Digital Filter Bank for Hearing Aid Application Using FRM Technique

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***Abstract*—A lot of effort has been devoted to the design of uniform and non-uniform filter banks for hearing aid applications. But in most of the cases, it never deals with large variation of hearing losses at mid frequencies. This paper presents a low complex design of a non-uniformly spaced digital finite impulse response (FIR) filter bank for digital hearing aid application. Frequency response masking (FRM) technique is used for the implementation of 8 non-uniformly spaced subband filters, with a single half-band filter as a prototype filter. With FRM technique and half-band filter, a drastic reduction in the number of multipliers and adders in linear phase FIR filter can be achieved. Further complexity-effective design can be achieved by producing masking filter from the prototype filter. FRM technique is achieved by cascading different combinations of prototype filter and its interpolated filters to produce subbands. The simulation results shows that, the proposed filter bank gives 120 dB attenuation with 13 multipliers only. The proposed FRM based filter bank can be effectively used for the audiogram matching with large variation of hearing threshold in mid frequencies.**

***Keywords*—non-uniform filter bank, FIR filter, FRM technique, hearing aid, half-band filter, audiogram.**

I. INTRODUCTION

A hearing aid is an electro-acoustic device, which is designed to amplify sound, with the aim of making speech more intelligible [1]. The main task of the hearing aid is to selectively amplify the audio sounds such that the processed sound matches one's audiogram. Audiogram is a graph that shows the softest sounds a person can hear at different pitches or frequencies. It measures the people hearing threshold as a function of frequency. Hearing thresholds become high at certain frequencies causing hearing loss ie, they have low hearing sensitivity at certain frequencies. To compensate this type of hearing loss, it is necessary to selectively amplify sounds at required frequencies. The main purpose of hearing aid is to provide precise adjustment of the gain in the required frequency. To compensate different types of hearing loss, hearing aid is designed in such a way that it should be applicable to different types of hearing loss. Basic block diagram of digital hearing aid is shown in Fig. 1.

Fig. 2 shows an example of an audiogram for normal hearing and hearing impairment by noise exposure. Hearing losses can be compensated through tuning the subband gains of uniform or non-uniform filter banks. In-order to reduce the maximum matching error in hearing aid fitting, an increased number of frequency subbands are required because the actual hearing loss pattern (audiogram) differs from person to person [2]. So the filter-banks must have high tuning flexibility to fit various audiograms [3, 4].

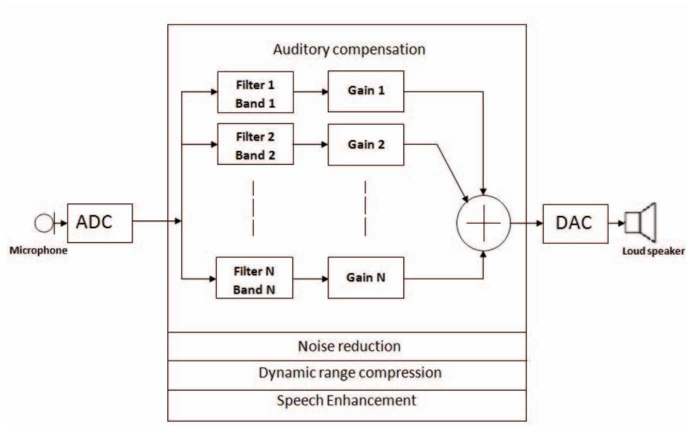

Fig. 1: Block diagram of digital hearing aid.

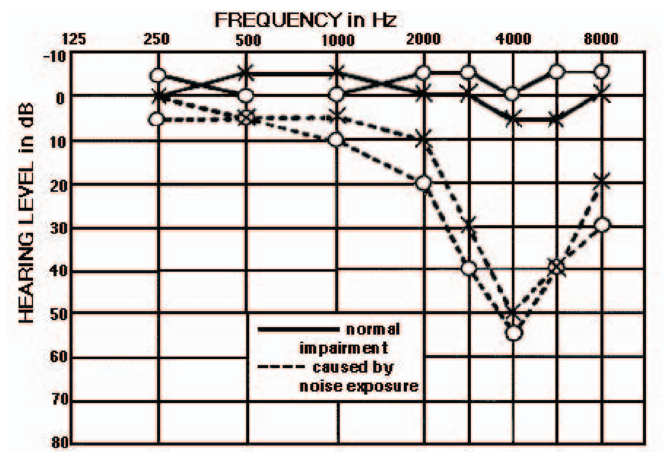

Fig. 2: A typical audiogram for the normal hearing and hearing impairment by noise exposure. 'O' and 'X' represent the thresholds of left and right ear respectively [14].

