Assistive Listening Device based on a dsPIC

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Abstract— **Hearing impairments are a condition that affect a high percentage of the Society. Although there are devices that improve the quality of life for people with these problems, they tend to be expensive and not to satisfy all the user requirements. This work focuses on the development of a portable device for assisting people with some kind of hearing impairment, which is implemented using a Microchip dsPIC. In order to validate the results, a model capable of hearing impairment loss is used, which allows to analyze the performance of two compression algorithms: linear and SPINC. Subjective experimental results show the advantages of using SPINC function, which is implemented in a Microchip device dsPIC33FJ128GP802-E/SP.**

I. Introduction

Hearing impairments are a condition that affect a high percentage of the Society. There are several types of hearing impairments, one way to classify them is by their origin, in: congenital hearing impairment, disabilities due to the aging of the person or disabilities originated in a trauma. However, the most practical way to represent it is through the curve that represents the Sound Pressure Level (SPL) as a function of the frequency, also called **Absolute Threshold of Hearing** (ATH) [1], [2], [3], [4], [5]. It represents the minimum required energy for a signal of a specific frequency, so that it can be perceived by a person with a healthy ear in a noise-free enviroment. Then, the components under this threshold can not be perceived by the listener, in this way, is possible to classify a hearing impairment as a function of the ATH curve. For example, an auditory model with a **Severe Hearing Impairment** is characterized by an ATH curve with an additional reject of at least 60dB in the range of frequencies above 300Hz or 500Hz. This disability, generally is due to the aging of the person. Another hearing impairment under study, is a traumatic deficiency, which is characterized by an ATH curve with an additional reject of at least 50dB in a specific band of frequencies. This type of disability is originated by the exposition of the person to a constant sound of a specific range of frequencies, as in an industrial environment. This hearing impairment causes that the sounds with frequencies in the range of rejection can not be perceived by the person.

To assist people with this and others disabilities, applicable techniques have been developed to correct this deficiency, which have been used to design several types of assistant devices, for instance, headsets. Examples of these devices are developed by Widex or Samsung. These Companies offer several types of products, from analog headsets (in some cases only a fix-gain amplifier) to devices based on digital signal processors[6], [7], which are capable of obtaining better results than other analog devices. The last ones allows to perform tasks like voice compression, noise reduction and dynamic equalization, etc. The most important feature of this type of devices is that it allows to obtain a solution for each hearing impairment, adjusting it to a particular user. But the special case under study in this work are the hearing impairments where the deficiency affects a particular range of frequencies, also called , which can be faced using **voice compression algorithms**, which compresses the spectra in a way that the information that was originally in the dead zone, is mapped to an area where can be perceived by the person. This is the case of hearing impairments like a traumatic deficiency or severe hearing impairment. Compression is accomplished by using a mapping function that compresses the information according the needs of the user. Several types of these functions are cited in the bibliography. The most employed ones use linear compression based on in the **Fast Fourier Transform (FFT)** [8], [9], [10], there are also some based on sinusoidal models [11], [12], and systems implemented using filter banks [13], [14]. The goal of the present paper is to develop techniques to assist people with a particular hearing impairment, the results of this analysis will be obtained using a model that allows to simulate several types of degree of impairment. Then, the compression algorithms will be implemented in a DSP device dsPIC family from Microchip. The device choosen to face the problem is the dsPIC33FJ128GP802-E/SP [15]. Thus, it is possible to make a comparison between the developed and existing systems, aiming to improve the specifications and to reduce the costs. This work is structured as follow: In Section II a theoretical introduction of the auditory model used in this work is developed, to help the reader get familiarizated with this model. In Section III the hearing impairments that will be simulated are presented. In Section IV, the proposed algorithms to face the problem are developed, analyzing the advantages and disadvantages of each one. In Section V, the results of subjectives experiments are presented and discussed. In Section VI, a practical implementation of the proposal is developed. Finally, the main conclusions are summarized in Section VII where the guidelines to be followed in future work are also presented.

II. Auditory System Model

The hearing impairment simulator is based on an implementation of MPEG-I Audio Layer 3 compression standard [1], [2], which obtains high compression levels by considering the perceptual limitations of the human hearing system. Then, it removes the unnecessary information that the human hear can not perceive. Fig. 1 shows a schematic of a generic coding system of this model.

