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Chapter

Cost-Effective Design of Amplifiers for Hearing Aides Using Nullors for Response Matching

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Abstract

This chapter starts reviewing Fixator-Norator Pairs (FNP) as an effective tool used to design analog amplifiers for a prescribed bandwidth and frequency profile. Among number of cases and applications, designing for hearing aides are particularly important, where the hearing frequency profiles, known as audiograms, are changing from person to person, and also for a person by the age. The design is mainly focused on front-end or stand-alone amplifiers. In case of a front-end the response from the amplifier can be digitized, properly controlled and adjusted to fit the digital application. Here is how the design proceed. For a given audiogram, an Audiogram Generator Circuit (AGC) is initially constructed. This AGC, usually a complete passive circuit, produces a frequency response that closely matches with the audiogram, obtained from a hearing impaired patient. The AGC is then embedded in an amplifier circuit where a fixator is placed at its output port, "forcing" the amplifier to generate the desired output frequency response profile. A flat band frequency response, for example, compensates the hearing losses and provides a uniform hearing to the patient in the entire audio bandwidth. The amplifier can be further enhanced to perform other requirements, for example, to cancel undesirable noises in certain frequencies or to magnify the voice in critical frequencies for clarity. Another alternative design methodology is also introduced in this chapter, which uses the negative feedback technique.

Keywords: Analog circuit design, audio amplifiers, feedback theory, fixator-norator pairs, frequency profiles, hearing aids, nullors

1. Introduction

Hearing aid market is definitely dominated by fully digital hearing aids. With many recent advancements in the industry the prices are also keep rising and getting almost unaffordable for some hearing- impaired patients. This chapter provides a simple and very cost effective method for the design and implementation of stand-alone analog amplifiers or pre-amplifiers for digital hearing aids. Although somewhat behind in the technology and the market, analog hearing aids can still provide some advantages in some aspects over the digital technology.

The chapter is the extended version of [1], and the objective here is to design amplifiers that exhibit frequency responses that can vary and match with any specific frequency profile in demand. In this chapter, we are considering amplifiers that are applicable to hearing aid designs. There are several main criteria associated with this design as:

- The design needs to be simple and highly modular. By this modularity, we mean to separate the active device, as an engine, from the rest of the circuit, as the controllers.

- To be easily adaptable to variations, either for different hearing-impaired patients or the natural changes happening in the hearing situation of an individual over time.

- Low cost and affordable with high quality.

2. Fixator Norator pairs and their properties in design for frequency profiles

Fixator-Noratpr Pairs and their properties in analog circuit designs are covered in [2]. A fixator, denoted by $Fx(V_j, I_j)$ or $Fx(I_j, V_j)$, symbolically shown in **Figure 1**, is a two terminal component with both its current I_j and voltage V_j specified. A nullator, denoted by $Fx(0, 0)$, is a special case of a fixator where V_j and I_j are both zero. So by definition, a fixator can be assigned to a design constraint to keep it unchanged during the design process. Then a paring norator, with its V and I unspecified, can provide the conditions in the circuit to allow the fixator to hold onto the values. Hence, a fixator and its paring norator work together to satisfy the Kirchhoff Laws [3], and they must be mutually sensitive to each other. It is important to note that, because a fixator needs to keep its variables (I and V) as designated, its pairing norator must be ultra-sensitive to small variations in the fixator in order to keep the fixator values unchanged.

A major property of a fixator is its ability to stick to a design constraint, whether fixed or variable in time or frequency, based on a pre-specified setting. For example, a fixator can be assigned to a circuit port to keep its frequency response close to a given frequency profile. In return the pairing norator must be capable of providing the necessary condition in the circuit for the fixator to operate. In short, a fixator is used to keep a design constraint as specified, shifting the problem to the

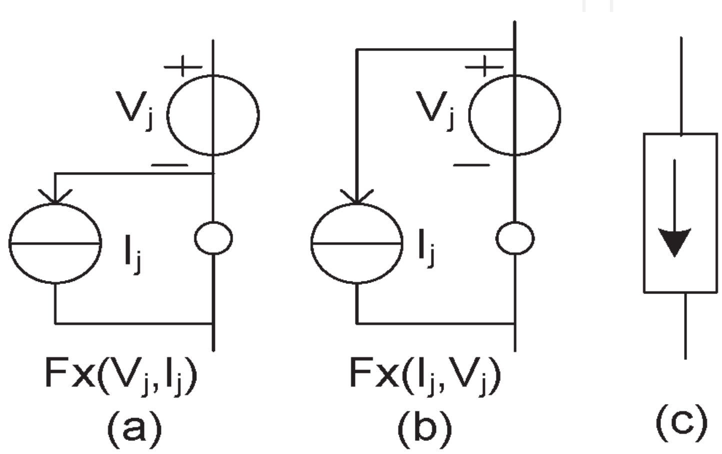

Figure 1.
Fixators; (a) a voltage fixator; (b) a current fixator; (c) the symbol.

design of a two terminal component/circuit that needs to replace the pairing norator. This is, in fact, the key property of a fixator that we are able to use in this chapter to design amplifier circuits that exhibit some specified frequency profiles and bandwidths.

Next, we are going to investigate how this property of a fixator works for us to design hearing aid amplifiers.

3. Frequency profiles matching in hearing aid applications

Consider designing an analog amplifier for a hearing aid application. Given a hearing profile (audiogram) of a hearing impaired patient, the question is how can we compensate for the hearing losses of the patient within the entire dynamic frequency range? We may go even further and design for a response that is beyond the mere compensation of the hearing losses, but enhancing or reducing the response in certain frequency areas as needed. For example, if the individual works in a factory and he/she is exposed to certain excessive sounds (noises) within certain frequencies, the hearing device must be capable of acting as a noise cancelation device [4], helping to reduce the noise as it provides amplification in other areas of the bandwidth.

This last point might be of interest to those working in the occupational technology, construction workers, and those working long hours with heavy equipment and machinery. Other applications might be in public health services such as in nursing homes to enhance certain alarms like passing vehicles, and so on, for the elderly safety. What is interesting in our analog hearing aid is that, to add those extras, such as noise cancelations or sound enhancements to the system all we need to do is to redesign the passive portion of the system without touching the active (amplifier) device.

So, we can define two objectives here: 1) compensate for the hearing losses and make it uniform within the entire dynamic frequency range, and 2) add a certain selective frequency response profile on top of the flat normal hearing. In other words, given the audiogram of a hearing impaired patient and also a desire hearing frequency profile constructed for the patient's need, how can we design an amplifier that satisfies both?

To put the problem into a mathematical perspective, suppose $H(s)$ denotes the audiogram of a hearing impaired patient, $F(s)$ is the final desirable voice spectrum that is tailored for the individual, and $T(s)$ is the transfer function of the hearing amplifier that provides such a response. Then

$$F(s) = T(s) * H(s) \tag{1}$$

To simplify the problem, we split it into two cases, just described. First, we only assume a flat frequency response for the final hearing comprehension, i.e., $F(s) = 1$ for the entire frequency bandwidth. In the second case, we try to enhance the response to follow a certain desirable frequency profile $F(s)$. We continue our design strategy for the first case here and will follow it for the second case in a later Section.

4. Design for flat frequency response

Suppose $H(s)$ is the transfer function of an audiogram, being represented by:

$$H(s) = N(s)/D(s) \tag{2}$$

Then by referring to Eq. (1) and assuming $F(s) = 1$, the amplifier response, $T(s)$, becomes

$$T(s) = H(s)^{-1} = D(s)/N(s) \tag{3}$$

So, the objective here is to design an amplifier that has a frequency response profile which is the inverse of an audiogram of our choice. In addition, this amplifier must be modular and adaptable to the changes that might happen to the hearing profile (audiogram). This change might be either due to the aging, or the amplifier may be used for another audiogram (patient) all together.

There are two known methods we can use for this functional inversion. One method is to apply the FNP technique as we introduced before, and the other method is to uses the negative feedback procedure [5], which is well known in control theory. We will introduce both methodologies in this chapter, although our preference and emphasis will be more on the former technique, as it is shown to be more reliable and accurate.

5. FNP implementation of analog hearing aid amplifiers

This implementation uses an FNP as a design tool. However, the FNP is later replaced with a high gain operational amplifier when the amplifier is constructed. Before we go into the details, here is the Problem Statement.

Problem statement – Given an audiogram of a hearing-impaired patient, design a front-end or stand-alone analog amplifier that is fully adaptable and has a wide voice dynamic range covering the audio range from 250 Hz to 8 KHz, as specified in the audiogram [1].

Design procedure – The design proposed is modular with two parts: a) a controlling circuit, generating the hearing loss frequency profile as the output, and b) an amplifier acting as an engine module for the system. The control unit must be closely equivalent to the patient's audiogram, and any performance variations, such as tuning and modifications, are done on this module, which is usually a passive circuit. Therefore, the design of hearing aid is mainly concentrated on the design of the passive control unit, leaving the amplifier undisturbed during the application. This means, once the amplifier (engine) is designed it is left unaltered, and all other variations and adaptations are done on the controlling module. This is one of the main criteria of the system, where the variations and control is concentrated on the passive unit, which is more stable and design friendly.

Let us begin our design procedure from the transfer function $T(s)$, given in Eq. (3). **Figure 2** shows an audiogram taken from the left ear of a hearing impaired patient. Notice that the hearing loss is quite large, and it is more than 60dB at high pitch voices. So, to compensate for this loss we need to use an amplifier with high gain, getting to 60 dB or higher at high frequencies. A typical amplifier suitable for this design can consist of one or two stage of Op-Amps with wide enough bandwidth. Next, we proceed with the design of the control module.

Control Unit - Our next stage of the design is to construct the controlling module for a flat comprehended hearing profile ($F(s) = 1$). The module must be so designed that it generates an output frequency profile duplicated from the selected audiogram, or simply have a transfer function close to H(s). Apparently, because of the losses in the magnitude of the response, the controlling circuit, called *Audiogram Generator Circuit (AGC),* can be totally passive RC (or RLC) circuit.

There are different methods available to construct such an AGC, and because it is a passive circuit its design and synthesis can be quite straight forward [6]. A more

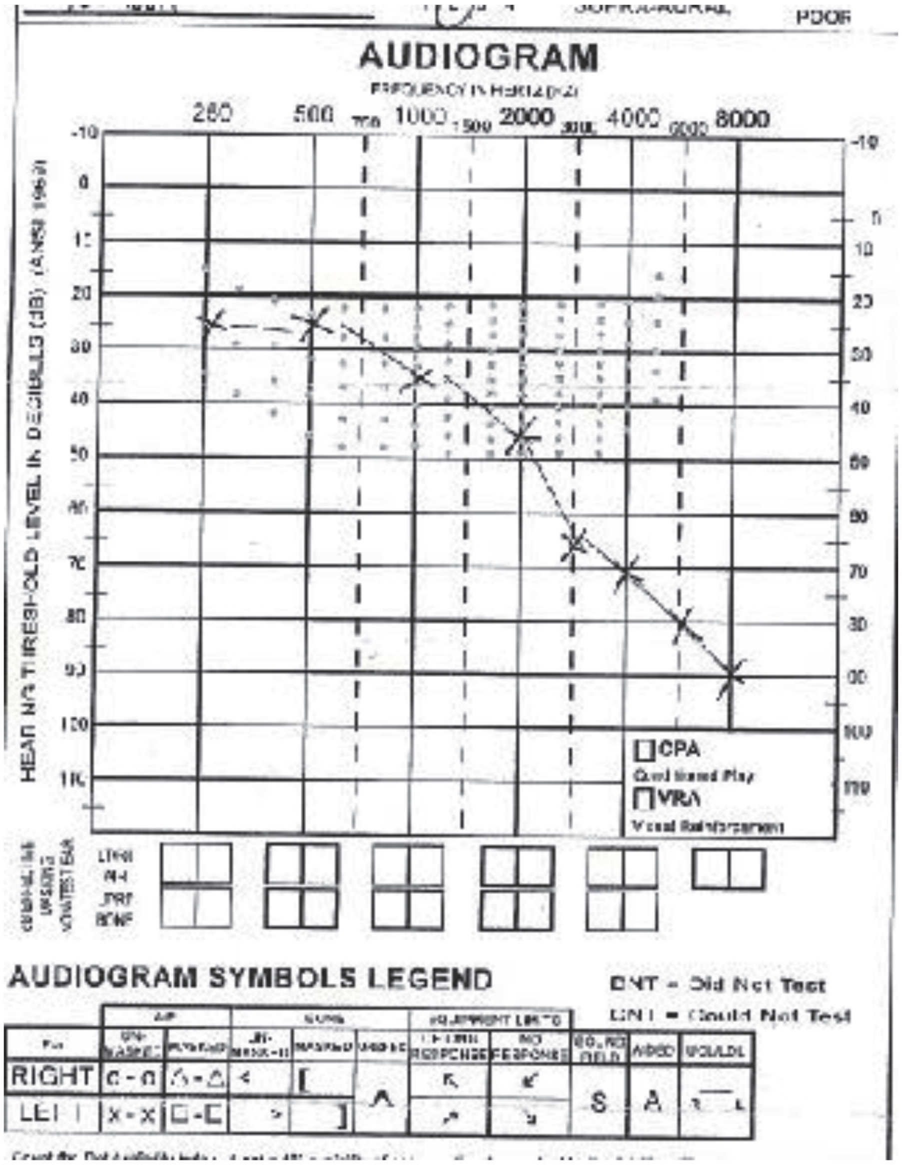

Figure 2.
Audiogram of the left ear of a patient with hearing impairment.

