*Int. J. Engng Ed.* Vol. 24, No. 3, pp. 625–633, 2008
Printed in Great Britain.

0949-149X/91 $3.00+0.00


# Instructional Material for Laboratory Sessions Aimed at Designing a Robust Low Distortion Audio Amplifier*

WILMAR HERNANDEZ
*Department of Circuits and Systems in the EUIT de Telecomunicacion at the Universidad Politecnica de Madrid, Campus Sur UPM, Ctra. Valencia km 7, 28031 Madrid, Spain. E-mail: whernan@ics.upm.es*

*Instructional materials for Analog Electronics and Control Systems laboratory sessions aimed at designing a robust low distortion audio amplifier are presented. The paper is aimed at both senior-level undergraduate and first-year graduate students of Electrical Engineering, and its main objectives are to present key concepts and information to assist students to investigate the distortion performance of audio amplifiers, and to design a real audio amplifier based on an $H_\infty$ robust controller. Here, the analysis and design of the audio amplifier is carried out using PSpice simulations, which is a very suitable method for studying distortion mechanisms.*

**Keywords:** audio amplifier; total harmonic distortion; intermodulation distortion; negative feedback compensation; $H_\infty$ robust control

## INTRODUCTION

IN MOST ELECTRICAL ENGINEERING CURRICULA, Analog Electronics and Control Systems are two of the most important undergraduate subjects. At the Universidad Politecnica de Madrid, Analog Electronics and Control Systems are two nine-credit subjects, with four hours of lecture and two hours of laboratory scheduled per week for each subject. The Analog Electronics course, one of the prerequisite courses that students have to take before taking Control Systems, is taught at the second-year undergraduate level and Control Systems is taught at the senior undergraduate level.

However, in spite of the fact that these two subjects are closely related, students learn about distortion mechanisms in amplifiers and the benefits of the negative feedback (NFB) on the reduction of the distortion of the amplifiers, they do not develop skills in the application of the control techniques taught in Control Systems in order to design optimum NFB compensators for audio amplifiers. Even worse is that students taking Control Systems courses are given many examples and homework problems that have some relationship to real-world applications of the use of control methods in various aspects of system analysis and design but none of these examples and homework assignments constitute a 'case study' aimed at improving the distortion performance of audio amplifiers by using robust and optimal control techniques. As a result, there is a gap between Analog Electronics and Control Systems courses that needs to be bridged.

In addition, engineering design teams in industry do not usually design audio amplifiers using advanced or post-modern control techniques [1–4]. In fact, instead of taking advantage of robust and optimal linear control theory, audio amplifier designers usually focus on reducing distortion by guaranteeing open-loop linearity and implementing local and global NFB compensations [4].

The aim of this paper is to provide key concepts and information from the engineering education point of view that could help students to understand both the total harmonic distortion (THD) and the intermodulation distortion (IMD) in audio amplifiers, and to learn how to reduce these distortions using $H_\infty$ robust control techniques, which is a novel idea.

Comments on the application of Quantitative Feedback Theory to improve the performance of the operational amplifier (OP AMP) 741 can be found in [1] (in Section 1–6.8 on page 13); however such comments are not focused on Engineering Education.

## TOTAL HARMONIC DISTORTION AND INTERMODULATION DISTORTION

In this section, the definitions of the THD and the IMD to be recalled by the students are provided, and the students are expected to review the Distortion Mechanisms in the Amplifiers topic, which is taught within the Analog Electronics course.

In short, because there are non-linearities in the input stage, the voltage amplifier stage and the

* Accepted 7 December 2007.

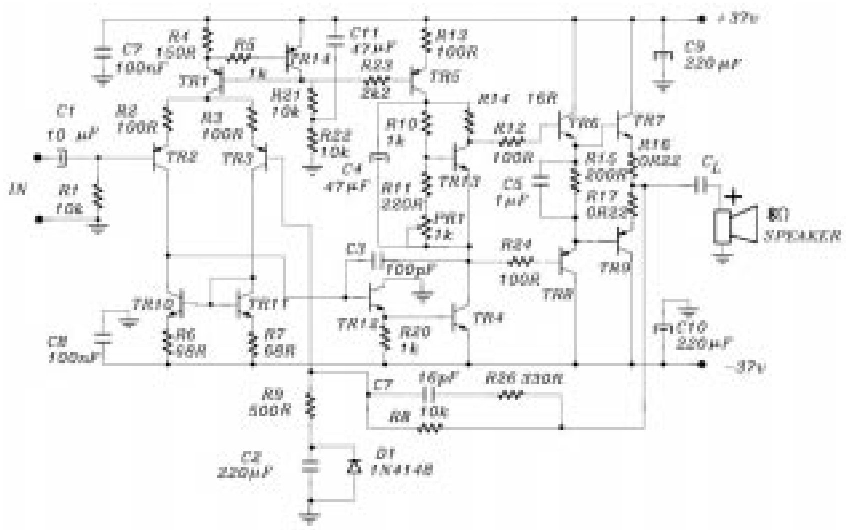


Fig. 1. Class-B ultra-linear Hi-Fi power amplifier with classical NFB compensation.

output stage of any audio amplifier, the signal is gradually distorted as it passes through these stages. This distortion (or deviation from linearity) is characterized by the THD, which is a measurement of the harmonic distortion present in the signal.

*Definition of THD*

$$\mathrm{THD} = \sqrt{\frac{P - P_1}{P}} \qquad (1)$$

where $P$ is the total output power, $P_1$ is the fundamental frequency power, and it is assumed that the input signal is a pure tone.

