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MULTIMEDIA APPLICATIONS FOR THE HEARING IMPAIRED

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Hearing impairment is one of the fastest growing diseases of modern society. Therefore it is very important to develop new methods for diagnosis and therapy of hearing disorders. Some of them were introduced to practice as a result of a co-operation between institutions mentioned in the header. The system for mass-scale hearing screening is one of multimedia programs for testing communication senses introduced by the authors. The further developments include among others an application of dithering theory to practical solutions for tinnitus patients and a method of fitting hearing aids employing soft computing. The implemented hearing diagnostic & therapy applications and systems with their underlying concepts are reviewed in this paper.

Keywords: hearing screening, tinnitus, hearing aids fitting.

1. Introduction

The epidemiological studies, conducted systematically in Poland since 1999 by the Institute of Physiology and Pathology of Hearing have shown, that one child out of five, in the age group 6-19 years, has hearing problems. Hearing disturbances have a very distinct influence on the child's learning progress, emotional and intellectual development. As early detection of hearing disorders may result in better treatment effects, it is highly important to detect these problems in early age of children. Being aware of these facts, an Internet-based system was devised to provide a popular, cheap and easy method for screening testing of hearing, mainly intended for schoolchildren.

Advances in teleinformatics as well as its mass employment in the recent years opened new possibilities of conducting mass screening of hearing, tinnitus (ear noises), speech and vision. Diagnostic and recovery systems associated with the interactive medical portal Telezdrowie (www.telezdrowie.com) designed by the institutions mentioned in the header of this paper serve as an example, how relatively simple diagnostic methods employed in screening tests can be mass-deployed thanks to teleinformatics, defining new quality of widespread diagnostic tests of communication senses [3, 6].

Moreover, on the basis of digital signal processing we proposed a theoretical explaining to the method of eliminating tinnitus originating from threshold quantisation, basing on the dithering technique. This technique involves supplementing low-level useful signals with noise of certain level, which has the effect of stopping the process of spontaneous noise generation in the human auditory system resulting from the threshold characteristics. The findings also formed a base for some multimedia applications developed to assist tinnitus patients [7, 8, 11].

Furthermore, the paper presents a general principle of operation of the designed system for fitting hearing aids as well as the results of experiments utilizing implementations of this system [4, 6, 14].

Thus, this paper provides a short survey of some selected topics from our common research & implementation works devoted to multimedia services for those who may have hearing problems. We choose these topics because we believe that they are related in many aspects to Professor's Andrzej Rakowski research interest whose 50 years of research activity is celebrated with this special issue of *Archives of Acoustics*. In this connection, his broad research interests include among others problems related to psychoacoustics and hearing impairments.

2. Hearing screening employing personal computers

Awareness of hearing problems in society is not common. Therefore, we conceived a system intended to provide easy access to information about hearing problems, ways of treatment and specialist's contact information. It is important in this context to state that "*I can hear...*" system (www.telezdrowie.pl) launched by our institutions some years ago is not just another Internet application. Several medical consultation centres were also started in Poland which were prepared to receive "Internet patients" and take care of them. This relatively simple concept pioneered telemedicine in Poland and as indicate numerous international prizes awarded to above solutions, it was considered innovative not only in our country.

The elaboration of "I Can Hear..." system was also motivated by scientific goals. Since this system is based on the Internet, it can be used in a straightforward way for statistical data acquisition by gathering information about hearing problems within a tested population. These data make an excellent source of information for scientific research purposes, giving a better insight into statistical distribution of hearing problems among children and youth. Obtained results may be exploited in health care prevention programs design.

Signal generating, analyzing and processing play a crucial role in diagnosing and treating hearing losses. This fact is associated, besides other factors, with the methodology of audiometric measurements and with supporting hearing with hearing aids and cochlear implants. The main advantage of speech audiometry is that, unlike tonal audiometry, it allows for measuring not only receptive or conductive hearing, but also for assessment of the whole complex hearing-perception mechanism. That is why speech audiometry was utilised in screening children and teenagers in schools. Two main types of thresholds are associated with speech audiometry. The first one is *Speech Detection Threshold – SDT*. It is defined as the lowest signal level, for which speech is heard over 50% of the time. The second threshold type is *Speech Reception Threshold – SRT*, defined as the lowest signal level, for which speech is understood during 50% of the time.

