HEARING AID. MEASUREMENT OF ELECTROACOUSTICAL CHARACTERISTICS

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ABSTRACT

In this work we show the results obtained from several measurements, on hearing aid (HA) prototype carried out at the Acoustic Laboratory of the Centro de Ciencias Aplicadas y Desarrollo Tecnológico (CCADET) at the Universidad Nacional Autónoma de México (UNAM). The hearing aid has been developed at the Electronics Laboratory of the same Center. All the electroacoustic measurements have been made according to the recommendations established in the ANSI and IEC Standards for hearing aids.

RESUMEN

En este trabajo, mostramos los resultados de varias mediciones para caracterizar un prototipo de un auxiliar (prótesis) auditivo. El prototipo fue desarrollado en el laboratorio de electrónica del Centro de Ciencias Aplicadas y Desarrollo Tecnológico (CCADET) en la Universidad Nacional Autónoma de México (UNAM). Todas las medidas electroacústicas se han hecho según las recomendaciones establecidas en las normas ANSI y IEC para los auxiliares auditivos.

KEYWORDS: Hearing Aids, Hearing Impairament, Hearing Assesment, Sensory Aids

1. INTRODUCTION

Hearing impairment is a problem very frequently presented in a large part of the population of our country. Even though different types of hearing aids exist in the market, its acquisition and maintenance turn out to be extremely expensive due to the fact that these systems require special batteries and the integrating components are hard to obtain in our country. These problems motivated us to develop a low-cost hearing aids of easy maintenance and high quality that permit us to solve part of this problem. To achieve this, we have based our design on low-cost devices that are easily acquired in the national market; furthermore, we have taken into account that the system should operate with commercial batteries, thus, we guarantee an easy maintenance. However, it is necessary that the system will be characterized in its electroacoustic components as well as in its full on gain behavior based on specialized standards for this type of devices. Therefore, in the present work the various evaluation results made to our prototype attached to the recommendations established in the corresponding standards are shown; in some instances, novel procedures were used that are not reported yet in the specialized literature.

2. MEASUREMENTS

The electroacoustic components of hearing aid (HA) prototype are a microphone and a earphone or receiver. The measurements initially were oriented to characterize the HA, but during the development process the need of accomplishing additional measurements emerged in order to improve the design principally on the selection of the microphone and the earphone. The measurements were planned in the following manner: Microphones measurement (without HA), Earphones measurement (without HA), and full HA measurement: microphone + HA + Earphone.

To evaluate the automatic gain control circuit (AGC) of the HA, we followed the recommendations specified in Standard IEC 118-2. These specifications demand to accomplish the evaluation of the AGC and determine the input/output sound pressure level; therefore it is necessary to develop an experimental set up similar to the one used in the electroacoustic measurements. For this reason, the evaluation of the AGC is considered as part of the electroacoustic measurements. The electroacoustic measurements consisted of the following steps:

1. Microphones measurement (without HA).
2. Earphones measurement (without HA).
3. AGC Measurement.
4. Full HA measurement: microphone + HA + earphone.

2.1 Microphones measurement (without HA)

To accomplish this measurement several methods can be employed, but we have used the impulsive method or impulse response. This method is based on the fact that the frequency response of the microphone is obtained through the Fourier transform of the impulse response

$$H(\omega) = \int_{-\infty}^{\infty} h(t) e^{-j\omega t} dt \qquad (1)$$

where h(t) is the impulse response.

In this method, it is necessary to generate a sufficiently narrow impulse to cover the broad-band in frequency of the microphone (0 to 20 000 Hz). The generation of this pulse by electronic means is relatively easy, but if loudspeakers are employed to produce the acoustic pulse is quite difficult to achieve large pressure values [1]. Therefore, other types of sources are employed, such as release of high voltage or air balls [2]. Other form of generating acoustic impulses that recently we have been using is by means of a plasma generated by a pulsed laser. The plasma is generated by focusing the beam pulsed laser through a lens at a point of the space (air at ambient temperature), something which provokes that the temperature increases instantly and extremely in that small region due to the sudden thermal expansion; at the same time, it generates a shock wave of a very high sound pressure level. It should be taken into account that the focal distance of the lens should not be greater than 5 m, since this provokes that the generation of the plasma becomes unstable [3]. We use a pulsed Nd:YAG laser (1.06 μm of wavelength, 5 ns pulse width) of Continnum, model Surelite II to generate the plasma.