Based on the functionality, the hearing aid can be classified into three categories: analog, programmable analog and digital hearing aid. Analog hearing aid is the least expensive type, but the drawback is that, noise is also amplified without discriminating the sound. In programmable analog hearing aid a programmable control circuitry is added to the analog audio circuitry to program the gain and frequency settings, which results in a more complex one. Digital hearing aid has much advantages over analog systems because of the advanced digital signal processing (DSP) algorithms contained is to compensate speech, improved intelligibility in noisy environment and echo or feedback cancellation as shown in Fig. 1. Researchers have investigated several techniques suitable for hearing aid applications. These techniques include uniform filter banks , non-uniform filter banks and fast Fourier transform. In these techniques the frequencies of the audio signal are split into different bands and then amplification is provided according to the different levels of hearing loss [5].

As the number subbands (filters) increases better audiogram matching can be achieved. But this will increases the computational complexity in terms of the numbers of arithmetic operations (additions and multiplications) and in turn costs much power. Much effort is invested in the design of uniform digital filter banks for hearing aid applications [6, 7]. Typical hearing loss such as noise-induced hearing loss (NIHL) occurs at mid frequencies, sensorineural hearing loss (SNHL) occurs at high and low frequencies and hearing loss at high frequencies caused by aging. Uniform filter banks may face difficulties in matching the audiogram in all frequencies.

To achieve a better compensation, narrower bands need to be allocated at frequencies where sharp transition of hearing loss occurs. Therefore, a non-uniform spaced digital finite-impulse response (FIR) filter bank becomes very attractive [2, 9-12]. But in these cases [2, 9-12], it is mainly applicable for sharp transition of hearing loss at low and high frequency regions which makes it not very attractive for certain types of hearing loss. Audiogram matching error will be high for hearing loss such as NIHL, since they have a sharp transition of hearing loss at mid frequencies ($3kHz$ to $6kHz$). For good matching of audiogram for this type of losses, narrower bands need to be allocated at mid frequencies while in [2, 9-12] narrower bands are allocated at high and low frequencies. Also in [2, 10-12], filter bank is constructed by two prototype filters and in [9] it uses three prototype filters. In these cases, prototype filters have to design separately, so the complexity is more [9]. To avoid this, the proposed system use only a single half-band FIR filter as prototype. Masking filter is produced from the prototype filter by interpolating the prototype filter by different factors.

In this paper a low complex non-uniform FIR digital filter bank using frequency response masking technique is proposed, which provides better audiogram matching for sharp transition of hearing loss at mid frequencies also. With the help of FRM technique and half-band filter, large reduction in the number of multipliers and adders in linear phase FIR filter can be achieved.

The paper is organized as follows. Section II gives the design method for the proposed filter bank. Section III consist of experimental results and the result for the audiogram matching for different hearing loss. Section IV consist of future work and Section V concludes the paper.

II. STRUCTURE OF PROPOSED FILTER BANK

With FRM technique, linear phase FIR filters can be realized with guaranteed stability and phase response [13]. Linear phase digital filters have many advantages such as guaranteed stability, free of phase distortion, and low coefficient sensitivity under certain conditions. But the main disadvantage of linear phase FIR filter is its complexity due to the involvement of large amount of multipliers. In-order to reduce the filter complexity, the proposed non-uniform FIR filter bank uses only a simple half-band filter $H(z)$ as prototype filter and the subbands are designed with symmetry at the mid-frequency point. Masking filter is produced from the prototype filter by interpolating the prototype filter with different factors. The main role of masking filter is to remove the unwanted passbands in interpolated prototype. The resulting filter has very sparse coefficients. Since only a very small fraction of its coefficient values are non-zero, its complexity is very much lower when compared to other filters.