However, in spite of the fact that the THD is usually used to evaluate the sound quality of a sound system, it does not necessarily correspond to the hearing sensation of the human ear: the higher the order of the harmonic components, the higher the sensation of auditory distortion in the human ear [2]. Nevertheless, the THD is a useful indication of performance [3].

Finally, the task of characterizing an audio amplifier would be incomplete if we measured only its THD. An adequate bandwidth with constant gain across the audio-frequency range (i.e., from 20 Hz to 20 kHz), linear phase versus frequency response, and a very low value of IMD, among other characteristics, is also desirable. In fact, measuring the IMD of a sound system is sometimes more important than measuring its THD.

*Definition of IMD*

If the input signal of an amplifier consists of the sum of two or more pure tones, then the IMD is a nonlinear distortion that appears as the presence of new tones in the output signal that are linear combinations of the pure tones present in the input signal.

A highly regarded book in the area of audio amplifier distortion that students can reference for more detail and further examples is by Self [4]. Also, in [4], is a very helpful discussion is given about the benefits of using NFB to abate distortion.

## A CLASS-B AUDIO AMPLIFIER BASED ON CLASSICAL NFB COMPENSATION

In this section, a real-world audio amplifier [4] is presented along with the effects of NFB networks connected around some of its stages and the whole amplifier, as well.

*The audio amplifier*

The class-B audio amplifier under study is shown in Fig. 1. *TR1–3*, *TR5* and *TR14* are MPSA56 transistors. *TR4* and *TR10–13* are MPSA06 transistors. *TR6* is the MJE340 transistor. *TR7* is the MJ802 transistor. *TR8* is the MJE350 transistor. And *TR9* is the MJ4502 transistor.

In Fig. 1, the NFB factor was chosen to be equal to 26.4 dB at 20 kHz, which should give generous high-frequency (HF) stability margins. The values of the input resistor $R_1$ and the feedback arm $R_8$ are made equal and kept as small as possible, consistent with a reasonably high input, so that base current mismatch caused by beta variations will give a minimal DC offset. Furthermore, the value of $C_2$ shown in Fig. 1 gives a low-frequency (LF) roll off with $R_9$ that is –3 dB at 1.4 Hz in order to prevent an LF rise in distortion due to capacitor nonlinearity. According to Self [4], protection diode $D_1$ prevents damage to $C_2$ if the amplifier suffers a fault that makes it saturate negatively. *C7* provides some stabilizing phase advance and limits the closed loop bandwidth; $R_{26}$ prevents it upsetting $TR_3$. More information about the performance of the class-B audio amplifier shown in Fig. 1 can be found in [4].

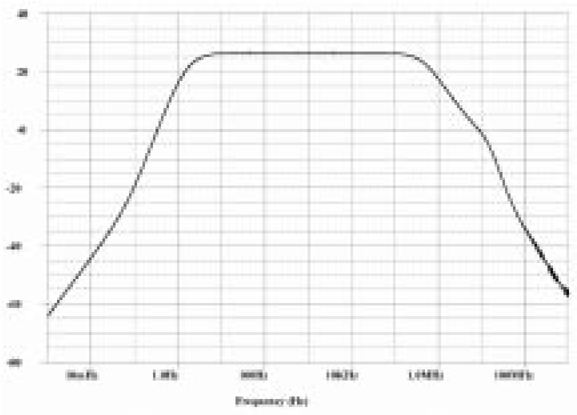

Fig. 2. Magnitude plot of the closed-loop system: magnitude (dB) vs. frequency (Hz); frequency is in log scale.

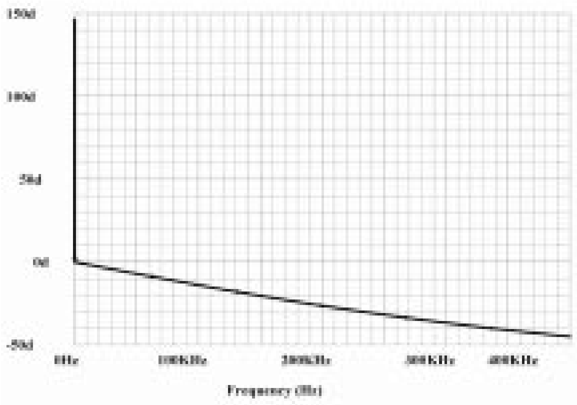

Fig. 3. Phase plot of the closed-loop system: phase (degrees) vs. frequency (Hz); frequency is in linear scale.

Table 1. Fourier components of transient response of the class-B audio power amplifier shown in Fig. 1; THD = 0.0084161%

| Harmonic number | Frequency (Hz) | Fourier component |
|---|---|---|
| 1 | $1{\cdot}10^3$ | $2.984{\cdot}10^1$ |
| 2 | $2{\cdot}10^3$ | $8.500{\cdot}10^{-4}$ |
| 3 | $3{\cdot}10^3$ | $1.065{\cdot}10^{-3}$ |
| 4 | $4{\cdot}10^3$ | $8.145{\cdot}10^{-4}$ |
| 5 | $5{\cdot}10^3$ | $8.104{\cdot}10^{-4}$ |
| 6 | $6{\cdot}10^3$ | $8.564{\cdot}10^{-4}$ |
| 7 | $7{\cdot}10^3$ | $8.765{\cdot}10^{-4}$ |
| 8 | $8{\cdot}10^3$ | $7.152{\cdot}10^{-4}$ |
| 9 | $9{\cdot}10^3$ | $8.385{\cdot}10^{-4}$ |
| 10 | $10{\cdot}10^3$ | $6.438{\cdot}10^{-4}$ |

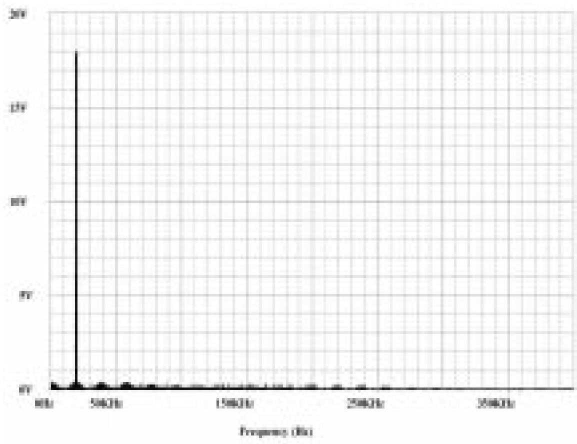

Fig. 4. IMD performance test for the amplifier in Fig. 1 delivering 40.5 W into 8 Ω; frequency is in linear scale.