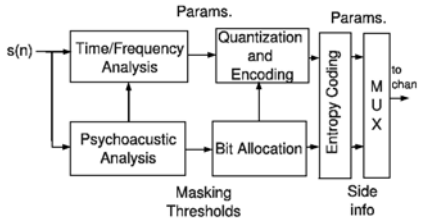

Fig. 1. Model of a perceptual audio encoder.

The system proposed in this Section only uses the Psychoacoustic model I and the filter banks of the coder. This decision is made because the main goal of the system is not to code the signal but to find the part of the information that is not necessary and must be eliminated. The most important aspects of this model are presented in the next Sections.

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The compression process starts with the analysis of the response of the auditory system at each of the input frequencies. It is made using the **Absolute Threshold of Hearing** ( ), that is a function which represents the minimum amount of energy needed by a pure tone of a given frequency to be detected by a listener in a noiseless environment. The ATH is typically expressed in terms of dB SPL. The dependence of this threshold with the frequency was quantified by Fletcher [16], who reported test results for a range of listeners. In the particular case of a quiet environment, this curve is modeled by 1.

$$T_q(f) = 3.64(\frac{f}{1000})^{-0.8} - 6.5e^{(f/1000-3.3)^2} + 10^{-3}(\frac{f}{1000})^4 dB \quad (1)$$

This threshold is representative of a young listener with acute hearing. A plot of this function is showed in Fig. 2.

Function of 1 is a general case, but in the case of a particular listener, it can be obtained through an audiometry, a medical study used to evaluate the frequency response of the ears of a listener. An audiometry is a test that can be used as a tool to obtain a model of an ear with a disability.

The hybrid filter bank is a polyphase filter bank composed by 32 subbands [4], which are narrower in the range between 2 and 4 kHz for the Layer 3. Due to this, the masking widths of the ear in these ranges are the slightest.

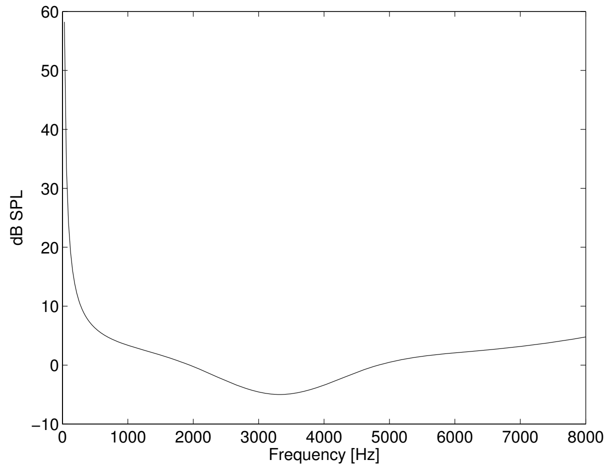

Fig. 2. The absolute threshold of hearing in quiet.

The reason for this implementation is to reduce the -
[17] due to the laps between bands, which originates audible distortions.

Once the two most important parts of a auditory system are defined, it is necessary to present a process known as masking. This process refers to a situation where one sound is rendered inaudible because of the presence of another sound. Simultaneous masking may also occur whenever two or more stimuli are simultaneously presented to the auditory system. The relative shapes in the magnitude spectra of the masker and maskee determine to what extent the presence of certain spectral energy will mask another spectral energy. There are several types of masking between spectral components, but it is convenient to distinguish between only three types of simultaneous masking:

- **Noise Masking Tone (NMT):** In this case, a narrow band noise masks a tone within the same critical band, provided that the intensity of the masked tone is below a predictable threshold directly related to the intensity.
- **Tone Masking Noise (TMN):** In the case of TMN, a pure tone placed in the center of a critical band masks noise of any subcritical bandwidth or shape, provided that the noise spectrum is below a predictable threshold directly related to the strength of the masking tone.
- **Noise Masking Noise (Noise):** In this scenario, where a narrow-band noise masks another narrow-band noise, it is more difficult to characterize than in the case of NMT or TMN, because of the confounding influence of phase relationships between the masker and maskee.

The spectral masking is asymmetrical, because the masking thresholds are not the same in each side of a component. This aspect is shown in Fig.3 where the masking limits are presented.