The experimental setup employed consisted of: a simple optical arrangement (beam splitter and lens of 5 cm focal distance), to concentrate the energy, a measurement amplifier B&K 2636, a photodetector of our own construction, and a digital oscilloscope Tektronix model TDS 254[a]. This setup is shown in figure 1. In previous measurements the broad band of this impulsive signal was determined, and it was greater than 50 000 Hz, sufficient for the characterization of the microphones to be employed in the HA. The pressure level obtained to 10 cm, was about 120 dB. This level can be modified by varying the laser energy or varying the separation between the source and the microphone. We used a level greater than 90 dB and we did not use an anechoic chamber.

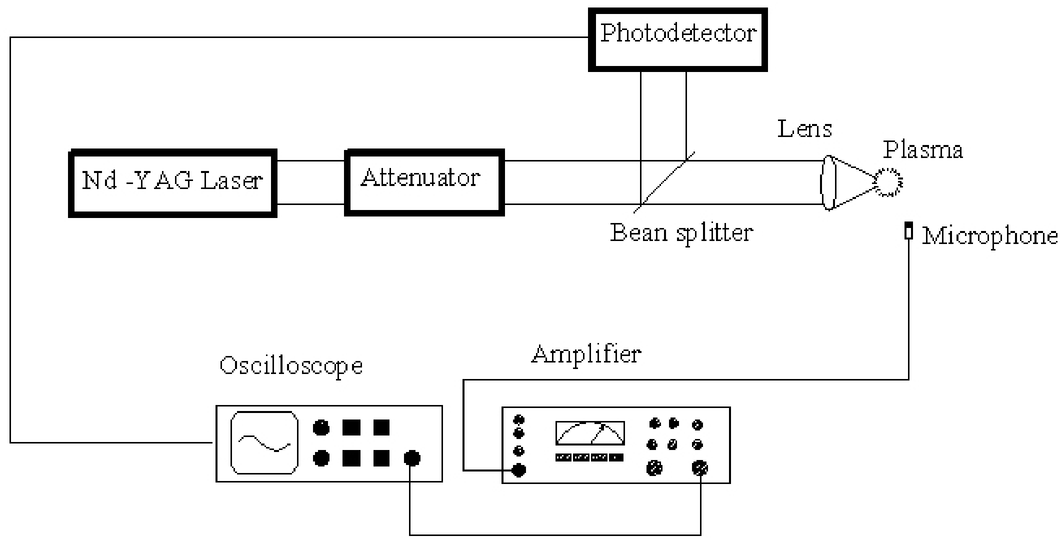

Figure 1. Microphones measurement setup: Laser impulse method

The acoustic signals registered by the microphone were conditioned with the measurement amplifier and captured by the oscilloscope through the signal synchronization supplied by the photodetector.

The oscilloscope was adjusted to register the waveform of the signal time in the following way: 200 μs/div of time frame (the screen had 10 divisions) and 5000 samples by screen. With this, a sampling frequency of 2.5 M Hz, a resolution in time of 0.4 μs and a capture time of 2 ms were obtained. The signal time was stored in diskette, with WFM format, for its subsequent postprocessing in MATLAB [4].

Using a MATLAB™ program, we processed the time signal obtained in the measurement. These time signal was mutliplied by a time window (Hanning window), and amplitude and phase of the frequency response was obtained by applying the FFT (Fast Fourier Transform). Figure 2 shows an example of measurement for an electret microphone.