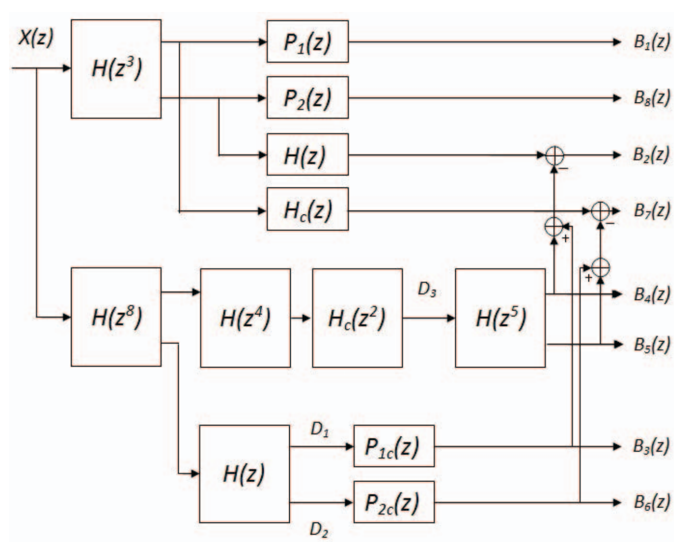

Fig. 3: Proposed filter bank.

The proposed filter bank shown in Fig. 3 is designed in such a way that subbands are symmetric. This is aimed at improving the matching performance at mid frequencies compared with the uniform filter bank. Each branch consists of two or more filters connected in series. Some of the filters in the Fig. 3 provide a pair of outputs, e.g., $B_4(z)$ and $B_5(z)$, D_1 and D_2 etc. At each branch, the top output comes from the original filter and the bottom one is from the complementary filter. This is formed with the help of a complementary filter pairs $H(z)$ and $H_c(z)$. $H_c(z)$ is a complement of an original filter $H(z)$, where $H_c(z)$ can be implemented by subtracting the output of $H(z)$ from the delayed version of the input as shown in Fig. 4. $H_c(z)$ is given by

$$H_c(z) = z^{-\frac{N-1}{2}} - H(z) \tag{1}$$

where N is the length of $H(z)$. The extra delays for deriving $H_c(z)$ from $H(z)$ need not be implemented explicitly, since the delays in $H(z)$ can be used for this purpose as shown in Fig. 4. Thus the hardware cost for producing the complementary output is minimized.

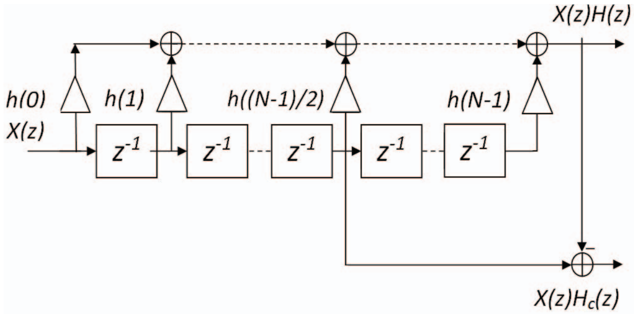

Fig. 4: Architecture of complementary filter.

First step is to design a half-band filter $H(z)$ as a prototype filter. In half-band filter case, all odd-indexed coefficients except for the central coefficient are zero valued making the implementation very attractive. An important advantage of a linear-phase half-band filter is the efficient implementation, which follows from two favourable properties of the filter impulse response. The first is that the number of non-zero valued coefficients is nearly half of the filter length. The second is the non-zero coefficients exhibit symmetry property. More reduction in number of multipliers can be achieved by designing a masking filter from the prototype filter when compared to [2].

The outputs of each subband are termed as $B_i(z)$, i=1,2,..,8, where $B_1(z)$ to $B_4(z)$ are mirror image of $B_8(z)$ to $B_5(z)$ with symmetry respect to mid-frequency point. Leading delays should be added to each filter in branches other than the top one to ensure that all branches have the same phase shift in order to achieve the desired frequency response and avoid frequency-dependant delay [2]. The basic idea is that firstly prototype filter $H(z)$ is interpolate by different factors ie, 2, 3, 4, 5 and 8, where interpolated by factor 3 and 8 is taken as primary filters. $H(z)$ and its interpolated form by 2, 4 and 5 and its complementary form is considered as masking filters. Then cascading primary filter by different combination of masking filters, each subband is formed. The working principle of proposed filter bank can be best illustrated by an example. $B_4(z)$ and $B_5(z)$ is produced when $H(z^8)$ is cascaded with $H(z^4)$ and $H_c(z^2)$ filtered through $H(z^5)$ and its complementary form respectively. In order to form the required bandpass filter as shown in Fig. 5, the unwanted bands at low and high frequencies are removed by masking filters. Similarly, $B_3(z)$ is a result of connecting $H_c(z^8)$ with $H(z)$ and the complement of $P_1(z)$ as shown in Fig. 5. The z-transform transfer function of subband bands are shown in TABLE I.