Another type of masking is the non-simultaneous masking. It is a phenomena that extends along the time. For a masker of finite duration, this type of masking occurs

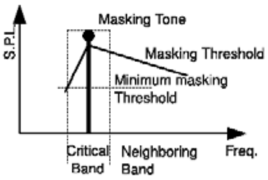

Fig. 3. Masker and its correspondent masking thresholds.

because the ear needs a finite time since the masker dissapears and the next component appears. The ear requires a recover time to return to its normal state. This recovering time is about 5ms for a healthy ear.

III. Hearing Impairments under study

In this Section some types of hearing impairments, and their representation using the Absolute Threshold of Hearing curve in each case are presented. In the first place the case of a **healthy ear of a young person**, which is shown in Fig. 2.

If simulation of a **severe deafness** is wanted to be modeled [3], [18], [19], where the auditory system does not respond to frequencies higher than a cutoff frequency of around 300Hz to 500Hz, can be modeled by increasing at least 50dB the ATH curve [3], in the range of hearing disability, as shown in Fig. 4. This impairment causes the loss of the majority of the components of a phoneme.

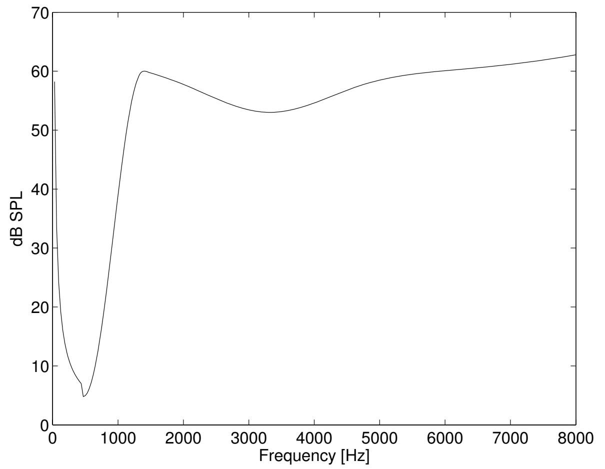

Fig. 4. An example of a curve of the absolute threshold of hearing with a severe deafness.

Fig. 5 shows the ATH curve of a person with a **bilateral acoustic trauma** [18], [19], [21], a hearing impairment that causes a rejection in a band of frequencies. It is a pathology characterized by a , and causes the loss of some types of critical sounds found in phonemes [17] like consonants as the /r/ and /s/, where the loss of critical components may cause that those two phonemes sound equal.

IV. Implemented Methods

This sections is divided in two parts: First one introduces a model that implements tools capable of simulating the hearing impairments developed in Sec. III. Second one presents an algorithm based on a voice compression method, that increases the intelligibility of the processed signal.

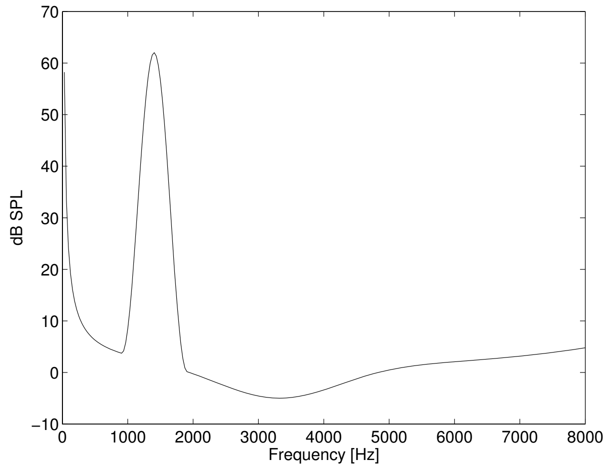

Fig. 5. The absolute threshold of hearing with a bilateral acoustic trauma.

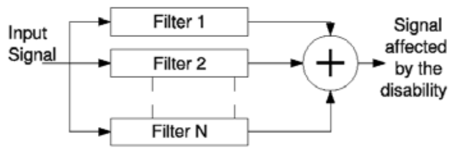

Fig. 6. Model of the hearing impairment simulator proposed .

This implementation is done by using a filter banks model [16], [4], [22], [23], [20], which is based on the parameters presented en the Sec. II. Fig. 6 shows the proposed model.