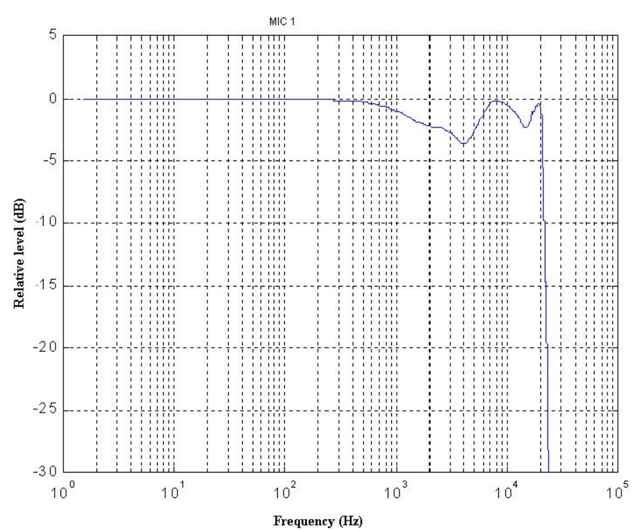

Figure 2. Exemplary frequency response (electret microphone)

3. EARPHONE MEASUREMENT WITHOUT HA

In this case, we have used random noise as source, and the earphone response in ear simulator was measured as it is specified in the Standard IEC 118-2 (amendment 2). Figure 3 shows the used experimental setup. In this method, the estimate of the transfer function H_1 is used, defined by:

$$H_1(f) = \frac{G_{xy}(f)}{G_{xx}(f)} \quad , \tag{2}$$

where $G_{xy}(f)$ is the cross spectrum between the input/output signals and $G_{xx}(f)$ is the autospectrum of the signal input.

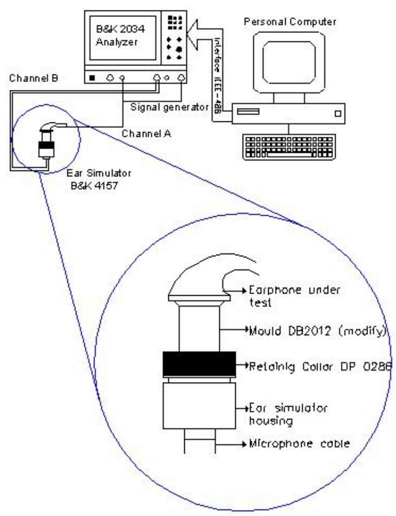

Figure 3. Earphones measurement setup

In this measurement neither was an anechoic chamber used. Figure 4 shows a measurement of earphone frecuency response for the case of Sony™ earphone.

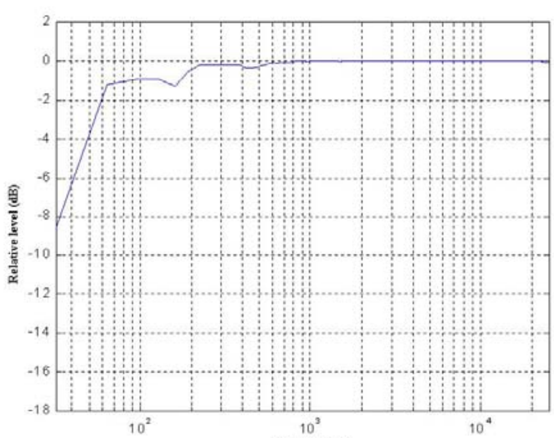

Figure 4. Earphone frequency response

4. AGC MEASUREMENT

It is important to know the frequency response of HA when the automatic gain control is active. The object of this test was to determine the dynamic characteristics of the circuit of ACC, particularly the attack time and recovery time. It should be emphasize that all these characteristics depend on the test frequency and another factors, such as the signal levels, position selection on controls and battery voltage.