practical design technique is to estimate the locations of the poles and zeros for a given audiogram first, and then construct a AGC that closely displays those poles and zeros, and hence, nearly mimics the audiogram [7]. An alternative design method is also presented in [8]. Here, in this chapter, we follow an ad hoc technique where we first try to assemble an RC ladder circuit to produce an AGC with the frequency profile close to the audiogram. We then modify the circuit and add more ladder stages if necessary to get it right and accurate enough. We can always leave some room for on (application) site tuning, of course.

Another issue to pay attention to here is the phase angle. The experiment show that we need to be concern about the phase delay in an AGC as well. Because of the reactive elements (C and L) in the circuit, phase delay is generated, which causes time delay in the signal processing. In case this time delay is uniformly distributed throughout the frequency spectrum, i.e., the phase in linear vs. the frequency, then the group delay will be constant and the uniform delay only causes a constant delay between the actual voice (signal) and what is received and comprehended through the hearing aid. However, in case the time delay is dependent on the frequency of the signal, and the time delay variation is large then it may cause poor fidelity and distortion in the comprehended voice. So, for a reliable design we need to pay attention to both magnitude and the phase of the signal getting out of an AGC.

Presently, we consider two AGC circuits given in **Figure 3(a)** and **(b)**, and their symbolic representation in **Figure 3(c)**. **Figure 4** shows the magnitude frequency responses of both AGCs in comparison with the actual audiogram. To further compare the two circuits, both the magnitude and the phase Bode plots are shown in **Figure 5(a)** and **(b)**. In comparing their responses, we realize that the RC_1 module, also structurally more involved, is showing more accurate results than the RC_2 module. Notice the followings points in the response of the RC_1 module: 1) its magnitude is closer to that of the audiogram, and 2) its phase delay is almost linear, providing a nearly constant group delay. So, we have two choices to select one. Either select the RC_1 module (**Figure 3(a)**) for less distortion and better comprehended voice, or alternatively chose the RC_1 module for its simplicity.

However, we may still need to observe the roots (poles and zeros) of the modules, in case we may want to modify the responds for a better fit. To clearly identify the roots, we use a technique initially introduced in [9]. This technique converts the real axis roots (poles and zeros) of an RC circuit to roots on the imaginary axis where the sweeping excitation signal encounter with the roots and so generates

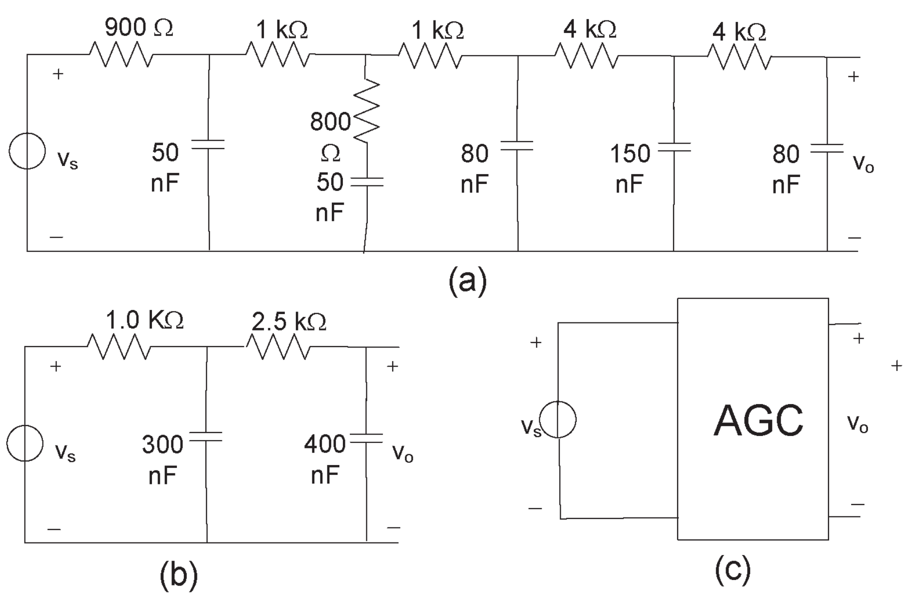

Figure 3.
(a) And (b) two AGCs matching with the audiogram given in ***Figure 1****; (c) the AGC block diagram.*

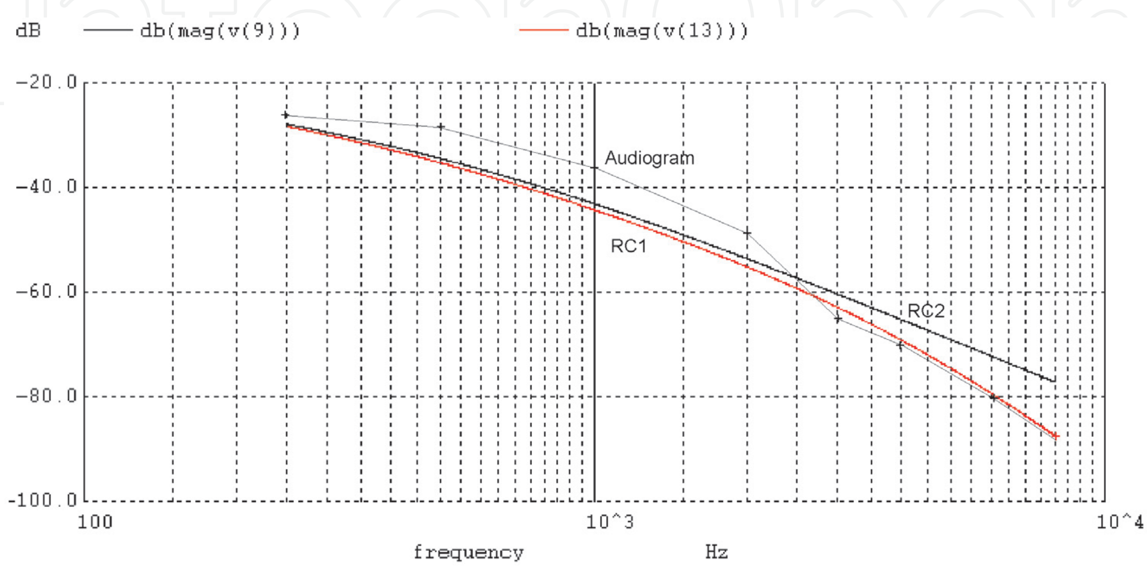

Figure 4.
Comparing the frequency responses from the two AGC candidates, RC_1, and RC_2 with the audiogram.

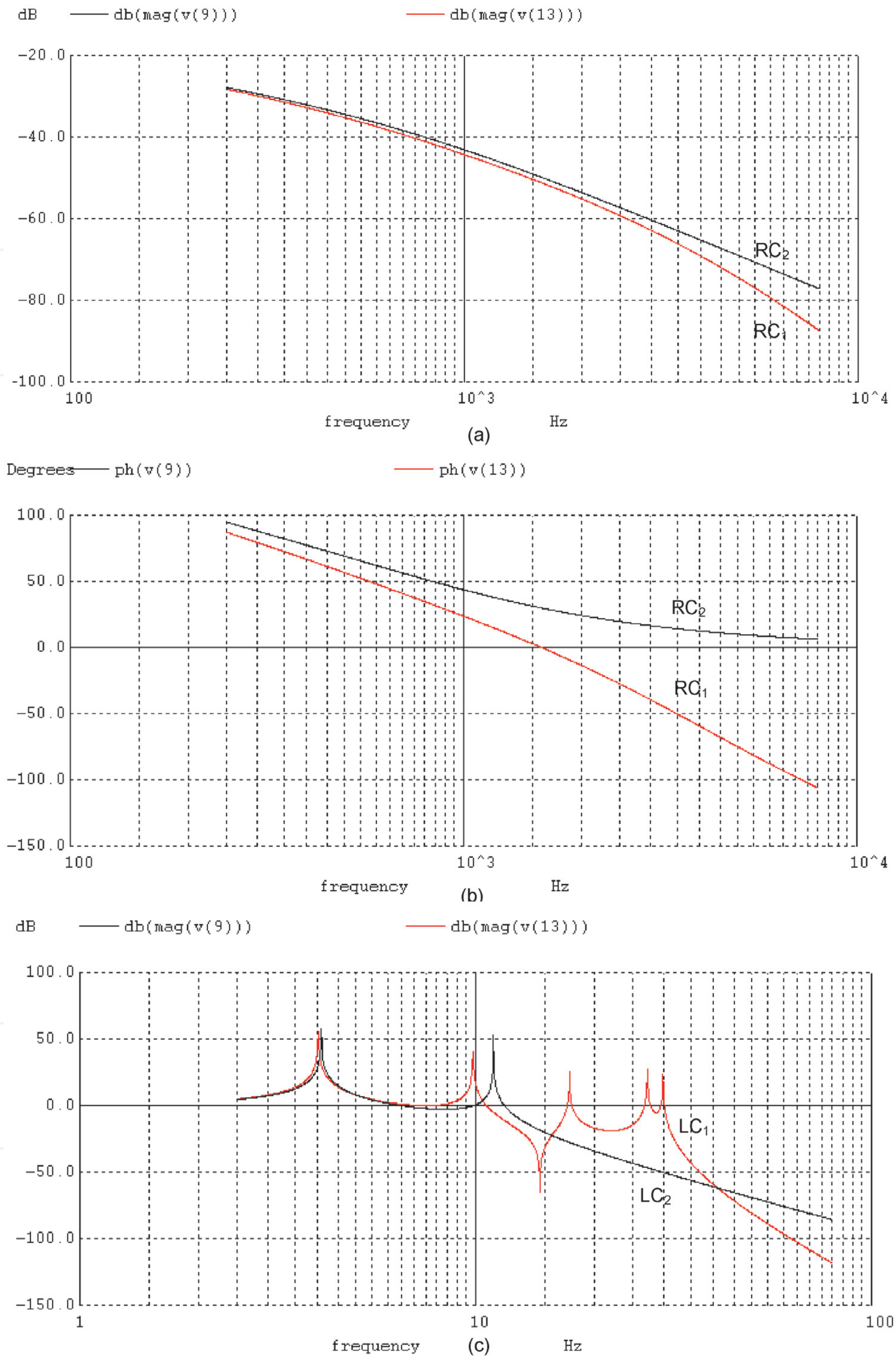

Figure 5.
(a) And (b) the frequency response of the two AGC corcuits; (c) the frequency response of the corresponding LC circuits.

peaks and notches. To implement this technique for our case we first need to create two LC circuits, LC_1 and LC_2 for RC_1 and RC_2, respectively. To create LC_1 and LC_2 we need to go through the following steps:

1. Change all resistors (R) in the RC circuit into inductors (L) with the same values.

2. The controlled sources and the controlling variables (I and V) must be of the same kind. So, for example, a voltage controlled current source VCCS must be changed to either a current controlled current source (CCCS) or to a voltage controlled voltage source (VCVS).