*Frequency response of the audio amplifier*

Figure 2 shows the magnitude plot of the closed-loop system (from *IN* to *OUT*). The magnitude is given in units of dB. The compensated class-B Hi-Fi power amplifier has large amplification when signals at frequencies lying outside the audio band are fed into it. Also, the phase plot of the amplifier is shown in Fig. 3. Figures 2 and 3 were obtained by using PSpice simulation (*PSpice* 9.1).

Here, it should be highlighted that having a linear phase characteristic is very important for stereophonic applications because it avoids a nebulous perception of a point source [5, 6]. Phase distortion comes from nonlinear phase versus frequency response and, according to Tuinenga [6], phase distortion gives rise to 'echoes' in the output that precede and follow the main response, resulting in a distortion of the output signal when the input signal has many frequency components.

*Distortion performance of the audio amplifier*

The PSpice Fourier analysis of this circuit shows that the THD of this amplifier is about 0.0084161% at 1 kHz (see Table 1), which is a very low value of THD. Moreover, this THD is reported for the audio amplifier in Fig. 1 delivering 56 W into 8  for a single input frequency at 1 kHz, which is one of the performance characteristics typically used by Hi-Fi audio amplifier manufacturers (see the 'Fletcher–Munson curves' [7, 8] and [5]).

Finally, in order to test for the IMD of this class-B audio amplifier, Fig. 4 shows the PSpice Fourier plot of the output voltage of the amplifier for an input signal consisting of the mix of two pure sinusoids at 19 kHz and 20 kHz, respectively. In this case, the amplifier in Fig. 1 is delivering 40.5 W into 8Ω. Any intermodulation created by the mix of these two signals would appear as peaks at 1 kHz intervals across the frequency bandwidth. Also, another form of IMD would appear at 39 kHz.

From Fig. 4, it can be seen that the IMD performance of this amplifier is not very good when delivering 40.5 W into 8Ω. However, from Fig. 5, it can be seen that if the power is decreased from 40.5 W to 26.5 W, then the IMD performance of this amplifier is very good. This amplifier is called a blameless amplifier in [4]. In fact, the average listener would be very happy with this Hi-Fi audio amplifier and would not see the need to improve its distortion performance. However, the high-end sector and audio specialists would say that the distortion performance of this amplifier should be improved.

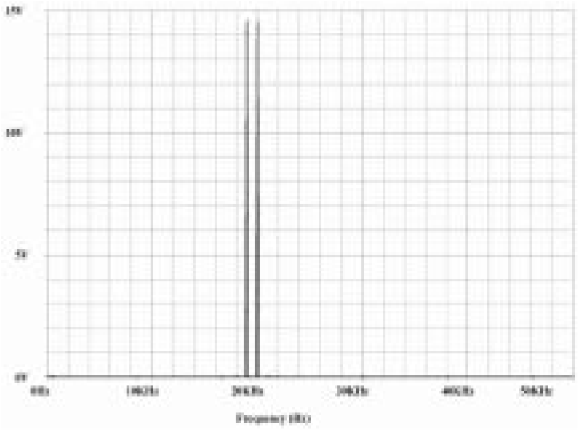

Fig. 5. IMD performance test for the amplifier in Fig. 1 delivering 26.5 W into 8 Ω; frequency is in linear scale.

## THE AUDIO AMPLIFIER BASED ON A ROBUST NFB COMPENSATION

In this section, the NFB loop around the amplifier is designed using $H_\infty$ robust control. Here, the mathematical model of the open-loop audio amplifier is obtained by using one of the basic system identification frequency-domain methods learned in the Control Systems course, and a comparative analysis between the performance of the robust amplifier and that presented in the previous section is carried out.

*System identification*

In order to control the audio amplifier in Fig. 1, the first step is to obtain the linear transfer function of the open-loop amplifier shown in Fig. 6 after having reduced internal offsets [9]. Figure 7 shows a schematic representation of DC offset with an equivalent voltage source placed at the non-inverting amplifier input terminal. For the case under study, the voltage source in Fig. 7 is 1.82315 mV.

Here, the transfer function of the open-loop amplifier in Fig. 6 was obtained by using the schematic representation of the amplifier circuit shown in Fig. 8.

In order to carry out the system identification process, the frequency response analysis of the open-loop amplifier was performed by using a PSpice AC signal voltage source (i.e., $v_{IN}$) of magnitude 1 V and phase equal to zero. In addition, a logarithmic frequency sweep with 500 points per decade in the sweep, starting at 1 mHz and finishing at 20 MHz, was used. Figure 9 shows the PSpice plot of the open-loop gain/phase versus frequency response of the amplifier, and the open-loop transfer function of this dynamic system is given by (2). The voltage gain is equal to 137.29 dB and the unity-gain bandwidth is equal to 14 MHz.

$$G(s) = \frac{90.9257 \cdot 10^6}{s + 12.4219} \tag{2}$$

*Robust controller design*

In this paper, the mixed-sensitivity $H_\infty$ control approach [10] is applied, and a method of formulating the control problem is presented below.