On this step, it is necessary to provide some definitions of the terms used in the characterization, such as:

- *Automatic gain control (*AGC): A mean in HA by which the gain is automatically controlled as a function of the magnitude of the envelope of the input signal or other signal parameters.
- *Steady-state input/output graph:* The graph illustrating the output sound pressure level as a function of the input sound pressure level for a specified frequency, both expressed in dB on identical linear scales.
- *Comprehensive frequency response curves:* A family of frequency curves obtained with the gain control in the reference test gain position using a series of input sound pressure levels to exhibit the input/output characteristics of the HA over its full range of operations.
- *Attack time*: It is the interval of time between the moment when the input signal level is increased abruptly by a stated number of dB and the moment when the output sound pressure level from hearing aid with the AGC circuit stabilizes at the elevated steady- state level within $\pm$ 2 dB.
- *Attack time for the normal dynamical range of speech*: The attack time when the initial input sound pressure level is 55 dB and the increase in input sound pressure level is 25 dB.
- *Attack time (response time) for a high level*: It is when the initial input sound pressure level is 60 dB and the increase in input sound pressure level is 40 dB.
- *Recovery time*: The time interval between the moment when the stated input signal level is reduced abruptly to a level stated number of dB lower (after an amplifier provided with AGC has reached the steady – state output state under elevated input signal conditions) and the moment when the output sound pressure level from the HA stabilizes again in the lower steady- state level within $\pm$ 2 dB.
- *Recovery time for the normal dynamic range of speech*: The recovery time, when the initial input sound pressure level is 80 dB and the decrease in input sound pressure level is 25 dB.
- *Recovery time for a high interval*: It is when the level of initial input sound pressure is 100 dB and the decrease in input sound pressure level is 40 dB.

We have used a measurement method for dynamic output characteristics for speech levels. At the maximum setting of the gain control, an input signal of 1600 Hz or 2500 Hz, when appropriate, with a sound pressure level of 55 dB was applied. Any adjustable gain control after the AGC loop should be adjusted in such a manner that an overload of the HA was avoided. This signal was modulated by a square envelope pulse raising the input level by 25 dB. The pulse length should be at least five times longer than the attack time being measured.

If more than a single pulse is applied, the interval between two pulses should be at least five times the longest recovery time being measured.

This test may be carried out at various control settings (several positions of tone control, gain control, etc.).

It is necessary to point out that the loudspeakers employed for the measurement of the dynamic output characteristic should be sufficiently free of transient distortion so that test results are not appreciably affected.

In resemblance to the method of previous measurement, we have used a measurement method for dynamic output characteristics for high input levels. At the maximum setting of gain control, an input signal of 1600 Hz or 2500 Hz, when appropriate with a sound pressure level of 60 dB was applied. Any adjustable gain control after the AGC loop should be adjusted in such a manner that an overload of the HA was avoided. This signal is modulated by a square envelope pulse raising the input level by 40 dB. The pulse length should be at least five times longer than the attack time being measured.

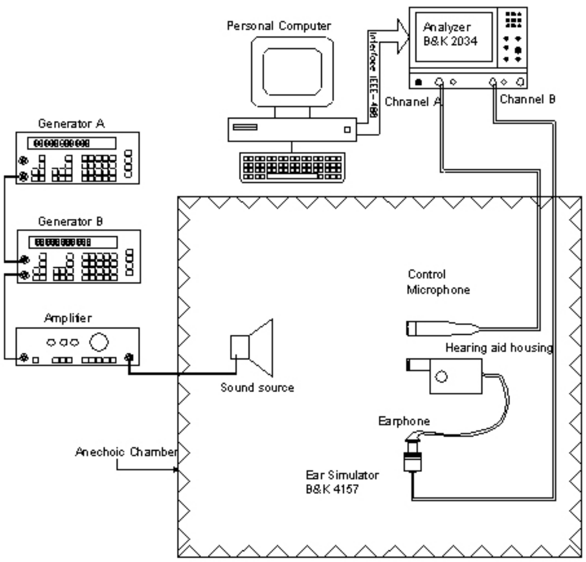

Figure 5. Attack time, response AGC

For the measurement of the attack time and the recovery time AGC, we have used the experimental setup shown in figure 5. Two generators were used, one to generate a signal of 1600 Hz and other to modulate with a squared signal and thus obtain a variation from 25 dB. The generator A was a Hewlett Packard model 33120a, and the generator B was a Stanford Research System model DS 345. The control microphone was a B&K 4165 (with pre-amplifier B&K 2639). The time signal was digitalized in the spectrum analyzer B&K 2034 and sent to a PC to be processed through MATLAB. Figures 6 and 7 show examples of the graphs obtained and the values of attack time and recovery of the AGC for certain control settings. The output sound pressure was measured connecting the earphone to the input of an ear simulator according to the norm IEC 118-2.