3. Simulate the LC circuit and generate the Bode plots, and then rescale the frequency axis.

Here is how it works.

Corresponding LC circuits – As fully explained in [9], it is simply proven that, if LC_i is the corresponding LC circuit of an RC circuit, RC_i, then for any real axis root ω_{RC} in RC_i there exist a pair of conjugate roots $\pm\omega_{LC}$ on the $j\omega$ axis for LC_i so that they are related through the relationship $\omega_{RC} = \omega_{LC}^2$, or in the log format the scaling factor is specified by

$$\log(\omega_{LC}) = \log(\omega_{RC})/2 \tag{4}$$

The advantage of getting the poles and zeros through the corresponding LC circuit is that, we can access the actual and accurate locations of the roots in terms of peaks and notches that ultimately guide us into a better design of the AGC. This means, we can study the location of the real axis roots of an AGC, make appropriate changes to the locations of the roots so that the frequency response of the AGC gets close enough to the actual audiogram (**Figure 1**). **Figure 5(c)** shows two such plots for the corresponding LC circuits LC_1 and LC_2. By observing the plots, we can extract several conclusions essential to the design. For instance, to produce a nearly constant group delay we need to create a balance between the poles and zeros of the circuit, as poles produce more lags and zeros generate more leads in the phase angle. Referring to our case of the AGCs, Bode plots in **Figure 5(c)**, we notice five poles and one zero for LC_1 transfer function that are well distributed within the bandwidth region. This, as shown in **Figure 5(b)**, produces a close to linear phase shift spanning about 200 degrees. Whereas for LC_2 the phase shift is far from linear distribution. So, the better choice for this design is clearly RC_1, although the circuit is more involved with more components. However, for the reason that is mentioned in Example 1, we choose RC_2 as the selected AGC for our design. This concludes our control unit (AGC) design.

Amplifier: Now that we have done with the AGC design, our next task is to design the amplifier and the system all together. In this design we must come up with constructing the transfer function, $T(s)$, given in Eq.(3). As we notice, the roots of $T(s)$ are the same as those of the AGC ($H(s)$) but the opposite, i.e., the poles of $H(s)$ become the zeros of $T(s)$ and vice versa. A new and rather simple method to realize $T(s)$ is to use an FNP as a tool and later replace it with real components [2]. For this implementation we start with the circuit in **Figure 6**, showing an AGC circuit with an audio input signal connection that has a unit amplitude for the entire frequency bandwidth. As expected, this input signal generates an output close to the designated audiogram. Now, we may change the problem statement and ask the following question. What do we need to connect at the input port of the AGC in order to get an output signal with unit amplitude within the audio bandwidth (250 Hz to 8 KHz)? To answer this question we refer to **Figure 6(b)**, where a

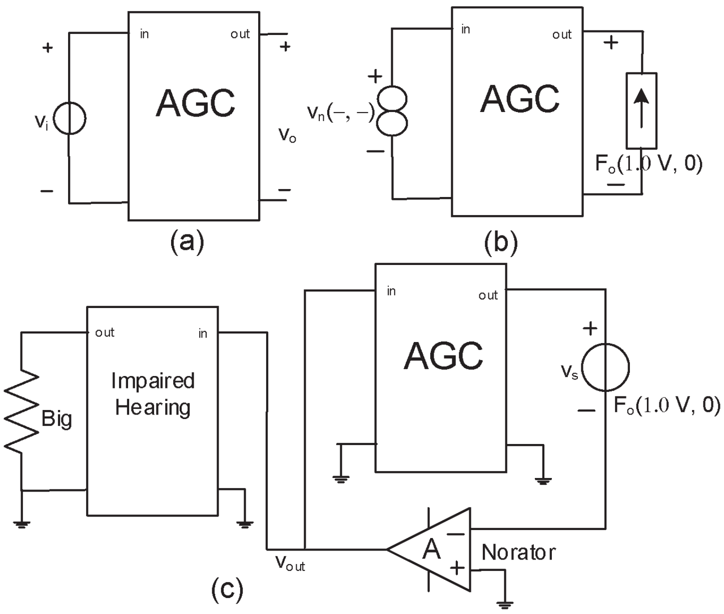

Figure 6.
(a) And (b) symbolic design of a hearing aid amplifier using an FNP; (c) a simplified equivalent circuit for test purposes.

fixator $F_o(1.0, 0)$ is providing the desired output as we stated, and a pairing norator $V_n(-, -)$ is instead added to the input port to allow this to happen. Again, the difference here is that we are now looking for a constant amplitude output from the AGC and not the input. Next we may ask, what type of the input signal the norator must provide to the AGC (replaced for the impaired hearing) so that the output is well achieved, i.e., the comprehended voice is uniformly constant? For the solution, we refer to **Figure 6(c)**. As we can see here, the norator is replaced with an Op-Amp, and as a feedback. It provides the necessary signal to the AGC for a constant amplitude output. So, if we now assume that the AGC represents the impaired hearing situation then the output of the impaired hearing is also flat as we desired, actually representing the improved hearing status of the individual.

What we need to do next is to see how we can replace the norator with a real sub-circuit, which turns out to be an amplifier, and then try to design it. **Figure 7(a)** shows a reconstruction of the complete hearing aid system presented in **Figure 6 (c)**, except the impaired hearing block is removed. To complete this design all we need to do is to design the norator amplifier. As mentioned earlier, because of the high losses that we experience at high frequencies the amplifier must provide a gain of 60 dB or more to compensate for the impairment. In this study, we selected an amplifier that uses a TI - LM318 Op-Amp with a bandwidth of 15 MHz. This Op-Amp can provide a gain of 66 dB (2000 V/V) at 8 KHz, which is well above the required value for this case study. **Table 1** provides the Electrical Characteristics of the TI - LM318 Op-Amp.

Figure 7(b) shows the amplifier constructed using LM318 Op-Amp along with its symbolic representation. With the rated gain-bandwidth product given, this Op-Amp is a very well fit to our design, although its power (0.5 W) is on the high side. There are certainly other choices of Op-Amps that can replace LM318 (not discussed here). **Figure 7(c)** is the response from the amplifier. As we can expect, this is exactly the opposite of the AGC, RC_2 frequency characteristic, shown in **Figure 5(a)**.

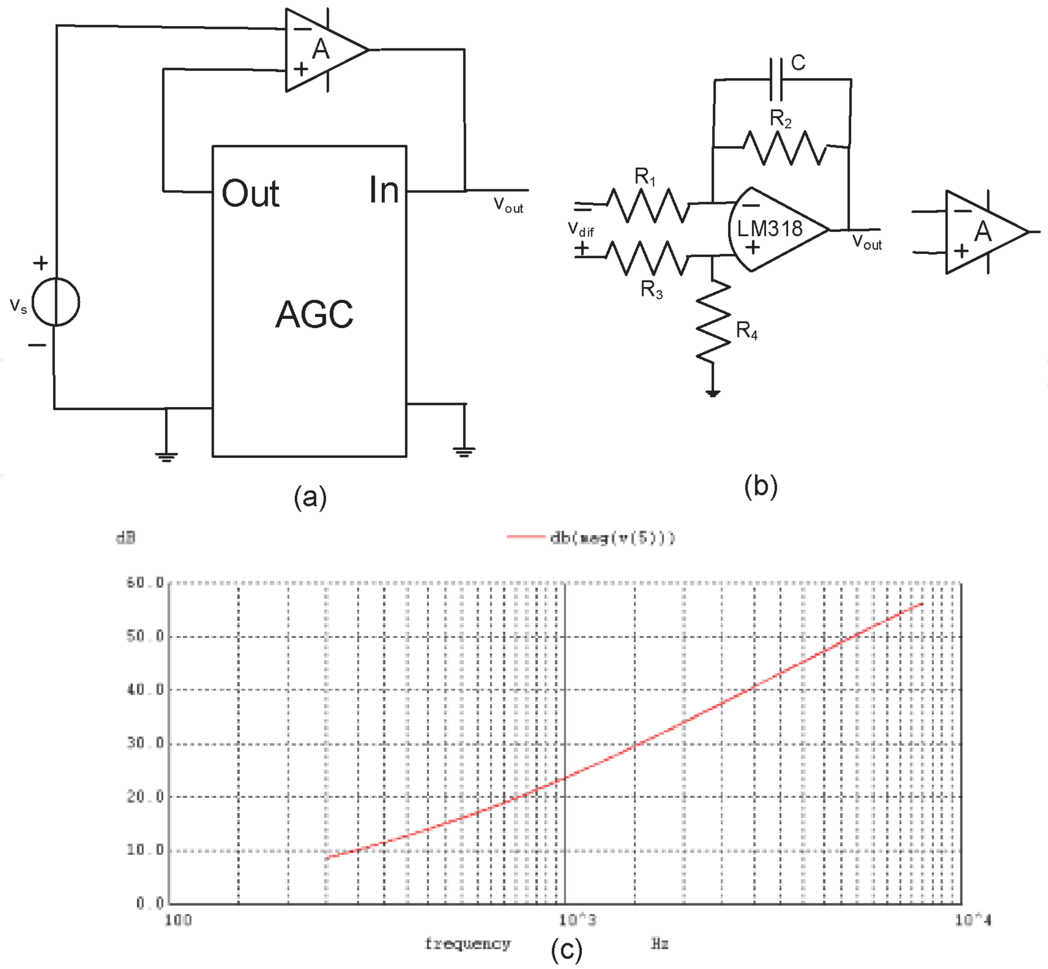

Figure 7.
(a) Hearing aid amplifier; (b) the construction of the amplifier using high gain Op-amp: (c) the amplifier frequency response.

We are now going through an examples to see how the technique practically works.

Example 1 – For this example we again take the case of the hearing-impaired patient with the audiogram given in **Figure 2**. We then construct the audio amplifier given in **Figure 7(a)** and **(b)**. However, there are some design considerations that needs to be addressed here. Our main challenge is to produce enough gain at higher frequencies (8 KHz) where the hearing loss is the most. By using TI - LM318 Op-Amp we get a small signal bandwidth of 15 MHz, which means, at 8 KHz frequency we can barely get 2000 V/V or 66 dB gain. Hence, this explains one of the reasons for selecting RC_2 instead of RC_1 for this design, which is to settle with lower gain requirement.

For testing purposes, we attach another AGC (audiogram) to the output port of the amplifier, resembling the hearing situation of the person with hearing impaired. **Figure 8** shows the combination of three parts: the input signal representing the voice received, the audio amplifier for voice processing, and the AGC model representing the hearing-impaired of the patient. As shown, the audio amplifier (hearing aid) receives the voice, amplifies it, and sends it to the patient's ear. The entire circuit is simulated and the results are plotted in **Figure 9(a)** and **(b)**, for magnitude and phase, respectively. In **Figure 9** we observe the frequency response of the amplifier that is exactly opposite of the frequency profile of the audiogram, represented by the AGC. The final result is a hearing profile which is flat for the entire frequency range.

Electrical Characteristics (1)								
Parameter	**Conditions**	**LM118-N/ LM218-N**			**LM318-N**			**Units**
		Min	**Typ**	**Max**	**Min**	**Typ**	**Max**	
Input Offset Voltage	T_A = 25°C		2	4		4	10	mV
Input Offset Current	T_A = 25°C		6	50		30	200	nA
Input Bias Current	T_A = 25°C		120	250		150	500	nA
Input Resistance	T_A = 25°C	1	3		0.5	3		MΩ
Supply Current	T_A = 25°C		5	8		5	10	mA
Large Signal Voltage Gain	T_A = 25°C, V_S = ±15 V	50	200		25	200		V/mV
	V_{OUT} = ±10 V, $R_L \geq 2$ kΩ							
Slew Rate	T_A = 25°C, V_S = ±15 V, $A_V = 1$ (2)	50	70		50	70		V/μs
Small Signal Bandwidth	T_A = 25°C, V_S = ±15 V		15			15		MHz
Input Offset Voltage				6			15	mV
Input Offset Current				100			300	nA

(1) *These specifications apply for ±5 V ≤ V_S ≤ ±20 V and − 55°C ≤ T_A ≤ +125°C (Im118-n), −25°C ≤ T_A ≤ +85°C (LM218-N), and 0 °C ≤ T_A ≤ +70°C (LM318-N). Also, power supplies must be bypassed with 0.1 μF disc capacitors.*
(2) *Slew rate is tested with V_S = ±15 V. The Im118-n is in a unity-gain non-inverting configuration. V_{IN} is stepped from −7.5 V to +7.5 V and vice versa. The slew rates between −5.0 V and + 5.0 V and vice versa are tested and specified to exceed 50 V/μs.*

Table 1.
Electrical characteristics of the TI - LM318 Op-amp used.

