A DIGITAL HEARING AID THAT COMPENSATES LOUDNESS FOR SENSORINEURAL HEARING IMPAIRMENTS

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ABSTRACT

We introduce a digital hearing aid that compensate the signal spoken in sensorineural impaired listeners with object of improving their intelligibility. The technique used is based on a digital analysis/synthesis of speech, we divided the input signal into short time blocks then we make a multiband analysis, non linear amplification and synthesis basing us in a sinusoidal model of the voice, according to the subjet's dynamic range in that band. This system has been implemented in real-time using a DSP (TMS320C30) based microprocessor board within a host personal computer IBM PC, having been one of the principal objectives of the project to make an easy implementation VLSI system and low consumption for it to be a portable digital hearing aid

1. INTRODUCTION

Of the two major types of hearing loss we could highlight two, as the most important. Caused by degraded transmission of the acoustic energy to the cochlea or the sensorineural losses caused by a malfunction of the cochlea, the auditory nerve or both. On the first case, the hearing loss could be modeled by a linear form, thats why traditional hearing aid with a linear amplification work well.On the other hand, the subjects with sensorineural hearing loss has a narrow dynamic range due to the recruitment phenomenon for which a hearing aid with linear amplification, although could help the subjet, he continus to mantain an absence in the intelligibility of the speech, since the linear amplification may easily fall outside their dynamic range. The traditional hearing aids clips the signal that exceeds the gap of audition of the subject (clipping) which provokes that loss of intelligibility. For this reason, to adequately compensate sensorineural losses, the processing would necessary be nonlineal, varying in time, or both. We introduce in this paper a compensation method for the sensorineural impairment based on a digital analysis/synthesis speech as well as the tests carried out with 10 subjects.

2. PREVIOUS TECHNIQUES

These have been numerous investigators that have contributed solutions for the realization of hearing aid with a signal compression in order to improve the intelligibility of the subject, although in recent years multi-channel compression has received considerable attention. From between these solutions, we could cite Moore [1] who used a system of two analogical channels, carrying out a compression for each channel that relied on the hearing loss of the listener.Villchur [2] split the signal in three analogical channels utilizing a compression system in each one of them for the compensation of the recruitment.Fernández et al [5] proposed a system in that which the signal could be divided in bigger number of channels, in order to reduce the complexity of the system, Hidalgo et al [6] utilizing the new technical design of integrated circuits and temporal multiplexation composed all the channels in only one resulting in a circuit with a smallest complexity and consumption. Instead of the this analogical treatment of the speech, Engrbretson, A. M. et al. [3] introduced

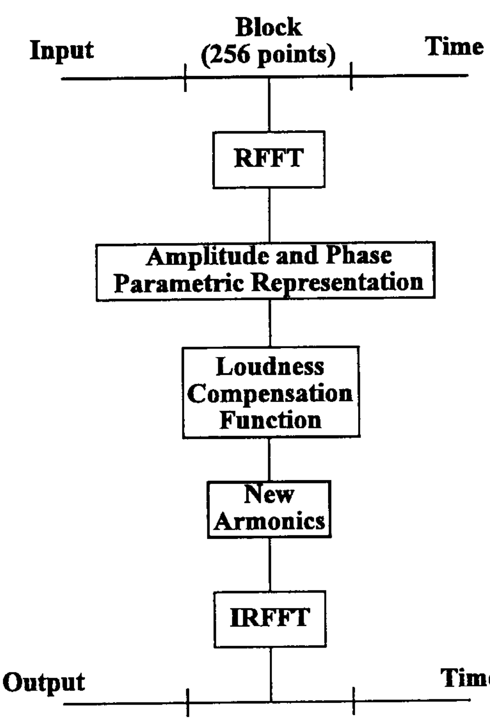

Fig. 1.- Block diagram of the proposed hearing aid system.