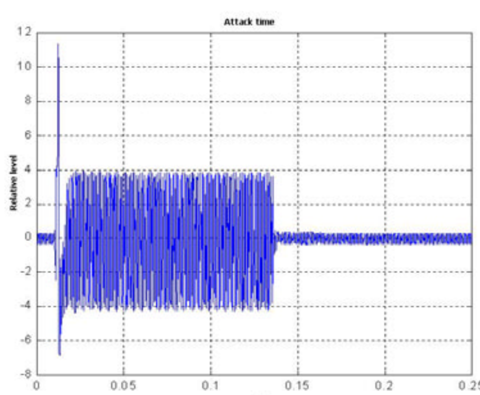

Figure 6. AGC Measurement setup

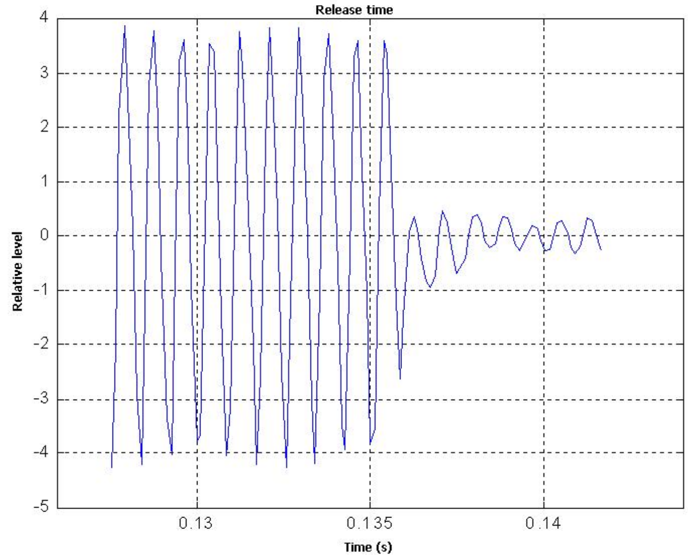

Figure 7. Recovery time, response AGC

5. STEADY-STATE INPUT/OUTPUT GRAPH

This graph (Fig. 8) represents the output sound pressure level as a function of the input sound pressure level for a specified frequency, both expressed in dB on identical linear scales. The input acoustic pressure levels are in the abscissa and the output acoustic pressure levels in the ordinate.

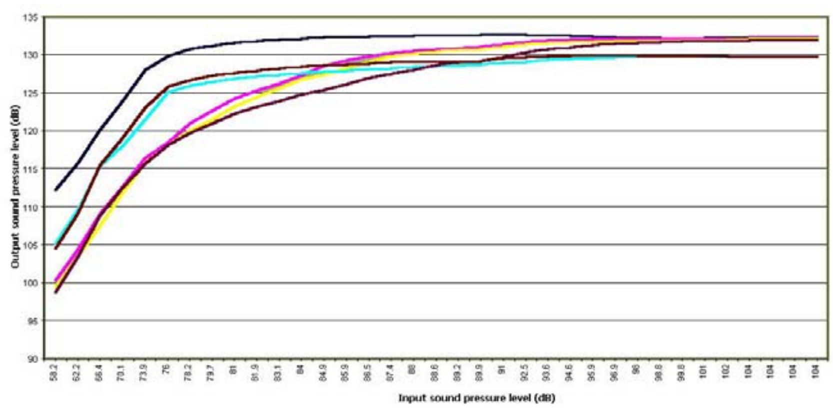

Figure 8. Steady-state input/output graph

In this graph different portions may be distinguished: below the lower AGC limit, the slope is esentially 45º (linear amplification mode). Above this limit, the graph curves over in a portion having a decreasing slope, often followed by another portion having a nearly flat slope (AGC mode)