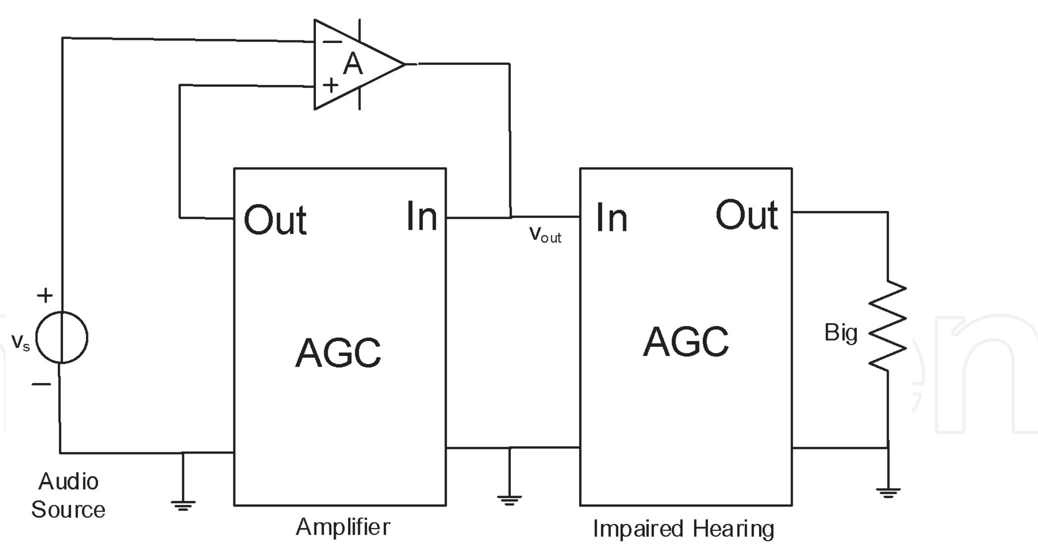

Figure 8.
A testing bench; testing the hearing aid amplifier.

Before constructing the system for laboratory testing, the circuit is simulated using WinSpice, and the following is the main portion of the code used for the simulation.

```
.control.
destroy all.
op
set units = degrees.
ac dec 1000 250 8 k.
```

```
plot ph(v(5)) ph(v(16)) ph(v(9)).
plot db(v(5)) db(v(16)) db(v(9))
.endc.
* Supplies and signal sources ********************************.
VCC     10      0       DC      5
VEE     0       20      DC      5
vi      1       0       DC      0       AC      90 m
* Combined Hearing Aid System ******************************.
rc      c       1       50
x3      4       c       10      20      5       Amp2
x4      5       7       4       AGC3
x8      5       15      16      AGC3
r3      16      0       10Meg
* AGC, Defected hearing profile ******************************.
re      e       1       50
x5      e       11      9       AGC 3
r6      9       0       10Meg
* Audiogram Generated Circuit *******************************
.subckt     AGC3    1       2       3
r0      1       2       1 k
c1      2       0       300n
r1      2       3       2.5 k
c2      3       0       400n
.ends.
* Amplifier for high gain, Gain = 5 k V/V, 74 dB **********************
.subckt     Amp2    1       2       10      20      5
x1      1       2       10      20      3       Amp1
r1      0       4       1 k
r2      4       5       100 k
x2      3       4       10      20      5       LM318
.ends.
* Amplifier using LM318 Op-Amp, Gain = 50 V/V, 34 dB ***************
.subckt     Amp1    1       2       10      20      5
x1      3       4       10      20      5       LM318
r1      2       4       1 k
r2      4       5       50 k
r3      1       3       1 k
r4      3       0       50 k
c1      4       5       0.3p
.ends.
* LM318 Op-Amp model parameters *****************************
.include op-models.txt
.end.
```

Following the simulation, the hearing aid circuit is constructed and the tested in a laboratory setup. **Figure 10** shows the experimental bread board for testing purposes, and **Figures 11–13** are the test results at different frequencies.

The output responses of the amplifier are shown at 250 Hz, 1.0 KHz, and 4.0 KHz frequencies. Notice, that not only the magnitude changes and increases for higher frequencies, but phase delay also increases up to 82 degrees. Finally, **Table 2** shows the experimental results for the gain vs. frequencies for the amplifier.

Example 2 – Now we are going to try a different audiogram in this example, one from a person with a rather mild hearing impairment. This audiogram is given in

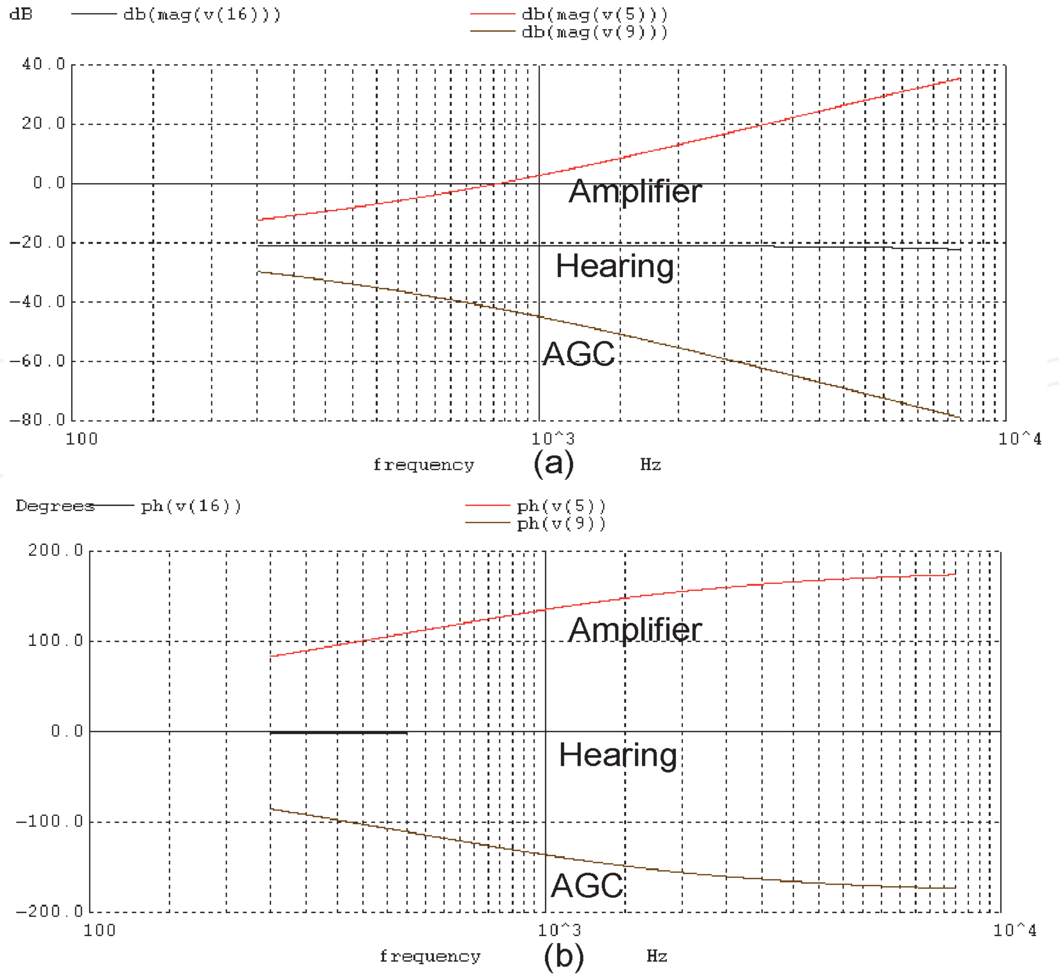

Figure 9.
*Simulation results: (a) magnitude plots from the test bench in **Figure 8**, including the amplifier response, audiogram, and the hearing profile by the hearing-impaired patient; (b) the phase responses.*

Figure 14. We start constructing an AGC model for this case, which is much similar to the one we did for Example 1. The circuit is constructed from R and C components and is then simulated for its frequency responses. **Figure 15** shown the magnitude Bode plot of the AGC. In addition, the audiogram is also added to the figure for comparison.

Our next step in the process is to construct the amplifier needed. Again, because of the modularity property the design procedure of the AGC is quite simple. All we need to do is to take the same amplifier constructed for Example 1 (**Figure 7(a)**) and replace its AGC, given at **Figure 3(b)**, with the new one created, for this example. For testing purposes, we again put all three units (the input signal representing the voice received, the audio amplifier with the new AGC, and a second AGC representing the hearing-impaired patient) together and simulate. The setup will be similar to the testing bench provided for Example 1 and shown in **Figure 10**. We then simulate the combined circuits again and plot the frequency responses. The responses from the amplifier and the one from the hearing profile, comprehended by the hearing-impaired patient, are given in **Figure 16**. Again, notice that the hearing has improved substantially by using the amplifier. As seen, the comprehended voice is quite flat just like the one we had in Example 1. Also notice that the ultimate phase angle has become flat, as well.

Further, in comparing plots in **Figure 16** with those in **Figure 9(a)**, we notice that the two amplifiers respond differently but the net results, i.e., the comprehended

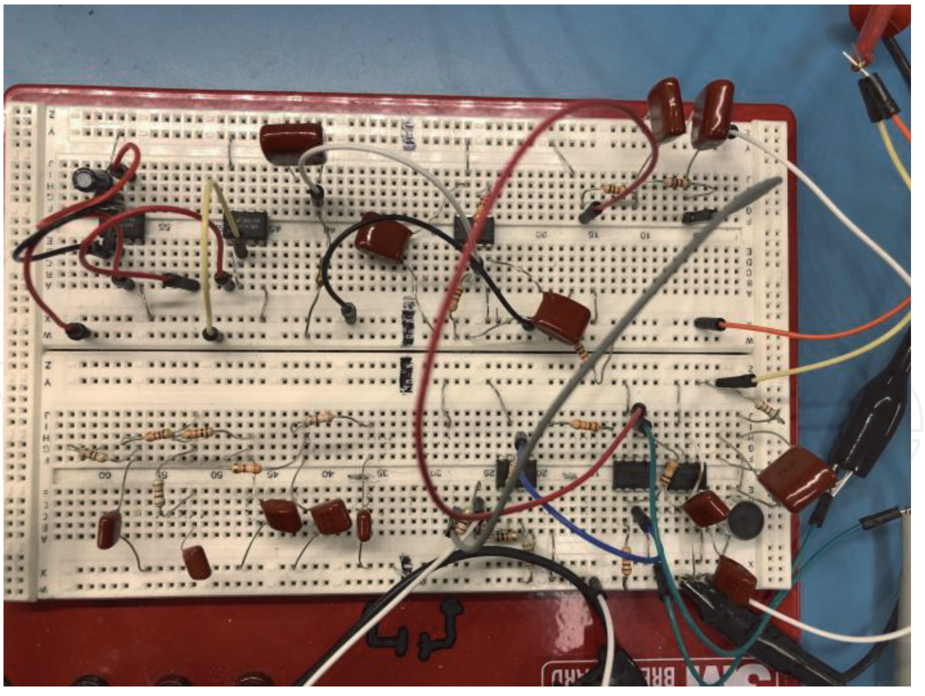

Figure 10.
A testing bench; experimenting the hearing aid amplifier.