The implementation of a 8-band filter bank needs six subfilters and their complementary filters and are made i the same way as shown in Fig. 4. Also the multipliers can be shared among interpolated $H(z)$ same as that in Fig. 4. The frequency response masking is achieved by the repeated use of $H(z)$ and its interpolated filters.

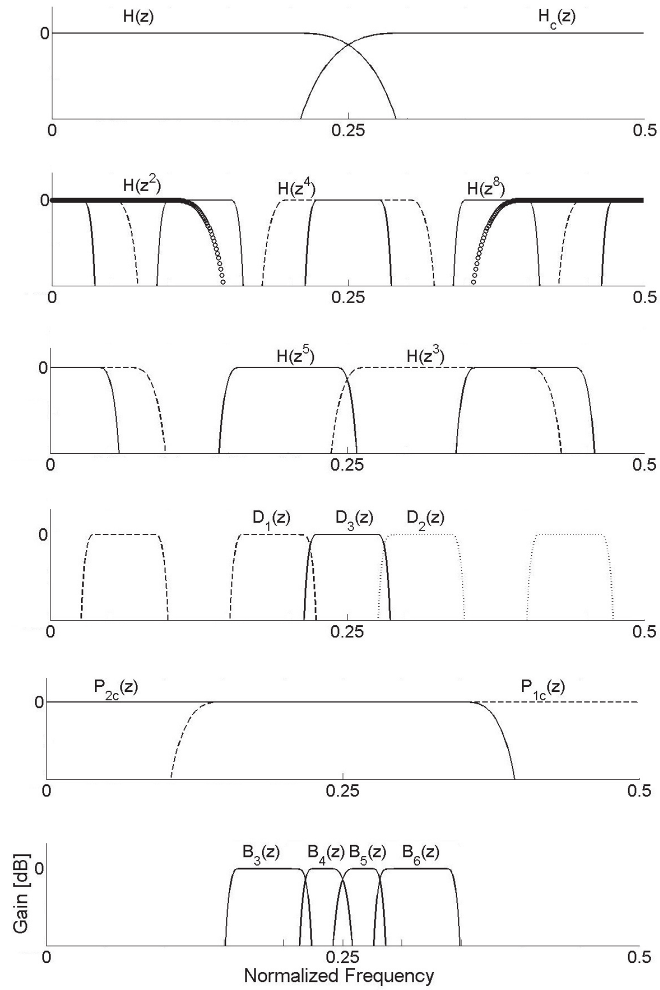

Fig. 5: Magnitude response of an 8 band non-uniform filter bank.

Band	Transfer function
1	$B_1(z) = H(z^3)P_1(z)$
2	$B_2(z) = H_c(z^3)H(z) - (B_3(z) + B_4(z))$
3	$B_3(z) = H_c(z^8)H(z)P_{1c}(z)$
4	$B_4(z) = H(z^8)H(z^4)H_c(z^2)H(z^5)$
5	$B_5(z) = H(z^8)H(z^4)H_c(z^2)H_c(z^5)$
6	$B_6(z) = H_c(z^8)H_c(z)P_{2c}(z)$
7	$B_7(z) = H(z^3)H_c(z) - (B_5(z) + B_6(z))$
8	$B_8(z) = H_c(z^3)P_2(z)$
where $P_1(z) = H(z^2)H(z)$ $P_2(z) = H(z^2)H_c(z)$	

TABLE I: Transfer function of each bands.