The implementation of the psychoacoustic model is done by using several tools that allows to represent the auditory system of a person with a particular hearing impairment [24]. This model allows to know -

, and consequently design solution for each specific case. In this case, the severe hearing impairment was modelled by using a low pass filter with a cut-off frequency and a attenuation variable according the disability and the model for the acoustic trauma was modelled using band-stop filter designed according the hearing impairment of interest.

The voice compression algorithms, are based on frequency shifts and scalings, for each input frequency a correspondent output frequency is obtained, which is related with the input frequency by a factor and a displacement of frequency. This relation between input and output frequencies can be given by a linear law, a logarithmic law, or a more complex law. Several functions that implement these laws [7], [10], [26], are cited in the bibliography. The main goal of a voice compression algorithm is to use a

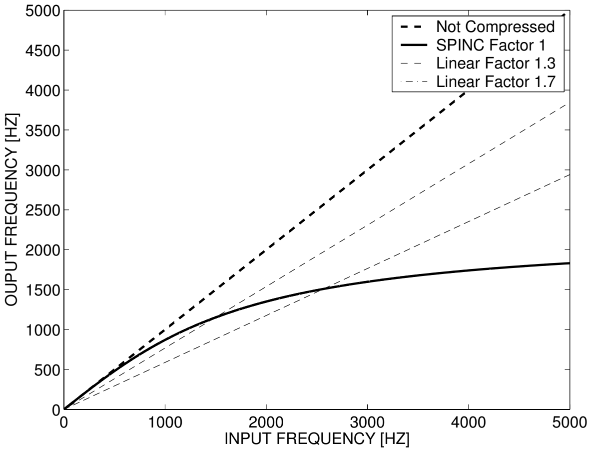

Fig. 7. Compression functions implemented.

low compression index in the lowest frequencies (due to the most important formants of the voice are placed between 0Hz and 1KHz, and these formants have most of the energy [10]), and a higher compression index in frequencies above 1000Hz. Another aspect to consider is that the information required to recognize a particular speaker is placed between 1000Hz and 2000Hz [25]. Although there are several types of compression laws, one of the most used is the linear law, which compresses the input frequencies by a constant factor. This method has the disadvantage that compresses the lowest frequencies and consequently the comprised voice sounds like artificial and unnatural. Also, if a factor of $k = 1.3$ is used, the voice frequency range (0-20KHz) is compressed to a band of approximately 15KHz. The advantage of the linear compression is that it has low computational load associated. In order to improve the performance of the linear compression, it is studied a compression law based on an arctangent function, called **SPINC** [26], which according 2, obtains the output frequency $\phi(f)$ as a function the input frequency (f).

$$\phi(f) = 1414arctg(f/1414Hz) \quad SPINC. \tag{2}$$

Fig. 7 shows the SPINC function (continuous bold line) and two linear compression functions, one with compression index $k = 1.3$ (dashed line), another with $k = 1.7$ (dashed and pointed line), and finally a reference linear function where the output frequency is the same than the input frequency, that is $k = 1$, (dashed bold line). The SPINC function has several advantages over the linear functions, but the most important is that the SPINC function does not compress the range of frequencies since 0 to 1400Hz, consequently the intelligibility of the voice is not affected. In Addition, the voice spectrum is completely compressed in a bandwidth of 2200Hz (a linear law with a $k = 1.3$ requires a bandwidth of 15KHz to transmit 20Khz), which represents a significative reduction of the bandwidth, but it causes a reduction in the ability of recognizing the identity of the speakers. This paper applies the Timms proposal [19], in which it is performed an overlapped analysis of the signal is performed by using the Fast Fourier Transform (FFT). Then, the spectral compression is accomplished by relocating the spectral components according to the SPINC function. Finally a low-pass filtering is applied and the signal is processed in order to recover the hermitical simetry. Parameters used in this implementation are listed below:

- **Antialiasing** filter [27], [17].
- Frequency domain operations using a 256-points FFT. Then, a spectral resolution of 62.50Hz is obtained (Sampling Frequency = 16KHz).
- Temporal analysis using 256-points frames.
- SPINC compression function.
- Frequency-time transformation using IFFT.
- Using a **TD-OLA** synthesis technique [17].