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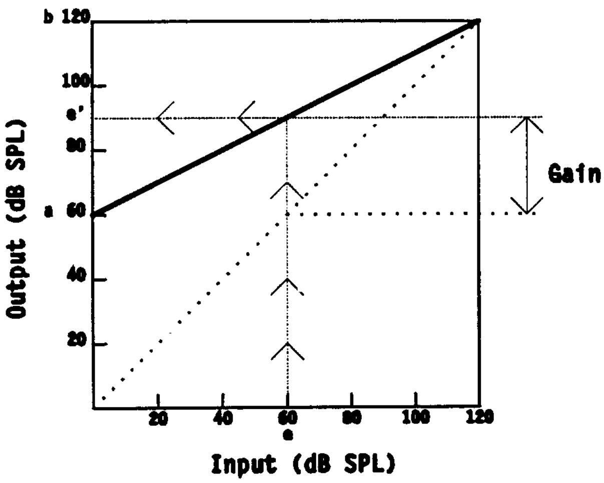

Fig. 2.- Loudness Compensation function. We suppose dinamic range of a normal listener from 0 to 120 dB.

a new digital hearing aid that splits the speech signal in four channels, although the treatment was continued doing in the time's domain, the same way other authors have utilized filterbanks in order to adapt the signal to the dynamic range of the subject [4][13].Later Asano et al [8] calculated the frecuency spectrum, from the measured frecuency spectrum calculated the hearing levels in six octave, then the optimum gain at each octave band is calculated using loudness compensation functions, finally, the input signal is processed with a digitalfilter using these gain characteristics.Genin et al [7] utilized a model in that which the loudness compensation function was totally in the frecuency's domain, influencing in sixteen different frecuency bands.Rutledge and Clements, M. A. [9] proposed to compensate for the effects of recruitments of loudness in sensorineural hearing impairments by solving a equations system generated for each frame of voice, the model of intercomponent masking is used to address the problem of adnormal spread of masking within speech components, in addition, the spectral relationships are maintained in terms of perceived loudness, this method has the problem that due to the great quantity of calculation was not possible to utilize it in real-time.Drake, L. A. et al [10] applied the analysis wavelet to compensate the effects of the recruitment

3. RECRUITMENT COMPENSATION MODEL

The Techniques proposed to compensate for the effects of recruitment of loudness in sensorineural hearing impairments exploits the espectral representation obtained by means of Fast Fourier Transform (FFT) attempting to reconstruct the signal spectral into the impaired listener's dynamic range by means of a loudness compensation function.

Figure 1 shows a block diagram of the proposed system on that which we could see the consecutive process until we obtain an output signal from the input signal

In order to get the samples of the speech signal we utilized at A/D converter whose sampling frequency is 10.4 KHz with a 4.7 KHz antialiasing filter.The input signal is divided into short-time block of 256 samples . Only 206 samples will be new ones, because the others were obtained by overlapping frames. From these frames the frecuency spectrum is calculated by using 256-point short-time Real-valued Fast Fourier Transform (RFFT) [11]. Doing an exponentials representation of the frecuency components, for each frecuency component w_k we obtain

$$S_k = A_k\, e^{j\theta_k} \qquad (1)$$

where A_k is the amplitude and θ_k the phase. Our objective will be to obtain a new speech signal that the frecuency components are within the dynamic range of the impeired listener. For that we will utilize all the harmonics available S_k modifying the parameter amplitude (A_k) by means of loudness compensation function of compensation shown in the figure 2, where we could get

$$\frac{x - P_n}{D_n - x} = \frac{y - P_p}{D_p - y} \Rightarrow y = K_1 x + K_2 \qquad (2)$$

being x the amplitude in dB of the corresponding harmonic and y the amplitude of the same modified harmonic.From (2) we can obtain a function with the modified parameter amplitude (A_k)

$$A_k^0 = K_4\, (A_k^i)^{K_1} \qquad (3)$$

being A_k^i the initial amplitud of the frecuency componet, A_k^o is the compensate amplitude and K_1 and K_4 two constants derived from the subject's threshold and discomfort level. Once modified the amplitude of the harmonics he obtain the new frecuency component keeping the original phase and we synthesize the new signal by means of Inverse Real-value Fast Fourier Transform (IRFFT)

Figure 3 shows an example of the processed signal. The threshold and disconfort levels of the impaired listener that were used to the system characteristics are also show in the figure. The frecuency spectrum of the input signal was projected into dynamic range of the impaired listener, even if the frecuency spectrum of the input signal varied. Furthemore, the fine structure of the spectrum remained intact after processing.