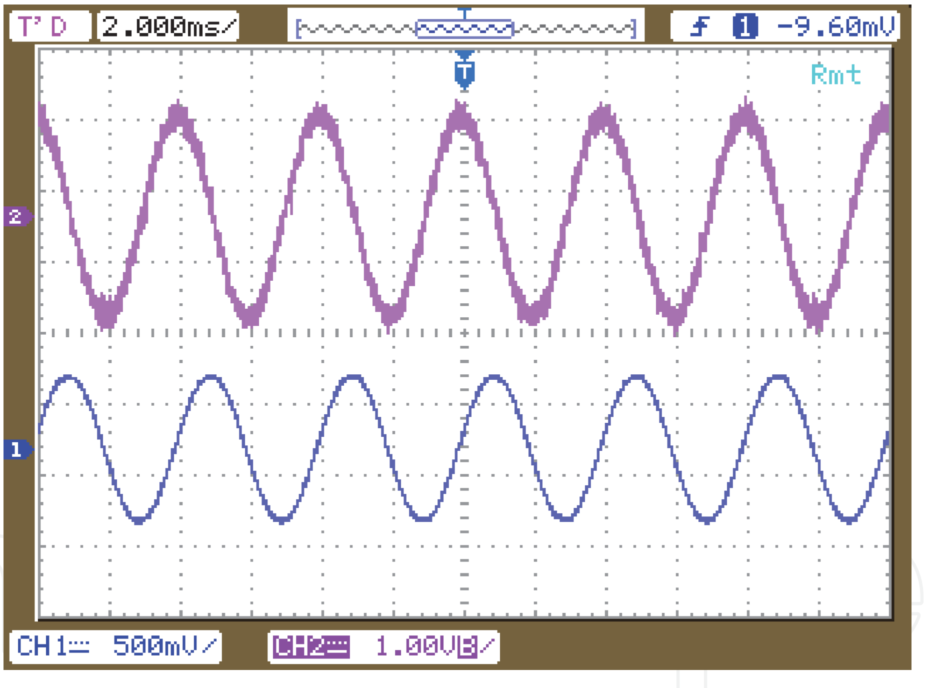

Figure 11.
Input signal (lower) and the amplifier response (upper) for 250 Hz. Note the scale difference.

voices are the same and completely flat. This shows the adaptability property of the amplifier. That is, as we discussed earlier, in shifting from one example (patient) to another all we need to do is to design a new AGC, while the amplifier unit remains unchanged, unless the ultimate gain of the amplifier in not sufficient to compensate for all the losses, recorded in the audiogram.

This brings us to the following algorithm for the construction of an adaptable amplifier for hearing aids.

Algorithm 1.

Given an audiogram similar to the one shown in **Figure 2** or **Figure 14**, we can construct an adaptable front-end or stand-alone analog amplifier that can be used to totally remove the hearing deficiencies and provide a convenient hearing. The procedure is as follows:

1. Construct a passive AGC that represents the audiogram profile of a hearing-impaired patient as closely as possible, like the ones shown in **Figures 4** and **15**.

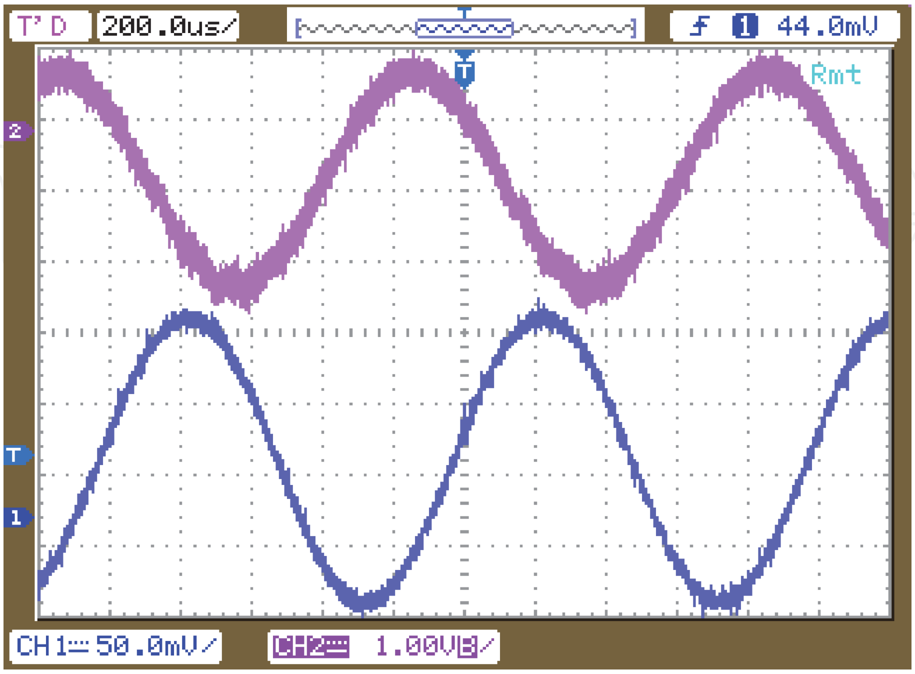

Figure 12.
Input signal (lower) and the amplifier response (upper) for 1 KHz. Note the scale difference.

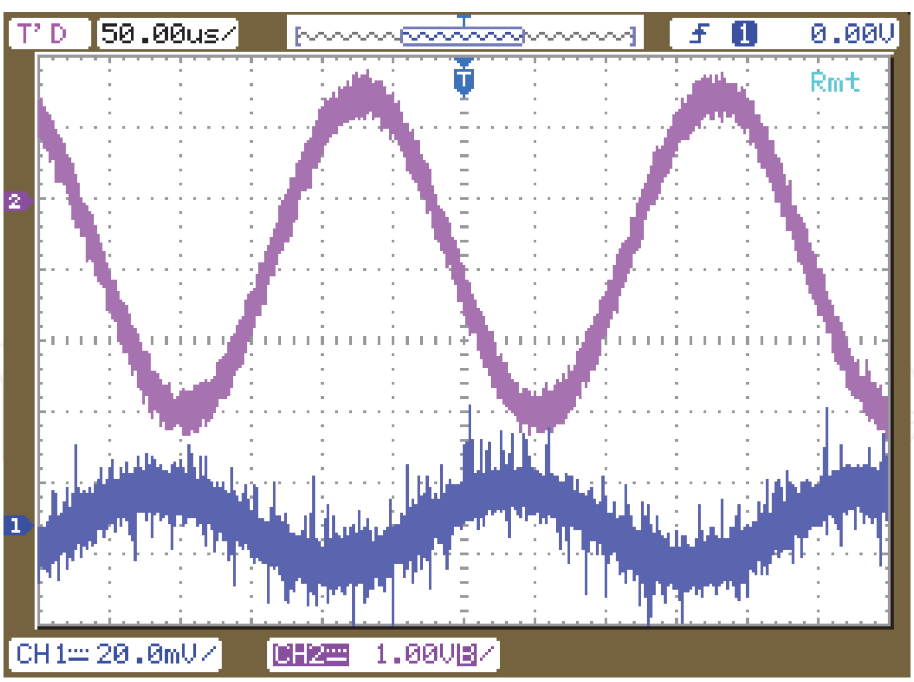

Figure 13.
Input signal (lower) and the amplifier response (upper) for 4 KHz. Note the scale difference.

Frequency Hz	250	300	700	1 K	1.7 K	2 K	3 K	4 K
Gain A_v V/V	3	4	13	20	39	65	98	200

Table 2.
Audio amplifier experimental results.

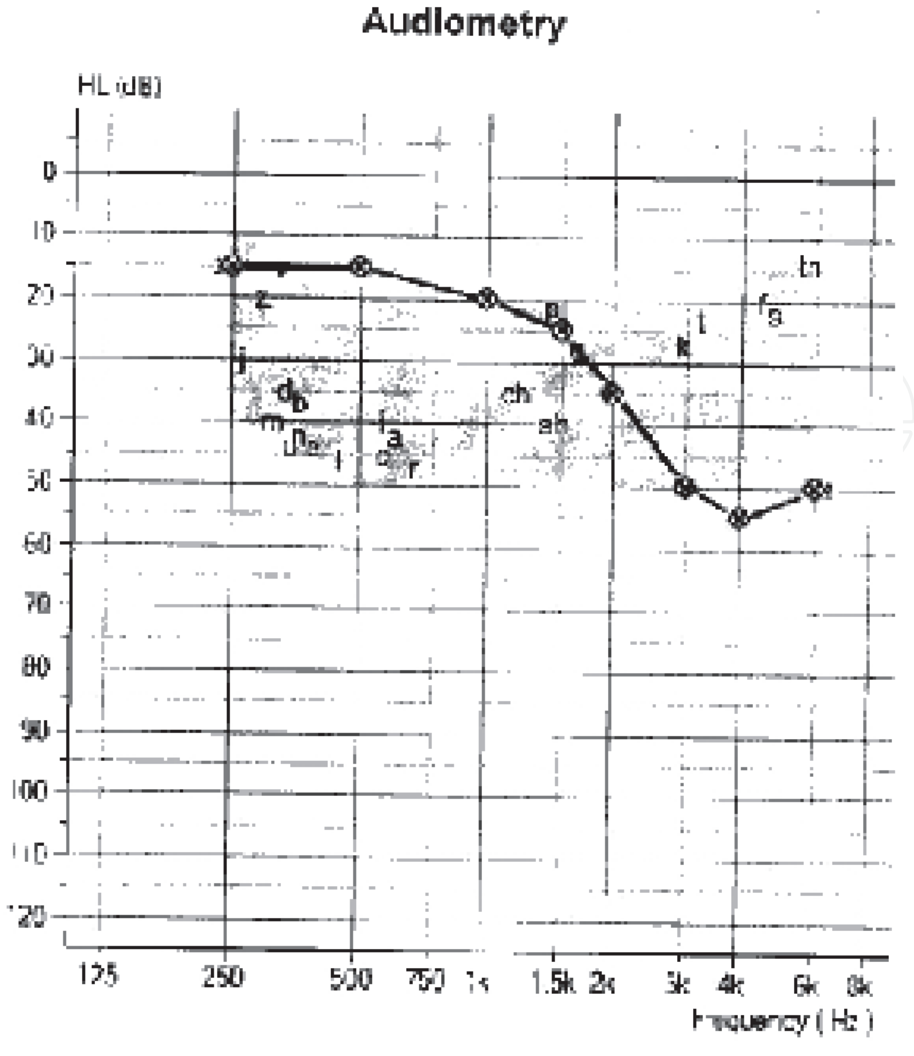

Figure 14.
Audiogram from a patient with mild hearing impairment.

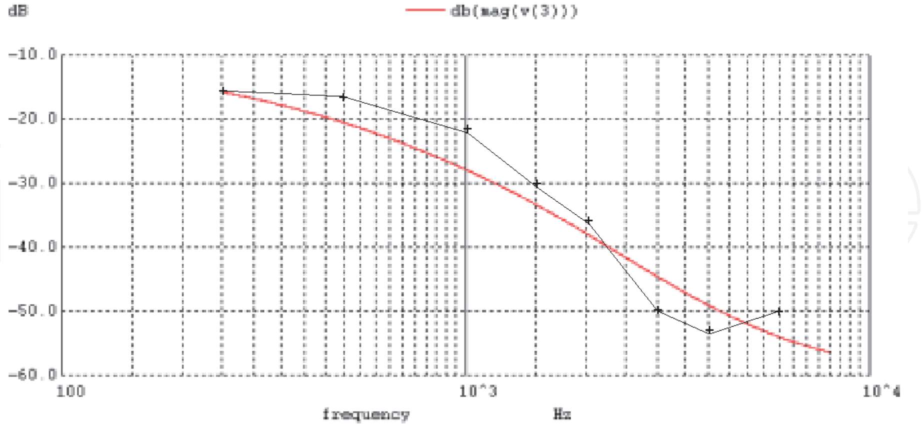

Figure 15.
The comparison between the audiogram and the frequency response of the adopted AGC.

2. Use the AGC and an amplifier with sufficient gain to construct an audio amplifier as discussed before and shown in **Figure 7(a)**.

3. The amplifier so constructed is adaptable, in a sense that for any other audiogram all we need to do is to replace the older AGC with a new one, constructed for a new patient.