When speech audiometry tests are performed using noise masking the investigated ear, we speak about so called speech-in-noise audiometry. The masking noise makes speech understanding harder, in this case on purpose. This effect can be seen on audiograms in the shape of classical noiseless speech audiograms being "shifted" right by a value equal to the masking noise level (Fig. 1). Widely used measurement methods are based only on three-tone tonal audiometry. Within the presented project a relatively new method, based on additional use of speech-in-noise audiometry, was used extensively. Besides simple tones this method employs speech signal mixed with noise in appropriate proportions. This method in combination with the three-tone test and with a set of electronic questionnaires forms a basis for a system of mass hearing screening, which has so far been used to test above 500,000 of school pupils. These results were published also earlier, e. g. [3, 6].

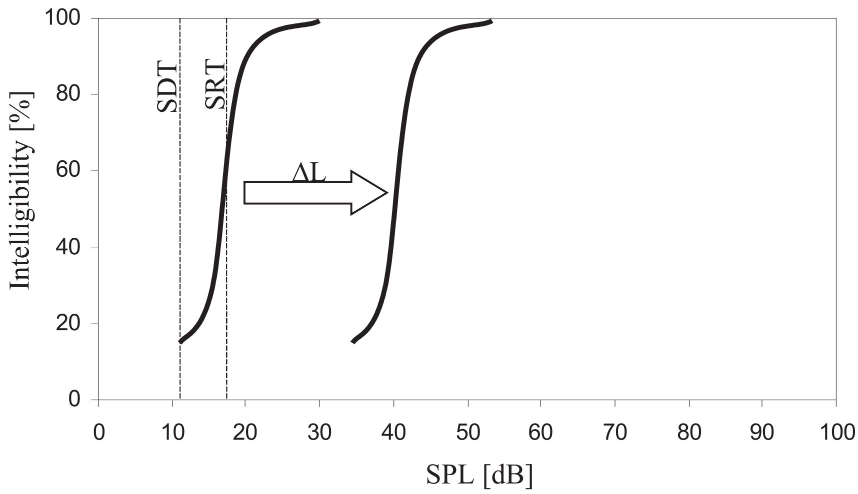

Fig. 1. Speech audiogram shifted due to influence of masking noise ΔL.

3. Digital technology in diagnosing and treating tinnitus

Tinnitus (ear noises) often appears in cases of elevated hearing threshold associated with hearing loss due to inner ear diseases. Such condition may be caused by degeneration of external hair cells, causing neurons to be activated by signals of levels higher than normal. In such a case a threshold system of elevated activation threshold is present. However, before such elevated threshold appeared for the given patient e.g. due to suffered disease or development of otosclerosis, signals were received and interpreted as hearing stimuli at higher levels of the hearing track.

According to our findings this fact results in introduction of an additional mechanism of threshold quantisation of weak acoustic stimuli, which in turn results from the elevation of activation threshold of nerve cells. The existing theories in audiology attempting to explain this phenomenon do not directly take into account the mechanisms of signal quantisation occurring due to presence of threshold characteristics in the transmission system. Such interpretation becomes possible only after using the knowledge of electric signal processing developed in other fields of science, e.g. in digital signal processing. Using this approach, we proposed a new interpretation of the phenomenon of tinnitus generation on the basis of the theory of audio signal quantisation [2, 3, 9]. Moreover, on the basis of digital signal processing we proposed a theoretical explaining to the method of eliminating tinnitus originating from threshold quantisation, basing on the dithering technique. Generally speaking, this technique involves supplementing low-level useful signals with noise of certain level, which has the effect of stopping the process of spontaneous noise generation in the hearing track resulting from the threshold characteristics. One can easily notice that a corresponding approach is used to reduce tinnitus in audiology practice, where masking noise generated by special masker devices is utilised. The widely known effectiveness of such techniques of reducing both ear noises (in audiology) and quantisation noises generated spontaneously in electronic circuits (in digital signal processing) indicates that ear noises are justified to be interpreted as a direct consequence of quantisation of weak signals in threshold systems. It may be useful at this point to present several aspects of analysis of quantisation phenomena. They will be further used to show, how the interpretation of the phenomenon of noise generation in quantising systems and its elimination with additional, "masking" dither noise may be useful for explaining the phenomena observed in the case of ear noises [6, 7, 9].