A general method of formulating control problems

Figure 10 shows the general control configuration for the nominal case [10], where $P$ is the generalized plant and $K$ is the generalized controller.

The control configuration shown in Fig. 10 is described by the following equations:

$$\begin{bmatrix} z \\ v \end{bmatrix} = P(s) \begin{bmatrix} W \\ u \end{bmatrix} \tag{3}$$

$$u = K(s)v \tag{4}$$

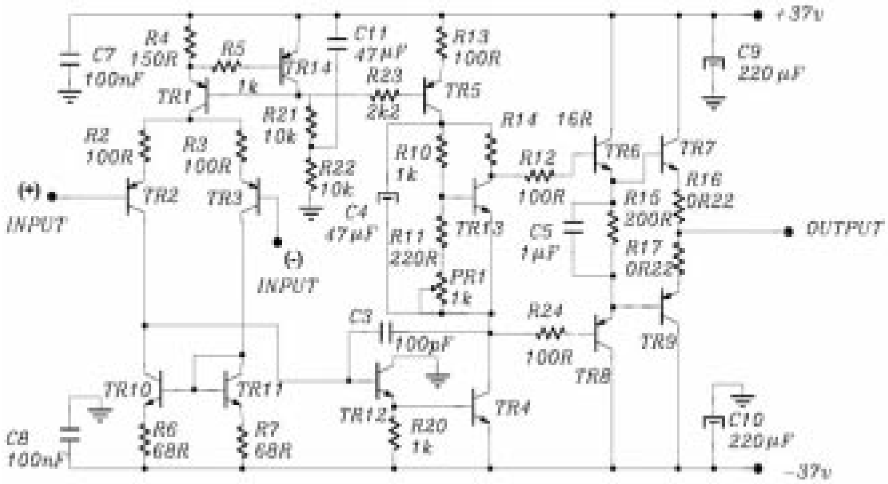


Fig. 6. Detailed schematic of the open-loop amplifier.

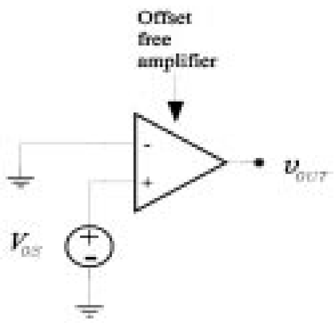


Fig. 7. Schematic representation of DC offset in the amplifier

where $P$(s) is given by

$$P(s) = \begin{bmatrix} P_{11}(s) & P_{12}(s) \\ P_{21}(s) & P_{22}(s) \end{bmatrix}$$

In addition, the state-space model of P can be written in a compact matrix form [10]:

$$P(s) = \begin{bmatrix} A & B_1 & B_2 \\ C_1 & D_{11} & D_{12} \\ C_2 & D_{21} & D_{22} \end{bmatrix} \tag{5}$$

Therefore, the lower linear fractional transformation $F_l(P,K)$ given by equation (6) describes the closed-loop transfer function from the exogenous inputs $w$ to the exogenous outputs $z$ in Fig. 10.

$$z = F_l(P, K)W \tag{6}$$

where

$$F_l(P, K) = P_{11} + P_{12}K(I - P_{22}K)^{-1}P_{21}$$

Generally speaking, the control problem is to design a controller that can guarantee that the feedback-controlled system meets the desired requirements for performance, robustness, noise and disturbance rejection, and a small magnitude of input signals. In this sense, the $H_\infty$ control is about the minimization of the peak of the largest singular value of $F_l(P,K)$.

The $H_\infty$ control problem can be stated as follows: For some $\gamma > 0$, find a stabilizing controller K for Fl(P,K) such that the $\infty$-norm of Fl(P,K) is bounded by (i.e., $\| Fl(P,K)\|_\infty < \gamma$, where $\| \ \|_\infty$ denotes $H_\infty$ norm). Furthermore, in accordance with [10], for the general $H_\infty$ control problem, the following assumptions are typically made:

I) $(A, B_2)$ is stabilizable and $(C_2, A)$ is detectable.

II) $D_{12} = \begin{bmatrix} 0 \\ I \end{bmatrix}$ and $D_{21} = [0 \ I]$.

III) $\begin{bmatrix} A - j\omega I & B_2 \\ C_1 & D_{12} \end{bmatrix}$ has full column rank for all $\omega$.

IV) $\begin{bmatrix} A - j\omega I & B_1 \\ C_2 & D_{21} \end{bmatrix}$ has full row rank for all $\omega$.

where $\omega$ denotes angular frequency (rad/s) and $I$ is the identity matrix. The MATLAB function ***hinfsyn*** can be used to obtain the $H_\infty$ controller and a general control configuration including model uncertainty is shown in Fig. 11, where D is the set of all possible uncertainty [10].

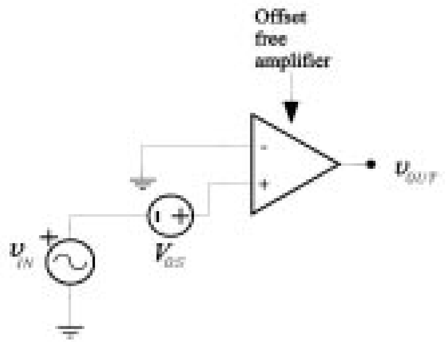


Fig 8. Circuit used to carry out the identification of the open-loop amplifier, taking account of DC offset.