At very high input levels, the flat or sloping portion may be followed by a portion with a steeper slope, generally due to saturation of the AGC. The lower AGC limit or AGC threshold is defined as follows: The input sound pressuere level that, when applied to the hearing aid, gives a reduction in the gain of 2 ± 0.5 dB with respect to the gain in linear mode.

On the other hand, the compression ratio (between specified input sound pressure values) under steady-state conditions reports the ratio of an input sound pressure level difference to the corresponding output sound pressure level difference, both expressed in dB.

To carry out these measurements, we have used the following method: The gain control was adjusted to its maximum setting (overload of HA is avoid). An input sound signal of frequency 1600Hz or 2500 Hz, when appropriate, was applied at the lowest possible level consistent with an adequate signal-to-noise ratio of preferably more than 10 dB. The input sound pressure level was increased up to 100 dB in suffiiciently small steps, and the corresponding output sound pressure level was measured after steady-state conditions were reached. The graph is plotted with the input sound pressure level as abscissa and the output level as ordinate.

When separate adjustable controls exist, such as AGC, gain or output controls, which will influence the shape and another characteristics of the steady-state input/output graph, it is recommended that input/output graphs should be plotted, when useful, for various additional stated setting of such controls.

The following family curves were obtained with the setup shown in figure 5; they correspond to gain curves for different settings of the controls of AGC and the equalizer.

The parameters that can be identified are shown in figure 9, which corresponds to one of several settings of gain control.

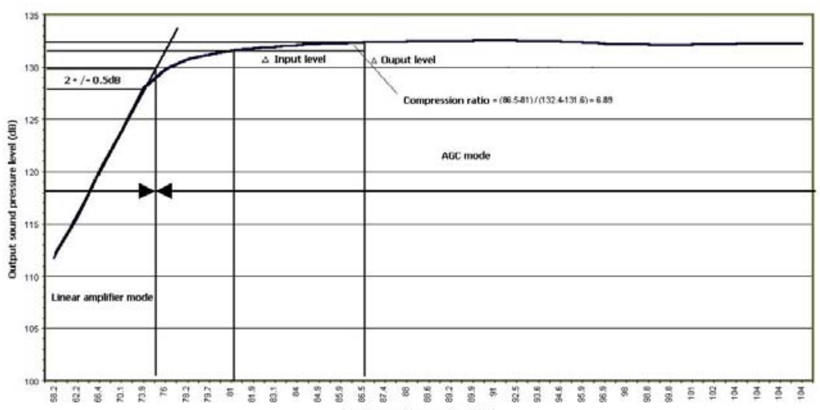

Figure 9. Steady-state input/output graph, for a specific setting of gain control

6. COMPREHENSIVE FREQUENCY RESPONSES AND BASIC FREQUENCY RESPONSE

The purpose of this test is to show the effect of input sound pressure level on the frequency response of the HA. The resulting family of curves will depict the comprehensive frequency responses and will indicate the input/output characteristics of the HA.

Test procedure.

- Adjust the gain control to the reference test gain position and set other controls to requiered positions.
- Vary the frequency of the sound source over the recommended frequency range from 200 Hz to 4 000 Hz keeping the input SPL constant at 50 dB, 60 dB, 70 dB, 80 dB and 90 dB.

- Plot the ear simulator SPL versus frequency at a constant input SPL, with a curve for each of the input sound pressure levels used.

From these curves, the frequency range is determined approximately to be the one in which the AGC is operating, and the approximate value of the threshold of the AGC in this range is determined.

An accuracy determination of the AGC operation can be made by using the input/output curves at specific frequencies.