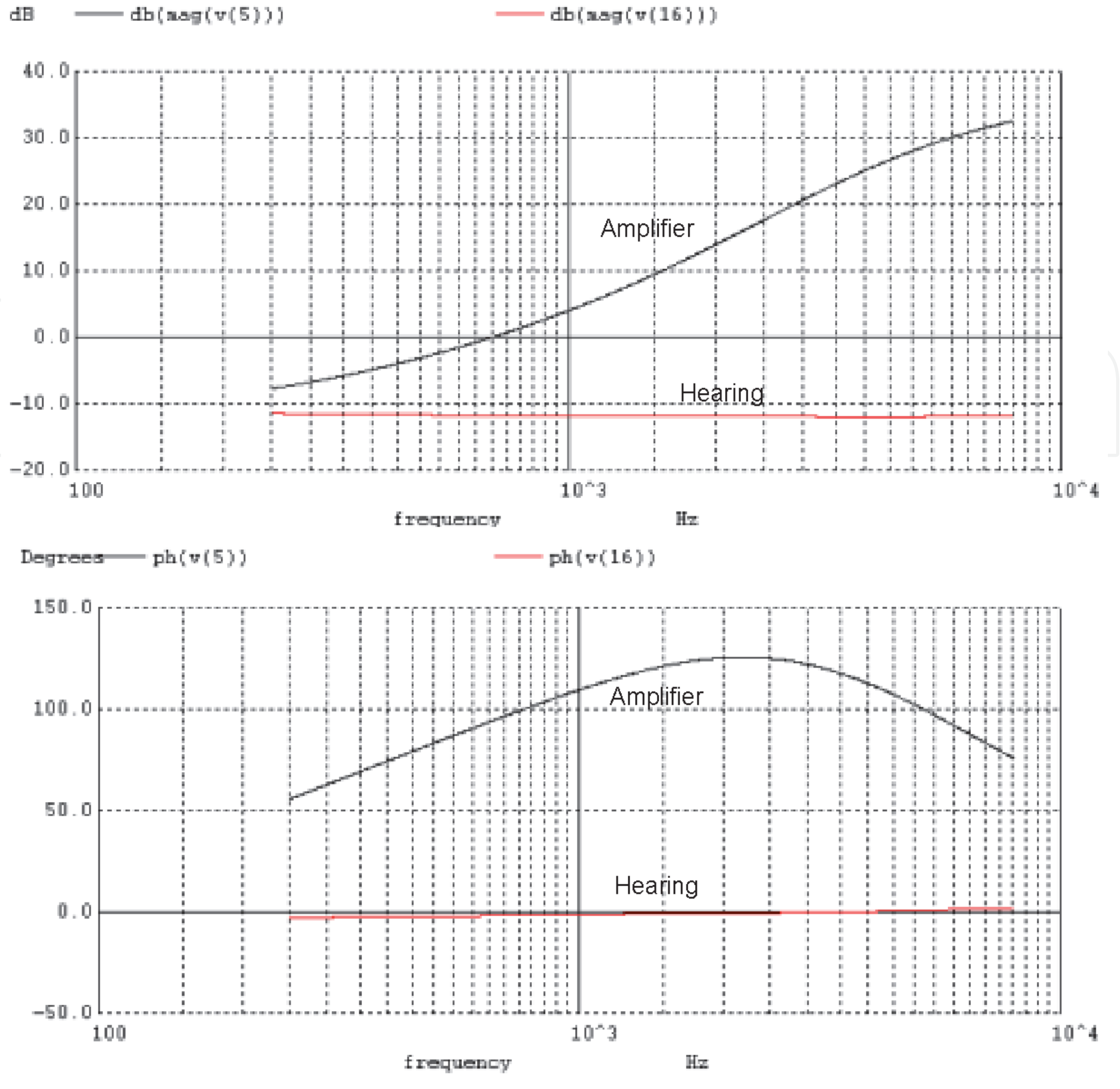

Figure 16.
Magnitude and phase responses from the amplifier and the comprehended hearing profile.

4. Simulate the entire system in a setup shown in **Figure 8**, for verification purposes.

This concludes our design procedure for hearing aids with flat responses, where $F(s) = 1$ in (1). What we need to do next is to extend the design to cover for the cases when $F(s)$ is not necessarily flat due to some preferences in the hearing quality, for example, for factory workers for whom it may be needed to reduce certain machinery noises or enhance those related to the onsite conversation.

6. Design with extra gain added

Up until now we have assumed that the comprehended voice by the patient needs to be flat, leading to the gain function of $F(s) = 1$. Now, we assume an arbitrary gain function $F(s)$ recommended for the hearing-impaired patient, suitable for his/her application.

This is done by splitting the design procedure into two parts. In the first part we again assume a flat response with $F(s) = 1$, and in the second part we modify the amplifier so constructed to generate the frequency response $T(s)$ for different $F(s)$. This modular design may also provide us with the options to switch between the two cases during the application. Knowing how to design for *F(s) = 1* by now, all we need to do here is to go for a desired arbitrary response, *F(s)* and add it to the system. Here is the problem statement.

Problem statement – Given an audiogram of a hearing-impaired patient, as shown in **Figures 2** or **14**, design a front-end or stand-alone amplifier that is fully adaptable and has a wide voice dynamic range covering from 250 Hz to 8 KHz, as specified in the audiogram. In addition, the amplifier is supposed to produce a comprehended voice frequency profile *F(s)* that is recommended for the patient.

Design procedure – To design such an amplifier we first follow the procedure explained in the previous section, i.e., design an amplifier *N* with the frequency response of $T(s) = H(s)^{-1}$. As we discussed, this produces a flat response with *F(s) = 1*. We then follow the method explained in [8]. This method uses nullors to modify the amplifier circuit until it produces a transfer function$T(s) = F(s)/H(s)$, where *F(s)* is the desired frequency response, which is produced by a *model circuit M*. Note that *M* does not need to be a physical circuit as long as it generates an output response *F(s)*.

So, the design starts by first assuming that we already have done the first part and constructed an amplifier *N* for *F(s) = 1,* which has $T(s) = H(s)^{-1}$ transfer function, as shown in **Figure 17(a)**. Next, let us assume we have been able to find a sub-circuit *P* (usually a feedback) so that by adding *P* to *N* the circuit can be realized to perform with a transfer function $T(s) = F(s)/H(s)$, for an arbitrary *F(s)*.

In summary, for a given transfer function *T(s,)* we first design the circuit *N* for its $T(s) = H(s)^{-1}$. We then add a sub-circuit *P* to *N* and modify *P* until we get $T(s) = F(s)/H(s)$. So, our main objective here is to find the sub-circuit *P*. This is stated in a stepwise procedure, Algorithm 2.

Algorithm 2

1. Consider an amplifier *N* already designed for a flat hearing, with *F(s) = 1*. Next, try to find a model circuit *M* that produces a desirable frequency response *F(s)* that is realizable. If circuit *M* is not physically available, try to artificially synthesize *M*, possibly through a cascade decomposition method. *Note* that, because the model circuit *M* may only be needed for simulation purposes, the use of any ideal components, such as ideal controlled sources, in *M* is permissible, which makes it easier to generate.

2. Find a location in the circuit *N* such that adding a two terminal sub-circuit *P* to *N* can bring a solution to the problem, as depicted in **Figure 17(a)**[1].

3. Connect the two circuits *N* and *M* together in parallel, and keep the two output currents at zero by adding a nullator between the two outputs, as shown in **Figure 17(b)**. To match the nullator, add a norator *P* to the designated location in *N*. This norator will be later replaced with a sub-circuit *P*.

Note 1: Practically, a norator can be a high gain controlled source or an Op Amp [2, 10].

Note 2: parallel connection of *N* and *M* in **Figure 17(b)** is only valid for voltage to voltage transfer functions. However, this is not the only option we have, and other configurations can also be adopted. For example, for a case of current sourcing the connections must be in series, as appropriate [8].

[1] Here we assume the connecting nodes of *P* to *N* are already specified. Going after the best location connecting *P* to *N* adds another dimension to the problem, which is outside of this chapter.

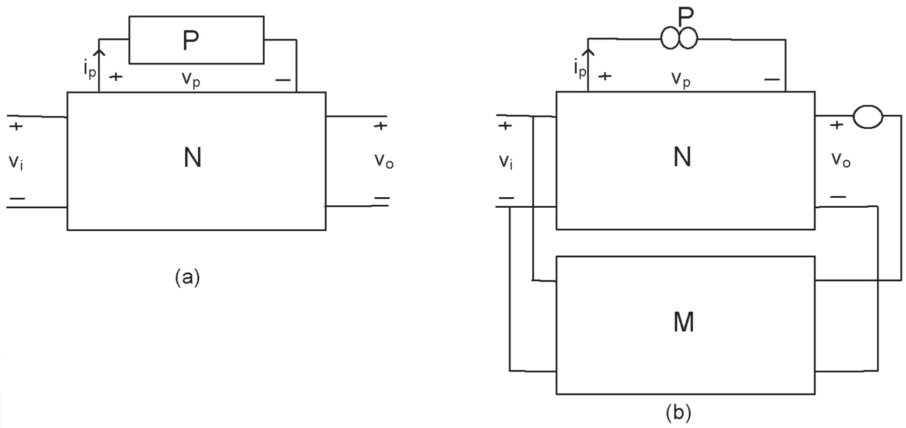

Figure 17.
(a) Circuit N *with a two-terminal* P *added; (b) realizing the two-terminal* P *by enforcing* N *to follow the desired response from* M.

4. Simulate the combined circuit. Evidently the frequency response of both circuits *M* and *N* must be identical, because of the parallel connections. So, *N* follows *M* in response. Because of the enforced response on the output of *N*, a virtual impedance function $Z_p(s) = V_p(s)/I_p(s)$ is created for the norator *P* through the simulation. This means, if we replace the norator with a two-terminal circuit that has the impedance $Z_p(s)$, then we get an independent response from *N* that is identical to that of *M*. Which means, *N* responds independently after being separated from *M*.

5. Now we need to synthesize *P* such that its impedance characteristic is close enough to $Z_p(s)$, for the specified bandwidth. Then we replace the norator *P* with the actual two-terminal *P* found. If $Z_p(s)$ is not realizable make proper approximations/adjustments to fit.

Note 3: The methodology works for nonlinear circuit as well, provided that the sub-circuit *P* does not disturb the biasing situation of *N*. This is very important point, which means that circuit *N* must be protected by coupling/bypass capacitors, if necessary.

The following example clearly demonstrates the steps given in Algorithm 2.