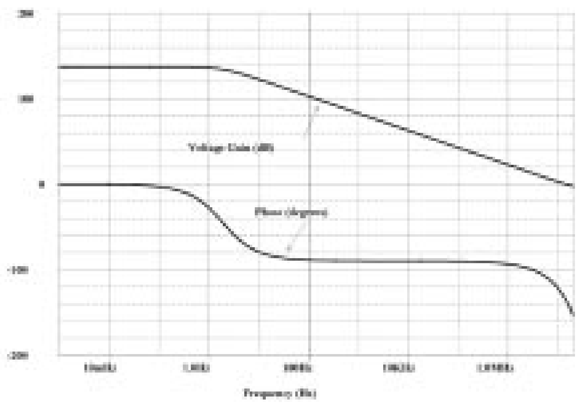

Fig. 9. PSpice plot of the open-loop gain (dB)/ phase (degrees) vs. frequency (Hz) of the open-loop amplifier; frequency is in log scale.

Mixed-sensitivity $H_\infty$ control

In this methodology for $H_8$ controller design, the designer shapes the sensitivity function $S$ (for performance) along with either the closed-loop transfer function $KS$ (to penalize large inputs) or

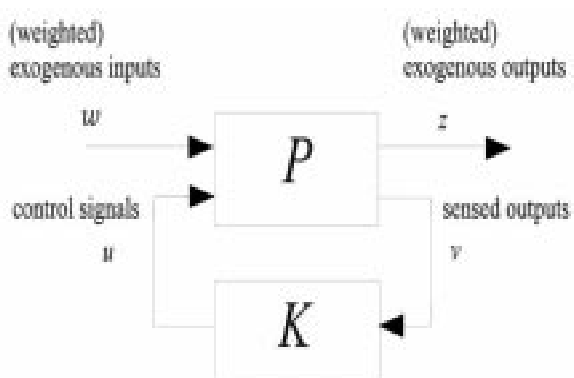


Fig. 10. General control configuration for the nominal case.

the complementary sensitivity function $T$ (for robustness and to avoid sensitivity to noise) or both closed-loop transfer functions $KS$ and $T$.

In order to solve the regulation problem presented in this material, the closed-loop transfer functions $S$ and $KS$ are shaped, and the one-degree-of-freedom control configuration shown in Fig. 12 is used, where $G$ is the open-loop transfer function given by equation (2), $w_P$ is a lowpass filter that is used to bound the maximum singular value of $S$ in order to reject disturbance $d$, and $w_u$ is a constant that is used for limiting the control energy used and for robust stability against additive plant perturbations. It should be said that in this paper $w_u$ is chosen to be a constant. But, in general, it is a highpass filter with crossover frequency approximately equal to the desired closed-loop bandwidth.

For the case under study, the cost function is given by

$$||F_l(P,K||_\infty \left|\left| \begin{bmatrix} W_P S \\ W_u KS \end{bmatrix} \right|\right|_\infty \qquad (7)$$

where $S$ is shaped by using the lowpass filter

$$w_p(s) = \frac{\frac{s}{M} + \omega_B^*}{s + \omega_B^* A} \qquad (8)$$

where

$M = 1.1 \qquad \omega_B^* = 2.5133$ Mrad/s $\qquad A = 10^{-6}$

and $KS$ is shaped by using

$$w_u(s) = 1 \qquad (9)$$

Figure 13 shows an asymptotic plot of

$$\left|\frac{1}{w_P(j\omega)}\right|,$$

which is the upper bound on $|S|$. From Fig. 13, it can be seen that

$$\left|\frac{1}{w_P(j\omega)}\right|$$

is equal to $10^{-6}$ at low frequencies and equal to 1.1 at high frequencies. In addition, if the closed-loop bandwidth is defined as the frequency where $|S(j\omega)|$ first crosses –3 dB from below, then the closed-loop bandwidth is approximately equal to 2.5133 Mrad/s.

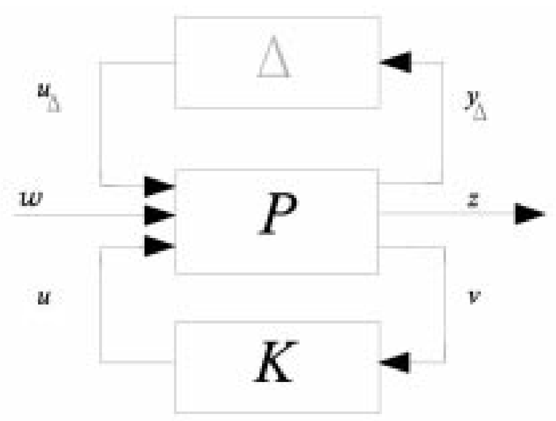


Fig. 11. General control configuration including model uncertainty.

From Fig. 12, it can be seen that

$$z_P = w_P w + w_P G u \qquad (10)$$

$$z_u = -w_u u \qquad (11)$$

$$v = -w - Gu \qquad (12)$$

Therefore, the elements of the generalized plant $P$ given by equation (3) are as follows:

$$P_{11} = \begin{bmatrix} w_P \\ 0 \end{bmatrix} \qquad (13)$$

$$P_{12} = \begin{bmatrix} w_P G \\ -w_u \end{bmatrix} \qquad (14)$$

$$P_{21} = -1 \qquad (15)$$

$$P_{22} = -G \qquad (16)$$

Taking into consideration the above information and using the MATLAB function hinfsyn, the $H_\infty$ controller given by equation (17) was obtained. For this controller, $g = 0.9097$.

$$K(s) = \frac{10.5533 \cdot 10^8 (s + 12.4219)}{s^2 + 347.3115 \cdot 10^8 s + 872.8889 \cdot 10^8}$$

Implementation and frequency response of the robust audio amplifier

The robust audio amplifier is shown in Fig. 14.