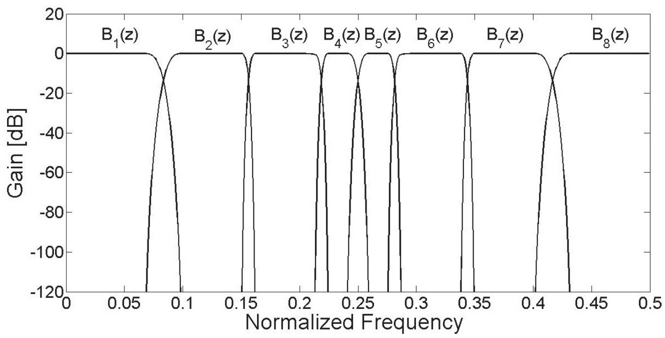

Fig. 6: Magnitude response of the proposed filter bank.

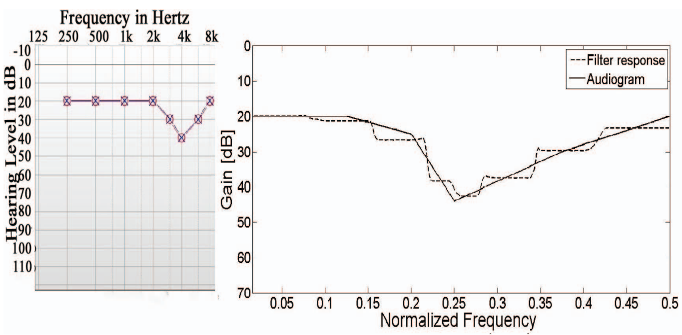

Fig. 7: Typical audiogram of NIHL and audiogram matching with 8-band filter bank [14].

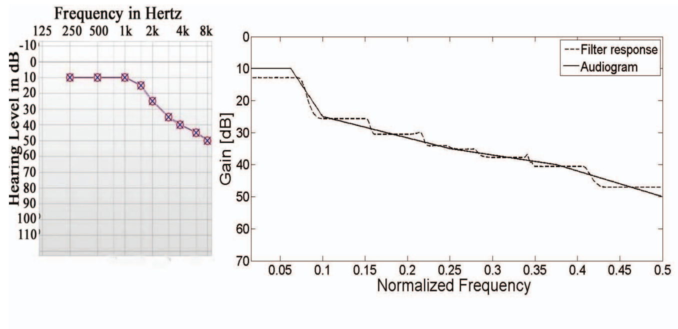

Fig. 8: Typical audiogram of presbycusis and audiogram matching with 8-band filter bank [14].

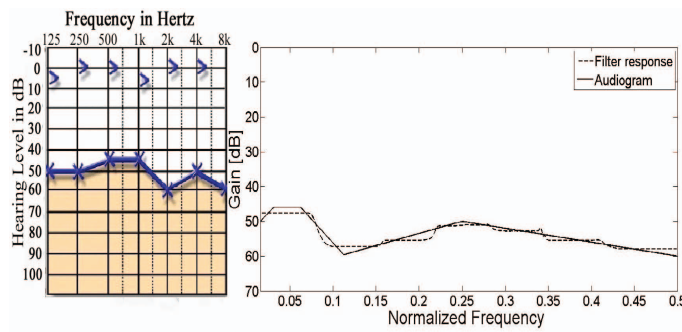

Fig. 9: Typical audiogram of bilateral conductive loss and audiogram matching with 8-band filter bank [15].

III. AUDIOGRAM MATCHING/PERFORMANCE EVALUATION/RESULT AND DISCUSSIONS

The complexity and matching error of the proposed filter bank largely depends on the length of prototype half-band filter. The prototype filter $H(z)$ determines the transition bandwidth of each subband. If the transition bandwidth is high the matching error will be high due to the overlap among different bands, especially in mid frequencies where the subbands are narrow. Further the matching between audiogram and magnitude response of the filter bank is closely related to transition bandwidth. Taking into considerations the maximum matching error and complexity, 0.05 is a reasonable normalised transition bandwidth. The selected specifications of the prototype filter for an eight band hearing aid with 0.05 normalised transition bandwidth is given below
Pass band frequency = 6.4 kHz.
Sampling frequency = 16 kHz.
Maximum pass band ripple = 0.0001 dB.
Minimum stop band attenuation = 120 dB.

To meet the above specifications, the required length of filter $H(z)$ is 47 and can be realized with 13 multipliers only. Magnitude response of the proposed filter bank is shown in Fig. 6. From Fig. 6 it is seen that narrower bands are allocated at mid frequency and subband filter bandwidth increases as it moves towards low and high frequency.

