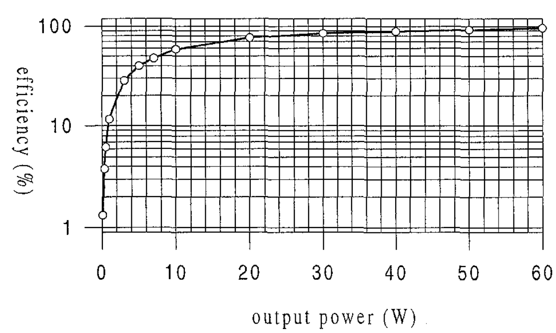
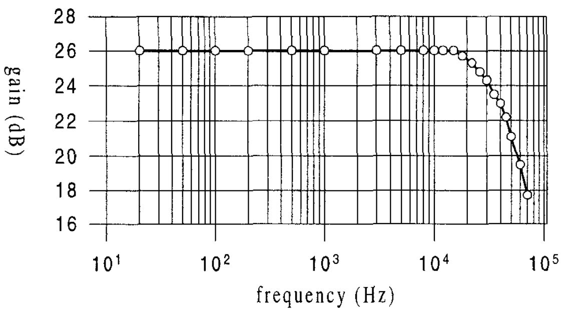


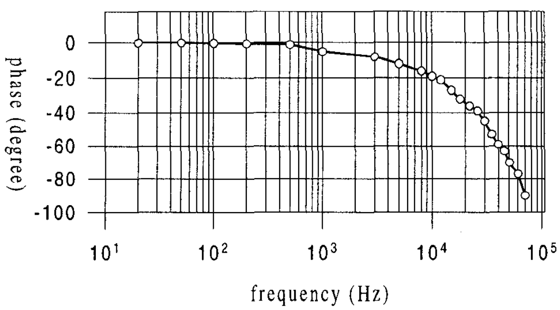
Fig. 10 total power efficiency

E. FREQUENCY RESPONSE

As mentioned previously, since the proposed system is approximately first-order system, it is always stable. The frequency response of the closed loop system is measured. Fig. 11 shows the frequency response: magnitude response and phase response. Note that the magnitude response is maximally flat in the audible frequency of 20Hz～20kHz and the 3dB bandwidth is located at around 40kHz.



a) magnitude response



b) phase response

Fig. 11 Frequency response

V. CONCLUSIONS

Almost all the audio amplifiers are based on the linear circuit technology. They show excellent fidelity, but poor efficiency. Recently, the digital switching amplifier such as class D type has been studied for commercial applications. Although class D amplifier has excellent power efficiency, it shows relatively inferior fidelity when compared with the analog amplifier. This paper has addressed the newly proposed *class k* audio amplifier that has only the advantages of analog amplifier and digital amplifier, that is, the high fidelity of analog amplifier and the excellent efficiency of digital amplifier are simultaneously adopted in a system.

The proposed audio amplifier is simulated and clearly implemented. A 50W prototype has demonstrated that newly proposed audio amplifier has the excellent THD of around 0.005% and the efficiency as high as about 90%. Moreover, no stability problem exists.

Authors believe that the newly proposed technology would be good for reducing size and cost as well as saving energy.

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