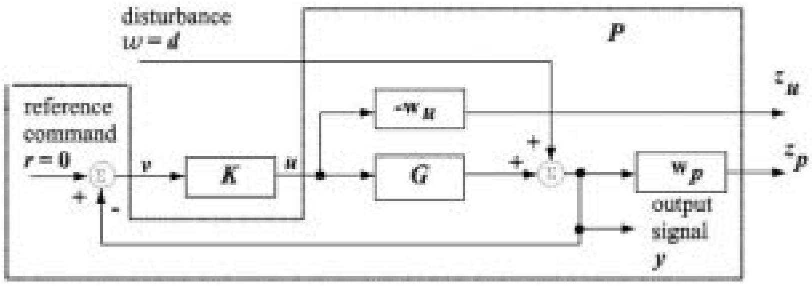


Fig 12. Block diagram of the one-degree-of-freedom control configuration for the $S/KS$ optimization problem.

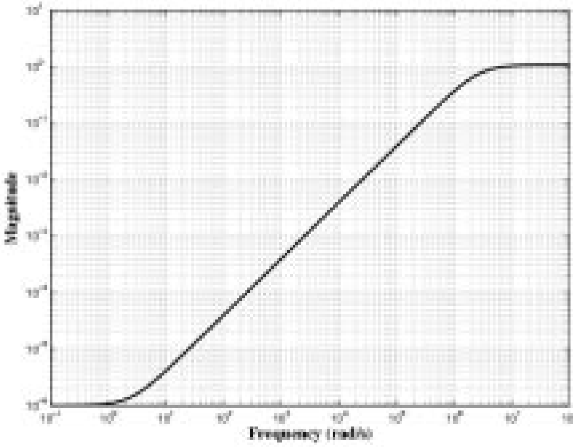


Fig. 13. Asymptotic plot of $\left|\frac{1}{w_P(j\omega)}\right|$, where $w_P$ is given by equation (8).

Figure 15 shows the PSpice plot of the magnitude plot of the new closed-loop system (from IN to OUT) and its phase plot is shown in Fig. 16.

The PSpice Fourier analysis of the robust audio power amplifier shows that the THD of this amplifier is about 0.0011335% at 1 kHz (see Table 2), which is 7.425 times better than the THD of the audio amplifier shown in Fig. 1. This THD is also reported for the audio amplifier in Fig. 14 delivering 56 W into 8 Ω. The results presented in Table 2 clearly show that the THD performance of the robust audio amplifier is superior to that of the non-robust audio amplifier. Furthermore, the IMD performance of the robust amplifier (see Fig. 17) is better than that of the non-robust amplifier (see Fig. 4).

In order to make good engineering judgement, a comparison between the IMD performance of both audio amplifiers (see Figs 1 and 14) is made in Figs 18 and 19 but in the interval [0, 400 kHz]–[0, 1 V]. Finally, from Fig. 20, it can be seen that if the power is decreased from 40.5 W to 26.5 W, the IMD performance of the robust audio amplifier is very good.

## RESULTS OF SURVEY OF TWENTY-FIVE STUDENTS

Generally speaking, the results of the survey were quite positive. Twenty five students participated voluntarily in the study of this material and were asked to fill out a survey and give written insights into the ways in which this material contributed positively or negatively to their overall experience in the engineering curriculum. Table 3 shows the evaluation data for the questions related to the content of this material. The maximum score was 10.

The predominant feedback was that the material provided them with an understanding of distortion mechanisms in audio amplifiers and $H_\infty$ robust control. It gives the students the opportunity to use what they have already learned and provides reinforcement of material cover in other courses.

Also, typical comments were the following:

1. This material helps to better understand robust control techniques and their use to solve typical problems in audio engineering.
2. The PSpice simulation of real-world problems is very good. To see that the fusion of the control theory and the low distortion audio amplifier design theory works in practice is really helpful.

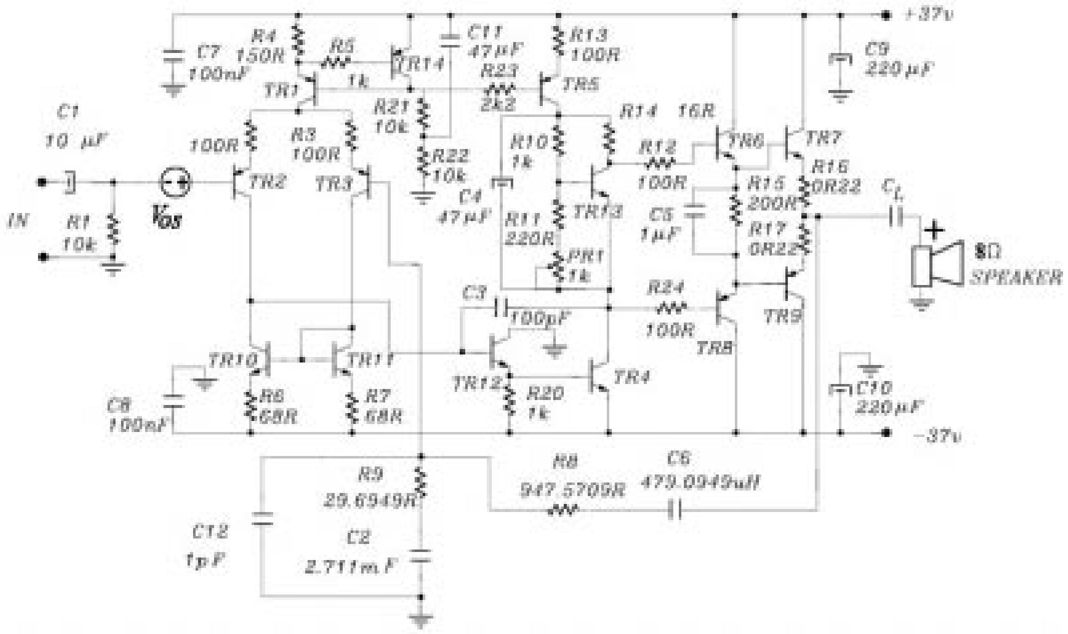


Fig. 14. Class-B audio amplifier with a robust compensation.