Thus, a system able to compress the sounds from a band where the listener has an specific problem is obtained, to another band where the user can perceive it is achieved. The computational load of this system depends on the number of samples of each frame, and therefore, also depends on the spectral resolution adopted. Because the main goal of this work is to obtain a real-time system, the development done looking for a trade-off between computational load and spectral resolution.

V. Experiments

In order to obtain a measure of performance of compression methods, the functions used are implemented in MATLAB and two subjective test are made. In this way, the sounds were processed using the SPINC function, and two versions of the linear law, ($k = 1.3$ and $k = 1.7$). The audio database used for this experiment contained 35 sentences in Spanish, uttered by two male and two female speakers. The sampling frequency was 16 KHz and the average duration of the sentences was 4 seconds. Other sounds are used in the test (for example, the sound of a train alarm), which are of vital importance in the everyday life of a person.

The system is was tested by using a simulated hearing impairment, which models a severe deafness 4, with a rejection of 50dB added to the ATH curve in the frequencies above 870Hz. In this way, an uncompressed sound may be unintelligible. Fig. 8 shows a schema of the proposed experiment.

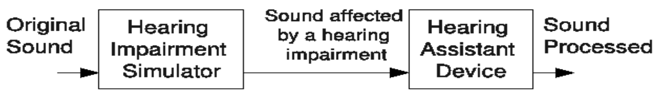

Fig. 8. Experiment Proposed.

Two subjective experiments were conducted: one to evaluate the **Quality**, and other to evaluate the **Intelligibility** of the processed sounds. In this stage, 15 volunteers were asked to listen to the processed sentences. Listeners were asked to evaluate the quality and the intelligibility of the processed sentences from 1 point (bad) to 5 points (excellent). The experiment is composed by audios compressed according the following functions:

- Low-pass filtering, $fc = 870Hz$, using a linear compression function with $k = 1.3$ and $k = 1.7$.

• Low-pass filtering, $fc = 870Hz$, using a SPINC function with compression factors $k = 1$ and $k = 1.3$.

	Intelligibility	**Quality**
SPINC C. - Factor: 1.0	**2.4**	**2.2**
SPINC C. - Factor: 1.3	2.1	1.9
Linear C. - Factor: 1.7	1.7	1.8
Linear C. - Factor: 1.3	1.7	1.7

TABLE I
Experimental results.

Table I presents the obtained results, which show an advantage of the SPINC algorithm over the linear compression functions. Also, the SPINC with $k = 1$ obtains the best results in the test of inteligibility, it is because it does not compress the lowest formants of the voice, which have the most important acoustic information. The other functions, compress that range of frequencies, so that the processed sound can be unintelligible.

Another important aspect, is that some alarms can not be heart without compression or a linear compression. This has to be solved with the SPIC algorithm, because a a hearing-impaired person must be able to hear those sounds in his normal life.

VI. Implementation in a dsPIC33FJ128GP802

Due to the results of the tests using MATLAB and MPLAB were promising, the algorithm was implemented using a device **dsPIC 33FJ128GP802-E/SP** from Microchip [15], [28], then is possible to implement similar capabilities to those of commercial devices. Examples of such devices are BRAVO [6] and SENSO [7], both of the company WIDEX, which have characteristics such as noise control, voice compression and equalization, etc. Another advantage of these digital devices, is that they allow to adjust the system to the requirements of the user, which depends of the degree of the hearing impairment, with just reprogramming, reducing the calibration process. For the reasons previously mentioned, these assistant devices are very versatile. However, this versatility has an impact on the cost, due to the value for a device is increased significantly, for example, a device similar to the one proposed here has a price between 1000 and 5000 Euros. Moreover, these devices are based on **Digital Signal Processors (DSP)**, and by their nature require external circuitry, which when trying to implement one of them results in an increase in the size of the system. On the contrary, the proposed dsPIC has a large number of integrated components, thus reducing the size of the system to be build, and a considerably lower price (the extended temperature version cost only 17 US\$). This makes it ideal for the construction of an assistant device.