Example 3 - **Figure 18** shows the same amplifier designed for Example 1 and illustrated in **Figure 8** for a flat comprehended voice bandwidth ($F(s) = 1$), except here a *model circuit M* is added to the configuration. The model circuit is connected to the amplifier in parallel and through a nullator at the output port. A paring norator *P* is also added to the AGC in the amplifier. The norator is in fact a “place holder” for an actual two terminal sub-circuit *P* that must be found to replace it. According to Algorithm 2, and for simulation purposes, we now need to replace the nullor with a high gain dependent source, and for this example we have selected a CCVS, with the SPICE code given as:

```
va    2     12    DC    0
vb    11    13    DC    0
h1    13    0     va    1.0e6
```

The choice of CCVS is not unique, and in fact any of the four types of controlled sources can be selected, depending on the situation. For the present case, we first decide which variable (i or v) in the nullator is going to control the norator. We notice that the current in the nullator, although presently zero, is an effective

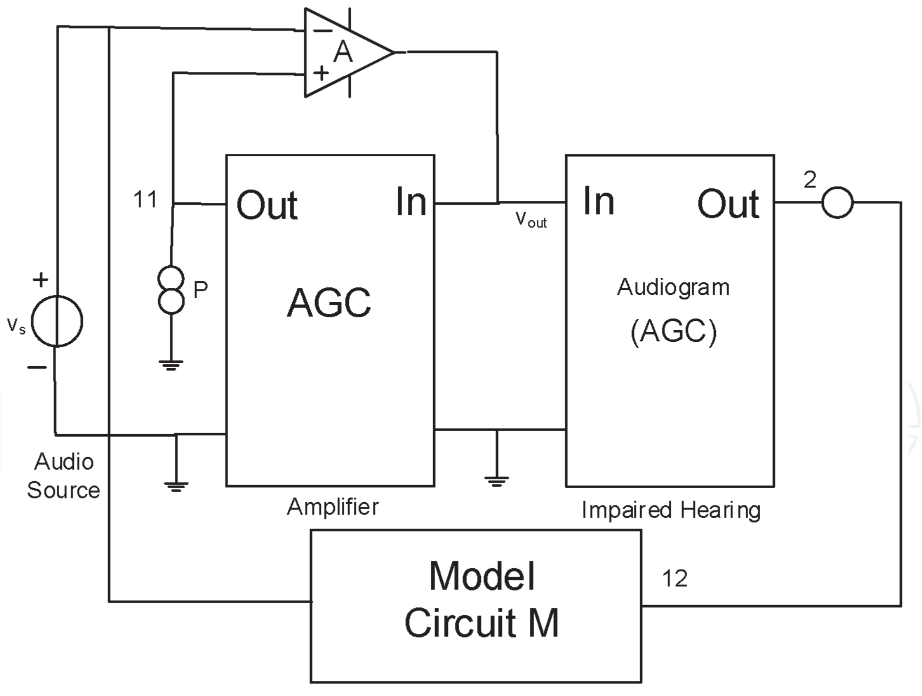

Figure 18.
Design procedure to modify the audio amplifier for a response given by the model circuit M (see ***Figure 17****).*

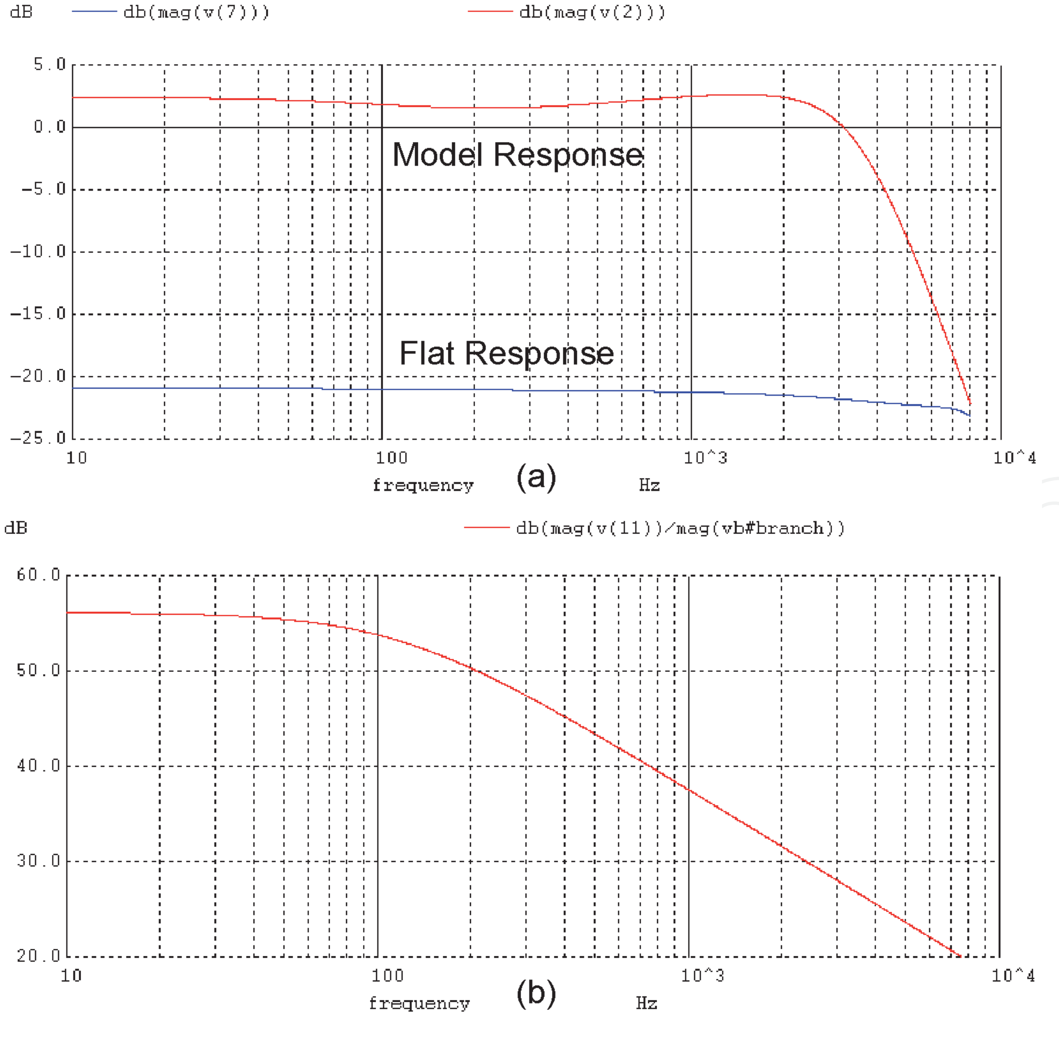

Figure 19.
(a) Comparing the frequency response of the model circuit M with the amplifier before being modified; (b) the frequency plot representing the virtual impedance of the norator in ***Figure 18****.*

candidate to control the norator. So either CCVS or CCCS can be selected. For more practical reasons, here we chose CCVS that closely resembles an ordinary Op-Amp.

The combined circuit also includes a patient's audiogram (AGC) representing his/her hearing situations. The entire circuit is then simulated and the results are plotted in **Figure 19(a)**. The SPICE code for the plots are:

plot db(v(2)) db(v(7)), the magnitude responses, **Figure 19(a)**
plot db(v(11)/I(vb)), the magnitude response of $z_p(s)$, **Figure 19(b)**

There are two response plots in **Figure 19**, one (*v(2)*) from the model circuit, and another one (*v(7)*) representing the original amplifier with the flat ($F(s) = 1$) response. In addition, **Figure 19(b)** shows the frequency response of the virtual impedance associated with the norator, i.e., $V_p(s)/I_p(s)$. What we need to do now is to find a two terminal sub-circuit *P* with the impedance function $z_p(s) = V_p(s)/I_p(s)$ and then substitute it for the norator.

In search for a realizable sub-circuit *P* we first make the assumption that *P* must be a passive RC circuit. This is realistic because *P* is going to be part of the AGC, which is already designed with passive (R and C) components. In our search we simply notice that the norator characteristic curve (**Figure 19(b)**) appears to be a

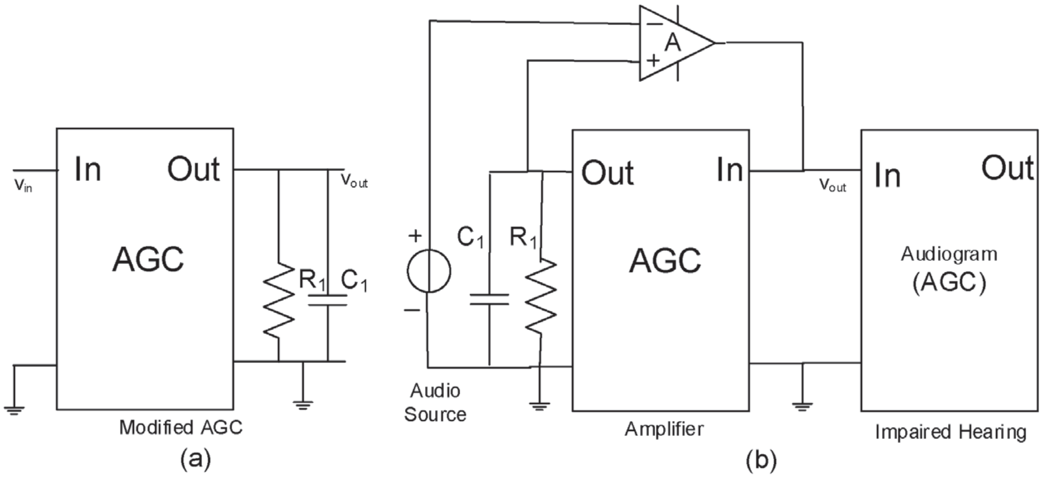

Figure 20.
Testing the modified hearing aid amplifier for the final results pertain to a hearing-impaired patient.

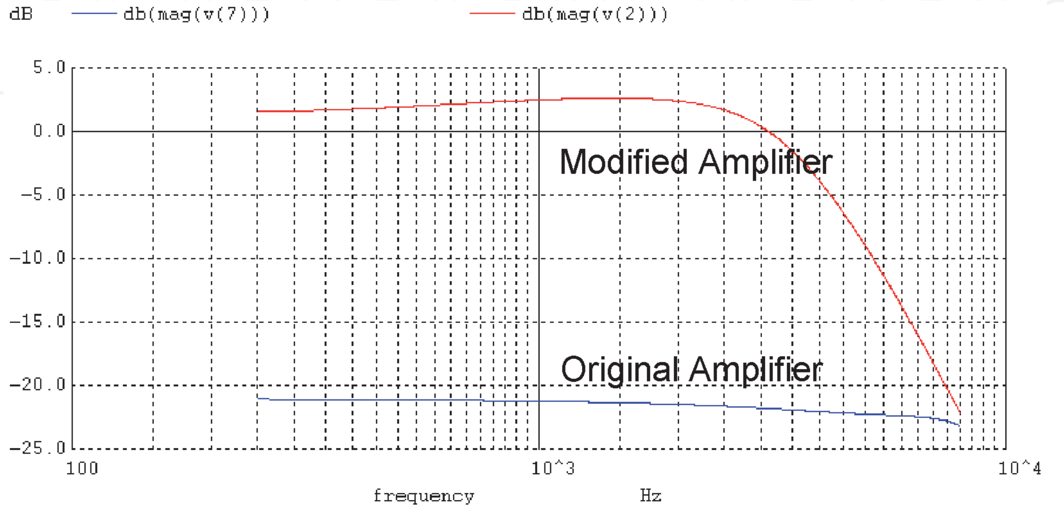

Figure 21.
Comparing the frequency response of the modified amplifier circuit with the original amplifier before being modified.

low pass filter realizable by a parallel RC circuit. The plot approaches 56 dB (631) at low frequencies and it falls at higher frequencies with break point frequency at $f_p = 118\ Hz$. Next, to find the RC circuit, we first find $R_1 = 631\ \Omega$, and then from f_p we get the equivalent capacitor $C_1 = (2\pi.f_p.R_1)^{-1}$, or $C_1 = 2.1\ \mu F$. So, now the two-terminal norator P can be replaced with the components R_1 and C_1 in cascade, as shown in **Figure 20(a)**. The next step is to replace the older AGC in **Figure 18** with this modifies AGC, and then remove the model circuit all together.

Finally, **Figure 20(b)** shows a testing setup for the new hearing amplifier. After the simulation we plot the frequency response of the entire system as "Modified Amplifier", plotted in **Figure 21**. The Bode plot actually represents the voice heard and comprehended by the hearing-impaired patient. Also for comparison purposes, the result of a similar testing setup for the original amplifier with flat response ($F(s) = 1$) is also shown in **Figure 21**. A point to notice here is that, the new R_1C_1 circuit although passive, it contributes to a higher gain in amplifier and helps for an enhanced hearing. However, the price we need to pay is to assume higher gain for the Op-Amp. For example, although a gain of 60 dB is sufficient for the hearing aid with a flat response, $F(s) = 1$, it is not enough for this kind of enhanced situation. With about 20 dB gain desired here, we need to add the same amount to the Op-Amp and make it for 80 dB gain. So, here we see that the gain of 66 dB we have adopted for the Op-Amp is not sufficient any more, or we may lose some gain and precision losses at high frequencies, as we notice it in **Figure 21**.