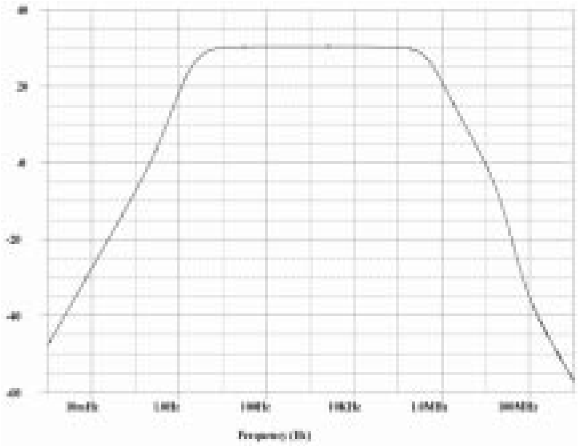

Fig. 15. Magnitude plot of the new closed-loop system: magnitude (dB) vs. frequency (Hz); frequency is in log scale.

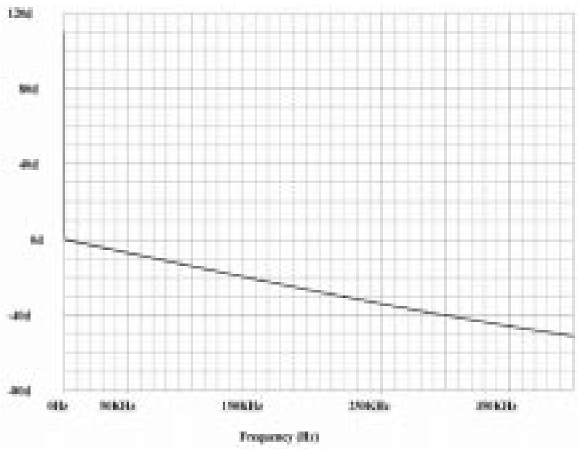

Fig. 16. Phase plot of the new closed-loop system: phase (degrees) vs. frequency (Hz); frequency is in linear scale. Distortion performance of the robust audio amplifier

Table 2. Fourier components of transient response of the new class-B audio power amplifier shown in Fig. 14; THD = 0.0011335%

| Harmonic number | Frequency (Hz) | Fourier component |
|---|---|---|
| 1 | $1{\cdot}10^3$ | $2.962{\cdot}10^1$ |
| 2 | $2{\cdot}10^3$ | $9.936{\cdot}10^{-5}$ |
| 3 | $3{\cdot}10^3$ | $2.697{\cdot}10^{-4}$ |
| 4 | $4{\cdot}10^3$ | $8.562{\cdot}10^{-5}$ |
| 5 | $5{\cdot}10^3$ | $7.796{\cdot}10^{-5}$ |
| 6 | $6{\cdot}10^3$ | $2.680{\cdot}10^{-5}$ |
| 7 | $7{\cdot}10^3$ | $7.892{\cdot}10^{-5}$ |
| 8 | $8{\cdot}10^3$ | $6.855{\cdot}10^{-5}$ |
| 9 | $9{\cdot}10^3$ | $4.299{\cdot}10^{-5}$ |
| 10 | $10{\cdot}10^3$ | $5.642{\cdot}10^{-5}$ |

3. This is the first time that we have applied $H_\infty$ robust control to improve the distortion performance of audio amplifiers.
4. In contrast to conventional Analog Electronics laboratory courses, which sometimes frustrate and intimidate those who are curious about how audio amplifiers work, this material allows us to experiment with the internal components of an audio amplifier in order to improve its distortion performance.

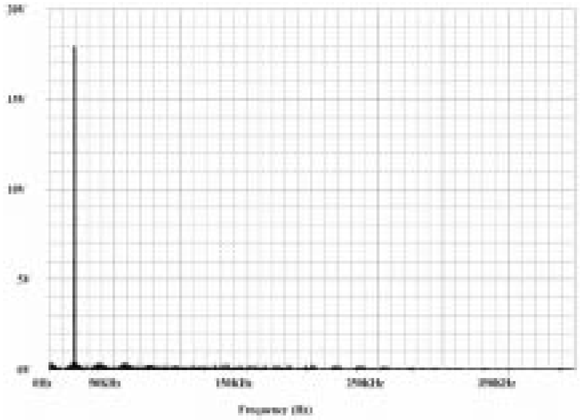

Fig. 17. IMD performance test for the amplifier in Fig. 14 delivering 40.5 W into 8 Ω; frequency is in linear scale.

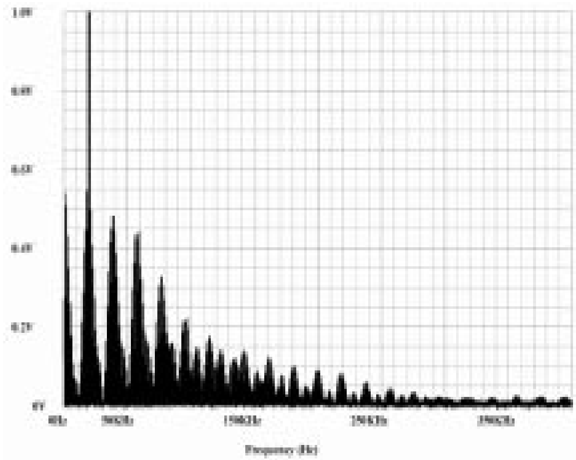

Fig. 18. Information shown in Fig. 4, but in the interval [0, 400 kHz]–[0, 1 V].