4. REAL-TIME SYSTEM

These digital hearing aid consists in a board (Texas Instruments TMDS 3260030) equipped with a single 32-bit floating-point DSP chip (Texas Instruments TMS320C30), 14 bit A/D and D/A and a host personal computer. Sampling frecuency is 10.4 KHz and a antialiasing filter of 4.7 KHz. The system should give result every 19.4 ms, because it is the length of the new block, which is perfectly possible if we see the table I in that which we show the time required for each one of the operations of the system.

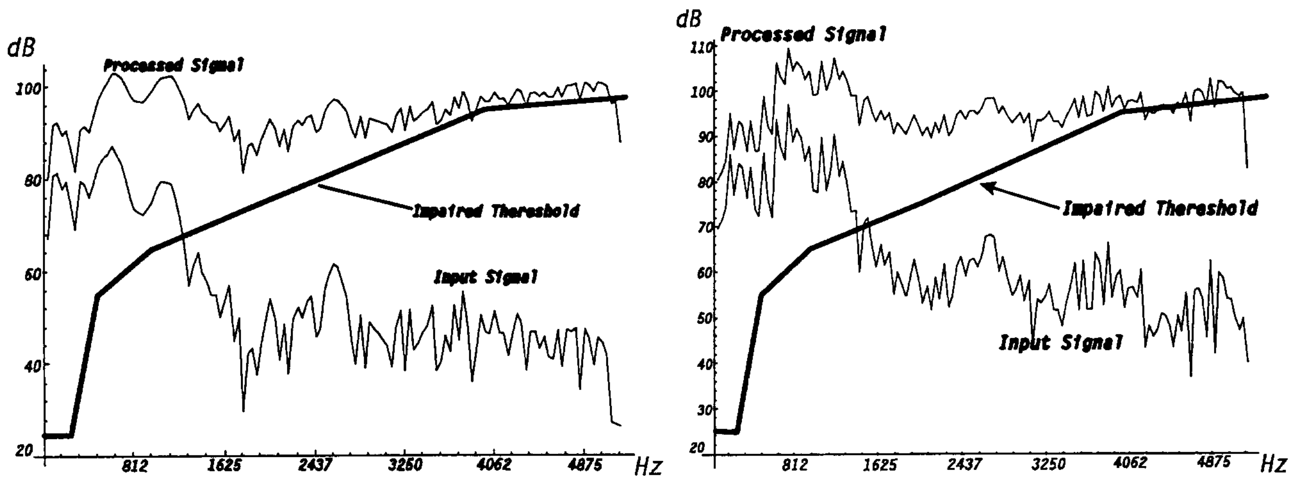

Fig. 3.- An example of a processed signal using the proposed hearing aid system. The input signals were (a) the consonant portion and (b) the vowel portion of spanish word /PA/.

5. EVALUATION EXPERIMENT

For the experiment the hearing aid we have utilized 10 patient diagnoses with Hypoacusia Sensorineural, or Mixed Hipoacusia with a perception component, the only common feature between them is, there has been no classification by age, audiogram, intensity of the hearing loss neither for the speech audiogram.The utilized equipment was a audiometer clinical AC-30, gauged for the aerial conduction according to the norm ISO 386-1992 and for the bony conduction according to the norm ISO 7566, a reproducer CD-ROM and a CD-ROM with the list of words of María Rosa Cárdenas and Victoria Marrero [14], composed for lists of 25 words, characterized by their phonic equilibrium, the proportion of phonemes and syllabic structures in each one of them correspond to that of the spoken Spanish and is utilized in order to quantify the level of auditive discrimination of the impaired listener (capacity in order to recognize and differentiate the language phonemes). The utilized methodology was to locate the dynamic range, subject's threshold and discomfort level by means of conventional tonal test, determination of the confortability threshold for a 50 dB signal and localization of the percentage of intelligibility and the intellibiligity threshold by means of vocal test carried out in three cases: 1) without hearing aid; 2) by the subject's hearing aid; 3) using the proposed hearing aid system.They resulted obtained for the proposed system and for vocal test is shown at the table I. Figure 4 show the vocal test for a sensorineural hearing impairment with the proposed system and the comparasion with a hearing aid use by the subject. The processed signal was monaurally presented using at headphone (Telefonics TDH39 of 10)