A. *Audiogram Matching*

In order to verify the effectiveness of the proposed filter bank for the hearing aid application, audiogram matching with different hearing loss pattern is simulated and verified. For presenting the results, the audiograms for common hearing loss types such as presbycusis, NIHL, bilateral conductive hearing loss are considered [14,15,16]. Fig. 7 shows the audiogram of NIHL and audiogram matching with proposed 8-band filter bank. Noise-induced hearing loss is the result of constant exposure to noise, of working in noisy environment, of frequent listening to loud music, etc. results in a larger loss of hearing in the area around the frequency of 4 kHz. Fig. 8 shows the audiogram of presbycusis and audiogram matching with proposed 8-band filter bank. Presbycusis is a type of hearing loss, which is generally seen in people above the age of 60. The hearing sensitivity has progressively worsened over the years, and this will be reflected in the audiogram especially in the higher frequencies. Fig. 9 shows the audiogram of bilateral conductive loss and audiogram matching with proposed 8-band filter bank. Conductive hearing loss occurs when there is a problem conducting sound waves anywhere along the route through the outer ear, eardrum, or middle ear. Fig. 10 shows the error plot of audiogram fitting for various hearing loss cases.

Proposed design can be realized with single prototype filter and six subfilters, which use 13 multipliers only. In existing

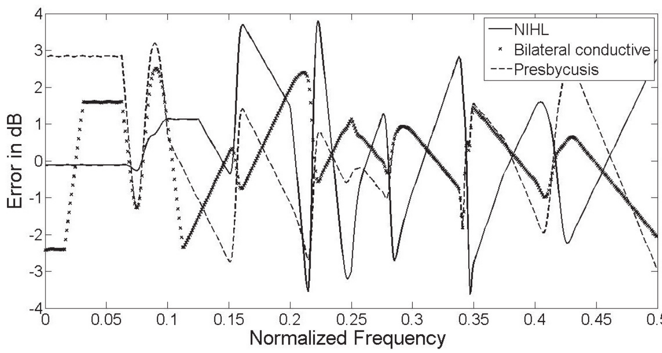

Fig. 10: Comparison of matching error.

method it require more than one prototype filter which need 10 subfilters and 15 multipliers for realization. Thus the complexity can be reduced. TABLE II shows the comparison of maximum matching error corresponding to each hearing loss cases. From the TABLE II, it can be seen that the proposed filter bank design is highly suitable for the audiogram with sharp variation of hearing loss at mid frequency regions compared to the existing method. Even though there is a small increase in maximum matching error for low and high frequency losses like bilateral conductive and presbycusis, the proposed filter bank gives very low complexity compared to the existing methods and can be used for different types of hearing loss. With optimal selection of the band edges of each band, the proposed method gives a better matching between audiograms and the magnitude responses of the filter bank.

Types of Hearing loss	Maximum matching error (dB)	
	Proposed system	Existing[2]
NIHL	3.8	12
Bilateral conductive	2.5	2.25
Presbycusis	3.1	2.75

TABLE II: Comparison of maximum matching error.

IV. FUTURE WORK

If the number of bands are increased in region where sharp transition of hearing loss occur good matching of audiogram can be achieved. The proposed filter bank design is mainly applicable for the audiogram with sharp variation of hearing loss at mid frequency regions. But sharp variation of hearing loss occurs at low and high frequency range, the proposed design may not be applicable. This type of hearing loss can be effectively compensated either by redesigning the filter bank structure for getting low bandwidth at positions of sharp variation of hearing loss or by introducing additional bands in the required range, then the maximum matching error can be reduced.

V. CONCLUSION

A low complex 8-band non-uniform linear phase digital FIR filter bank has been developed based on the frequency response masking technique for hearing aid applications. A single half-band FIR filter is taken as prototype filter. FRM technique is achieved by the repeated use of prototype filter and its interpolated filters. This reduces the number of multipliers to a minimum value with low audiogram matching error. Simulation results shows that, an 8-band filter bank can be implemented with 13 multipliers only and it gives a stop band attenuation of 120 dB with very low matching error. Various audiograms for the common types of hearing loss patterns are used for evaluating the effectiveness of the filter bank. Audiogram matching shows that, proposed method is good for sharp variation of hearing loss at mid frequency regions.

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