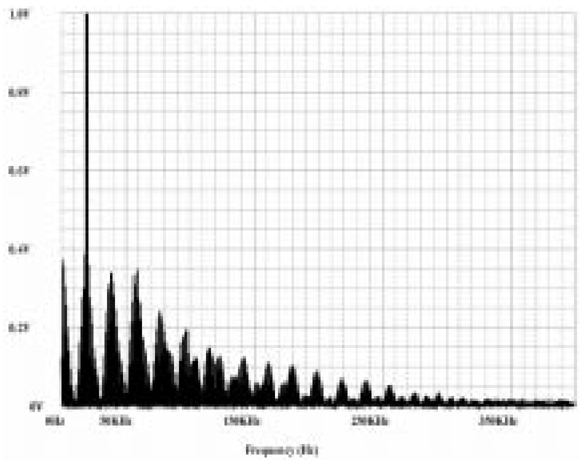

Fig. 19. Information shown in Fig. 17, but in the interval [0, 400 kHz]–[0, 1 V].

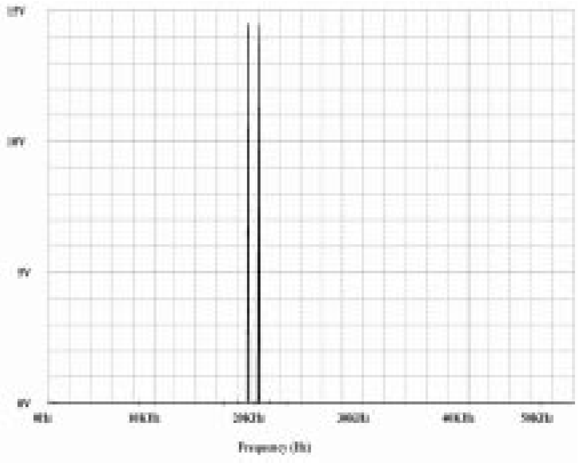

Fig. 20. IMD performance test for the amplifier in Fig. 14 delivering 26.5 W into 8 Ω; frequency is in linear scale

Table 3. Student evaluation results

| Questions | Average score |
|---|---|
| Overall, this is a good material to help students to understand the importance of the application of robust control techniques to improve the distortion performance of audio amplifiers. | 9.54 |
| Overall, the connection among this material, other materials and other subjects is good. | 8.96 |
| This material helps to better understand distortion mechanisms in audio amplifiers and contributes to the student's overall academic growth. | 8.81 |
| This material arouses the interest of the students in the subject and is well related to the lecture courses. | 9.36 |
| This material can contribute positively to participate in debates and discussions in classes. | 9.10 |

## CONCLUSIONS

In this paper, instructional material for Analog Electronics and Control Systems laboratory sessions aimed at designing a robust class-B ultra-linear Hi-Fi audio amplifier were provided. Here, in order to assist students to design and implement a laboratory project aimed at improving the distortion performance in audio amplifiers by using robust control techniques, a novel example based on a real-world application of $H_\infty$ control has been described in detail, giving students the opportunity to simulate real-world problems and find solutions that involve trade-offs and the use of engineering judgment.

*Acknowledgments*—The author would like to thank Professors Jordi Margalef and Xavier Senmartí at the Universitat Ramon Llull, Barcelona, Spain, for their comments on this paper and for carrying out part of the survey with their students at their university. This work has been supported by the Universidad Politecnica de Madrid, Spain.

## REFERENCES

1. C. Houpis and S. Rasmussen, *Quantitative Feedback Theory*, Marcel Dekker (1999).
2. H. Yue Lin, Measurement of Auditory Distortion with Relation between Harmonic Distortion and Human Auditory Sensation, *IEEE Transactions on Instrumentation and Measurement*, 35, 1986, pp. 195–200.
3. J. N. Ross, *The Essence of Power Electronics*, Prentice-Hall Europe, London (1997).
4. D. Self, *Audio Power Amplifier Design Handbook*, 4th edn, Newnes, Oxford (2006).
5. J. W. Dowling and D. L. Harwood, *Music Cognition,* Academic Press, San Diego, California (1986).
6. P. W. Tuinenga, *SPICE: A Guide to Circuit Simulation and Analysis Using PSpice*, 3rd edn, Prentice-Hall, Englewood Cliffs, New Jersey (1998).
7. H. Fletcher and W. A. Munson, Loudness, its definition, measurement, and calculation, *Journal of the Acoustical Society of America*, 5, 1933, pp. 82–105.
8. J. R. Hassall and K. Zaveri, *Acoustic Noise Measurements*, Brüel & Kjaer, (1979).
9. W. M. C. Sansen, *Analog Design Essentials*, Springer, The Netherlands (2006).
10. S. Skogestad and I. Postlethwaite, *Multivariable Feedback Control*, 2nd edn, John Wiley & Sons (2005).

**Wilmar Hernandez** gained his Electronic Engineer degree and his Specialist degree in microelectronics from the Instituto Superior Politecnico Jose Antonio Echeverria, Havana, Cuba, in 1992 and 1994, respectively. Also, he gained his M.S. in signal treatment and his Ph.D. in electronic engineering from the Universitat Ramon Llull, Barcelona, Spain, in 1997 and 1999, respectively. Currently, he is an Associate Professor of Telecommunication Engineering at the Universidad Politecnica de Madrid, Spain, where he teaches both circuits and systems courses, and sensors and actuators courses. Furthermore, he conducts research in sensors, electronic circuits, robust control and signal processing.