The main features of this device are:

• **128KB of program memory**. This makes it suitable for use with cross compilers.
• **16KB of RAM**. Of which 2KB are shared with direct memory access **(DMA)** buffer as dual ported RAM.
• **Up to 40 MIPS operation**.
• **Low cost**. As mentioned before, its price is 17 US\$, much lower than a classic DSP. This is a great advantage compared with comercial DSP devices.
• **16-bit wide data path**.
• **12-bit@500ksps integrated analog-to-digital converter (ADC)**.
• **16-bit@100ksps integrated digital-to-analog converter (DAC)**.
• **Double-buffered input/output registers**. This allows for faster operations on the ports (read-write), and also gives more flexibility on the handling of them.
• **In-Circuit Serial Programing and Debbuger**. The device can be programmed and configured in the end application circuit.
• **28-pin SOIC packaging available**. Allows for great levels of integration.

It should also be noted that the documentation about the Microchip devices and their libraries is available on the internet without any cost. Perhaps the greatest advantage of the dsPIC33FJ128GP802 is that allows of running multiple tasks **simultaniously**, in this particular case those tasks are the acquisition of a data segment and the processing of the previous one. This is acomplished by using the DMA module of the device [15], which operates independently from the main processor. As a result, the processing time of each segment is reduced to almost half. Considering the fact that the main goal of the project is to develop a device to work in real time, this time saving is of utmost importance. Another aspect to be taken into account, is that the utilization of DMA techniques increases the system performance, because it minimizes interrupt sources. In other words, peripherals perform the data transfer using the DMA module and so delays are not added to the main program execution. In particular, the reduced times are:

• Processing time of the interruption routine.
• System stack accessing, reading and storing time.
• Access time of peripherals.

The implemented voice compression algorithm is described in Sec.IV-B. This device can be adapted to the needs of each individual user, by simply changing the compression factor value and the displacement to be made. Because the employed dsPIC does not have hardware support to perform floating-point operations, those operations must be emulated in software, which greatly decreases performance. Hence, the FFT algorithm was adapted to work in 16-bit fixed-point arithmetic [29] using magnitude scaling. The obtained system is shown in Fig.9, and the involved times in data processing block can be seen in Table II

It is noted that the processing time of the compression algorithm is significantly less than the one for the FFT and IFFT algorithms in both configurations. Once implemented the system, full processing times were registered, which are presented on Table III.

The values recorded in the table were measured using a

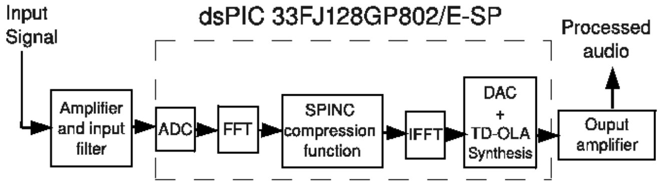

Fig. 9. Outline of the proposed system.

	FFT proc.	**SPINC proc.**	**IFFT proc.**
128	$2.3ms$	$0.14ms$	$3.1ms$
256	$5.2ms$	$0.29ms$	$7.1ms$

TABLE II
Segment processing times.

	Pr. T	**Acq.**	**RAM**	**DMA**	**Prog. M.**
128	$5.3ms$	$7.86ms$	19%	50%	4%
256	$12.2ms$	$15.72ms$	34%	100%	4%

TABLE III
Processing time, acquisition time and memory usage for two implementations of the compression algorithm, one with frames of 128 samples and other with frames of 256 samples.

$f_{sampling} = 16.288KHz$, where is possible to see a difference between the acquisition and processing times. This difference occurs because the tasks run in parallel, so the total processing time of the system is regarded as the greatest of them, which in the case of segments of 256 samples is $t_{total} = 15.72ms$. Another aspect to consider is that while the levels of RAM and program memory remain low, there is a limitation due to the amount of DMA memory available (in this case is 2KB). This imposes a limitation on the maximum performance that can be obtained from the system.

VII. Conclusions

Throughout this work was described the development of a device capable of voice compression for people with varying degrees of hearing impairment. It was possible to demonstrate that by using more powerful algorithms, such as SPINC, and devices with greater computing power, is possible to implement a device similar to those found in the market. This is because, for example, experiments showed an improvement of SPINC algorithm against linear compression, which is used in many consumer devices.

Another important aspect is that the development was performed using a commercial device, for which it was demonstrated that processing times are consistent with a real time implementation.

The only limitation that emerged in this development was the amount of memory available to the DMA module, which is why in future work will be used a microcontroller with larger capacities. Following the same line of work, it is intended to include more features to the hearing assistance device.

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