The curves of figure 10 show that the signal is not distorted to the output of the HA upon modifying the input SPL.

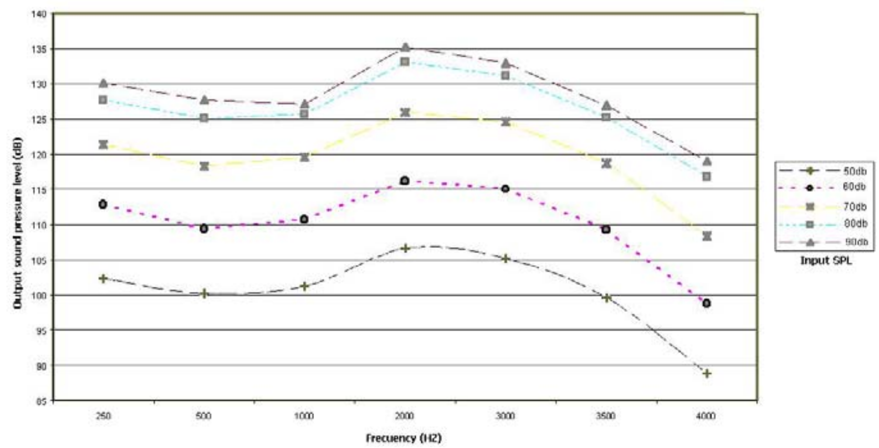

Figure 10. Frequency response of the AGC for various levels of input pressure

7. FREQUENCY RESPONSE MEASUREMENT, MICROPHONE + HA + EARPHONE

For frequency response measurement, a method for the measurement of HA frequency response using a steady-state broad-band input signal and employing dual channel spectrum analysis to measure the frequency response was used. A setup similar to the one shown in figure 5 was used, but only the B generator was employed. The coherence function was used to validate the frequency response. Figure 11 shows this response for an equalizer condition and a certain setting of the AGC.

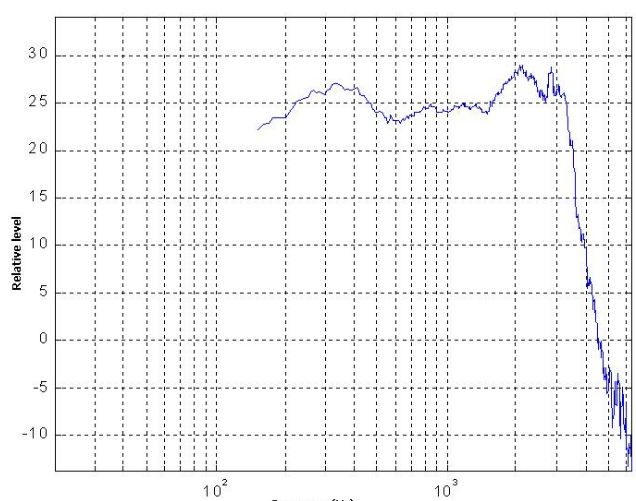

Figure 11 Frequency response (microphone + HA + earphone)

8. CONCLUSIONS

Various electroacoustics tests were carried out to our ptototype based on the different recommendations established in the international procedures ANSI an IEC for hearing aids. Also a new method for frequency response acoustic measurement (laser impulse method) has been presented. The setup used is simple and can easily be adapted for electroacoustic measurement of hearing aid microphones.

From the obtained results, it can be observed that our prototype hearing aid complies with the characteristics presented by some commercial equipment, something which makes it a good alternative for its utilization by people with certain hearing impairment, but with a lower cost compared with the commercial equipment of the same type.

9. REFERENCES

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Was born in Puebla, México. He received the M.Sc. degree in electrical engineering from *Centro de Investigación y Estudios Avanzados del Instituto Politécnico Nacional (CINVESTAV,IPN)*, México in 1992. Currently, he is working in the Acoustics laboratory of the *Centro de Ciencias Aplicadas y Desarrollo Tecnológico, CCADET, UNAM*. His current research is focused on the development new electroacoustical measurement techniques.

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