7. Feedback implementation of analog hearing aids

As mentioned before, there is an alternative implementation technique to design hearing-aid amplifiers. This technique uses the well-known negative feedback methodology, which consists of a high forward gain amplifier inverting a transfer function generated by a feedback circuit. Consider a high gain amplifier $A(s)$ in the forward path of a feedback system, and an AGC, with the transfer function $H(s)$, placed in the feedback. The system transfer function $T(s)$ then becomes:

$$T(s) = \frac{A(s)}{1 + A(s)H(s)} \tag{5}$$

and for a special condition

$$A(s)H(s) > > 1 \tag{6}$$

$$T(s) \cong H(s)^{-1} \tag{7}$$

Eq. (7) is similar to Eq. (3) except for the constraint given in Eq. (6). This shows that the two methodologies, the feedback and the FNP, have elements in common, but different in implementation. We will discuss the major differences between the two later in this section However, let us analyze the feedback system first.

Feedback Implementation – **Figure 22** shows a feedback implementation of the hearing aid system testing setup. In the amplifier circuit part, the Op-Amp A_1 is constructed similar to what we did for FNP method in **Figure 7(b)**. For a better comparison between the feedback method and one using the FNP based design, we try to use identical components such as the AGC and the Op-Amp circuit. We then simulate the testing setup (**Figure 22**). The result of the simulation is shown in **Figure 23**. We can now compare these results with those obtained for the FNP case given in **Figure 9**. Although they look similar in general but they are different in

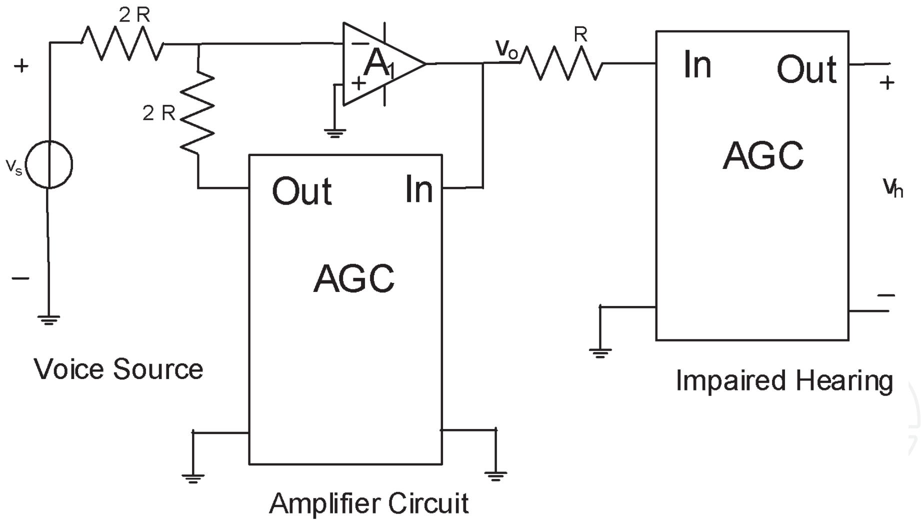

Figure 22.
A test setup for the hearing aid amplifier using normal negative feedback.

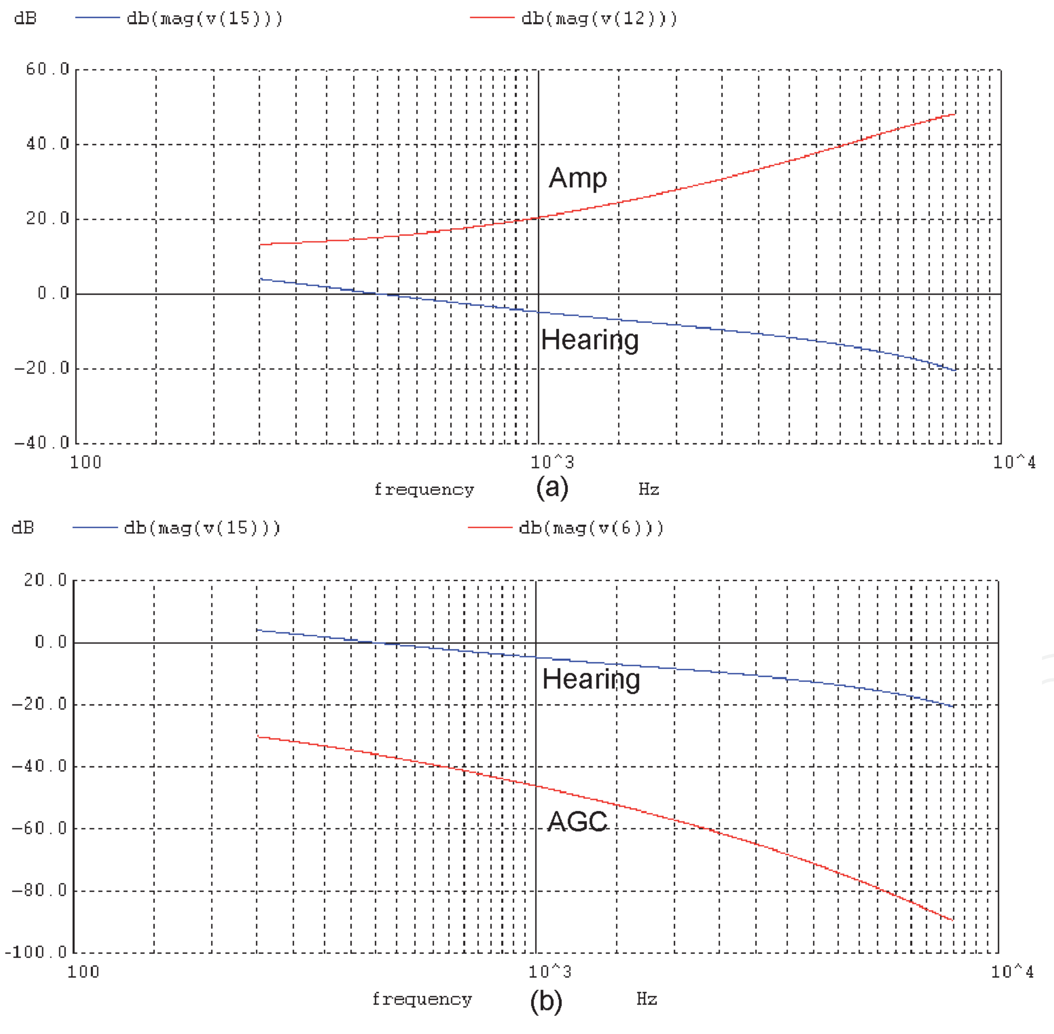

Figure 23.
Simulation results from the testing bench in ***Figure 22****; (a) the amplifier response and hearing profile by the hearing-impaired patient; (b) the frequency response of the AGC.*

performance. Notice that within the three plots (AGC, Hearing, and Amplifier) in each case the two AGCs are obviously the same, but the major difference is the following. In FNP implementation the frequency profile of the amplifiers is almost opposite of that of the AGC, whereas in the feedback case this is not true. In the

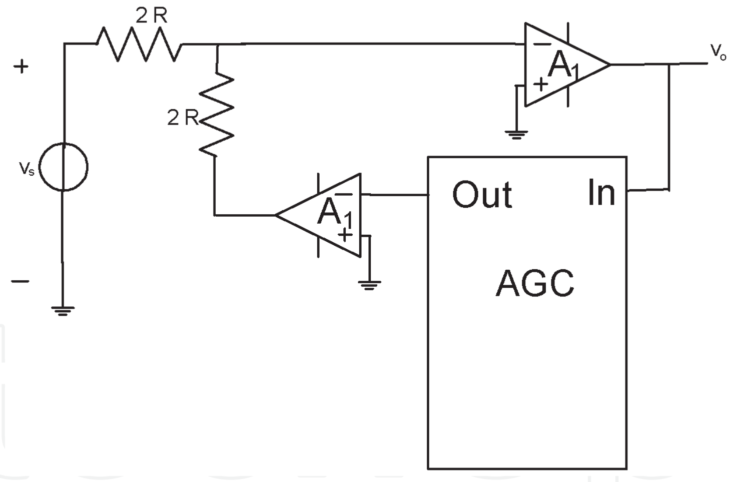

Figure 24.
Hearing aid amplifier using normal negative feedback. Another amplifier is added to prevent the AGC from being loaded.

feedback we are losing the gain for high frequencies, and that is why we are not getting a flat response at the end, as we do in the FNP case (Hearing in **Figure 9**). The consequence is that we are loosing the voice quality at high frequencies.

This, as it turns out, is due to the lack of impedance matching in the feedback case. For a passive circuit like AGC, the 2R resistive loading creates a clear variation in the AGC response because of loading. However, this is not the case for FNP methodology. Let us look at **Figure 6(b)**. Because a fixator with zero current is connected to the output port the AGC is never loaded. So it stays unchanged no matter what happens to the rest of the circuit. Back to the feedback case, one way to correct the loading problem is to use a buffer stage at the output port of the AGC circuit. This is demonstrated in **Figure 24**, where an extra amplifier is added to the output of the AGC. This of course fulfills the impedance matching and prevents the AGC being loaded. Afterwards, if we simulate the circuit we see the improvement, and the response received is almost the same as given in **Figure 8** for FNP realization.

Finally, we need to mention another major difference between the feedback method and the FNP technique. Let us revisit Eq. (7) and compare it with Eq. (3). To have the two equations identical we need to have $A(s)H(s) > > 1$, as stated in Eq. (6). This is basically a serious constraint for designing hearing aids when the corresponding audiogram (AGC) displays a large loss in higher frequencies. To make it clear, let us assume that we get satisfied with 1% accuracy for the system when we use the feedback method. This means $|A(s)H(s)| \geq 100$, for all the bandwidth. This means we always need to have the gain of the amplifier 40 dB higher than the largest inverse loss in the feedback. For example, for an AGC with 60 dB loss at high frequency we need an amplifier gain of 100 dB, instead of 60 dB, to satisfy the job, and this makes the system harder to design. The good news, however, is that if we intend to use the setup shown in **Figure 24** for impedance matching purposes then we can split the high gain requirement between the two amplifiers in the feedback loop and make it more distributed amplifier, Of course there are still some consequences involved but we ignore them here for simplicity. In any case, this shows the superiority of the FNP method compared to the feedback theory.

This concludes our design alternative using negative feedback technique.

8. Some basic comparisons with digital technology

As mentioned in Introduction, with digital high-tech so advanced the digital hearing aid dominates the market as well as the research and development areas.

However, in addition to its simplicity and cost effectiveness the analog hearing aid technique may still offer some advantages in certain areas. In particular, in comparing the two systems there are some operational factors to consider, and here are a couple of these factors. In digital technique, the incoming voice needs to go through ADC (analog-to-digital conversion), and after being processed the output signal reenters into an inverse process of DAC. This definitely adds to the *path delay* of the signal as well as reducing the *precision accuracy* of the signal due to the double data conversion. In case of analog methodology, however, we have neither of them. The signal stays analog all the way through, and the delay is just equal to the analog propagation delay of the signal through a limited number of devices.

9. Conclusion

A methodology is developed and explained in this chapter that uses nullors and FNPs to design amplifiers for certain frequency profiles. The method is applied for designing audio amplifiers for front-end or stand-alone hearing aids.

The method works as follows: For a given hearing profile (audiogram), a circuit model, called Audiogram Generator Circuit (AGC), is initially constructed. This AGC has the same, or close to, frequency response of the audiogram. Now, because of typical losses in hearing, it is shown that this AGC is all passive for a hearing-impaired patient, and it can be made quite simple and modular. Next, an FNP is used to construct an adaptable amplifier inversely following the AGC frequency spectrum. This amplifier converts the frequency profile (transfer function) of the AGC such that the poles and zeros of the AGC are exchanged and become the zeros and poles of the amplifier circuit. So, when the amplifier is used as a hearing aid, the result will be a flat frequency response for the comprehended voice heard by the patient. A second design method is also introduced in the chapter, which use the negative feedback theory. Although not a powerful as the FNP method, the feedback technique is modified for impedance matching.

The FNP design is further extended to cover the cases in which extra gain, and in general some added frequency profile is needed. This feature may help to enhance signals in certain frequencies for clear understanding, or conversely, cancel some unwanted sounds and noises. Here, it is shown how the original amplifier can be modified by adding some sub-circuits to the original AGC without touching any other part in the amplifier (such as Op-Amp circuits). Three examples of actual cases of hearing-impaired patients are worked out.

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