6. CONCLUSIONS

We have introduced a digital hearing aid system that compensate the speech signals. This system projects the frecuency spectrum of the input signal into the subject's dynamic range. The obtained results at the evaluation experiment improve a lot the subject's intelligibility with regard to the systems

Input Level [dB]	Subjects									
	S1 (5)	S2 (9)	S3 (15)	S4 (24)	S5 (29)	S6 (38)	S7 (54)	S8 (68)	S9 (69)	S10 (82)
50	-(-)	-(-)	-(-)	10(-)	60(-)	60(-)	-(-)	70(-)	-(-)	-(-)
60	40(-)	50(-)	50(-)	80(-)	80(10)	80(-)	30(-)	80(20)	60(-)	60(-)
70	60(-)	80(-)	60(10)	90(-)	70(40)	100(10)	60(-)	100(40)	70(-)	80(-)
80	70(30)	90(-)	80(20)	100(-)	50(60)	100(50)	90(-)	100(70)	90(10)	90(30)
90	40(50)	*(40)	90(80)	85(-)	*(40)	*(100)	100(-)	*(70)	100(40)	*(40)
100	*(50)	*(70)	70(70)	80(40)	*(*)	*(100)	*(50)	*(30)	100(80)	*(40)
110	*(40)	*(60)	*(30)	*(80)	*(*)	*(*)	*(80)	*(*)	*(100)	*(80)

TABLA I.- The score of vocal test for the proposed aid system. The scores for the vocal test of the subjects are also shown in parentheses. '-' indicates that the subject was not respose. '*' indicates that the test was not conducted.

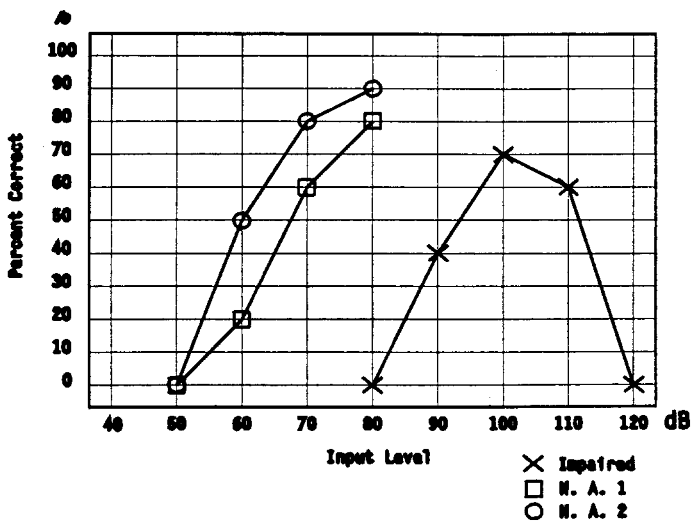

Fig. 4.- Show the vocal test results for a Sensorineural hearing Impairment. H.A.1 and H.A.2 are the subject's hearing aid and the proposed system respectively.

introduced by other authors, although in many cases it is not possible to arrive to the 100% since the possibility exists of confusing certain phonemes with great phonic vicinity , like /p/ and /t/ or /m/ and /n/, it would be solved with a small rehabilitation guided to distinguish well those phonemes.We also observe that the subject felt more comfortable with this digital hearing system and that they understood words resulted much more clearer. For all of this we consider that the digital hearing aid system here introduced offer a big possibility to the sensorineural impaired listeners although we also think that it still is necessary to continue investigating and evaluating the behavior of this hearing aid in other more adverse conditions.

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