As is widely known, typical transition functions of a quantizer are described by the following formulae:

$$Q(x) = \Delta \left[\frac{x}{\Delta} + \frac{1}{2} \right] \tag{1}$$

or

$$f(x) = \Delta \left[\frac{x}{\Delta} \right] + \frac{\Delta}{2}, \tag{2}$$

where x – value of the sample before quantisation (at input), Δ – height of quantisation step, [ ] – operator returning the integer nearest to the given real number.

In the case of complex input signals of high amplitudes successive errors are uncorrelated and therefore the power density spectrum is similar in character to that of white noise. Error signal is also uncorrelated with the input signal. Distribution of error probability density for a quantizer of transition function described by formula (6) is a rectangular window function:

$$p_\delta(x) = \begin{cases} \dfrac{1}{\Delta} & \text{for} \quad |x| \le \dfrac{\Delta}{2}, \\ 0 & \text{for} \quad |x| > \dfrac{\Delta}{2}. \end{cases} \tag{3}$$

For complex input signals the maximum error is equal to the least significant byte (LSB) and, provided that approximation is good, samples of quantisation error δ_n can be considered independent of the input signal. For input signal of such type uniform quantisation can be easily modelled by adding white noise to the input signal. However, for input signals of low level the model of additive white noise is no longer valid. In such a case the error becomes significantly dependent on the input signal. Signals from the range $(-\Delta/2, \Delta/2)$ are ascribed zero value by the converter and therefore are not conducted along the track; this effect is known as "digital deafness". In such case there is no signal at the output and the error is equal to the input signal, but has the opposite sign. This type of error is noticeable by ear and therefore is a disadvantageous phenomenon associated with quantisation.

The dither technique is aimed at modifying statistical values of total error. In quantising systems which do not use the dither technique the instantaneous error is a defined function of the input signal. If the input signal is simple and comparable in altitude to the quantisation step, the error is strongly dependent on the input signal and introduces audible distortion and modulation noise. Use of the dither signal of appropriately shaped statistical properties may cause the audible distortions to be similar in character to stationary white noise.

Today's digital audio tracks use the dither technique utilising noise of triangular function of probability density and peak-to-peak value of 2 LSB. The dither noise is therefore an additive noise introduced into the signal, usually before the quantiser. The averaged response obtained at the conversion system output is the following function of input signal:

$$\overline{y}(x) = \int\limits_{-\infty}^{\infty} y(x+\upsilon)p_\upsilon(\upsilon)\mathrm{d}\upsilon, \tag{4}$$

where $p_\upsilon(\upsilon)$ is the probability distribution density of the noise, in the case of noise of rectangular distribution defined as:

$$p_\upsilon(\upsilon) = \begin{cases} 1/\xi, & \text{for} \quad |\upsilon| \le \xi/2, \\ \qquad 0 & \qquad\qquad , \end{cases} \tag{5}$$

where ξ is the peak-to-peak amplitude of the dither noise.

Figure 2 illustrates the fundamental phenomena taking place, when a signal of amplitude comparable to quantisation threshold is fed onto the A/C converter input.

Figure 3a shows the result of quantisation of a sinusoidal signal obtained without introducing the dither noise, while Fig. 3b shows the result of quantisation of the same

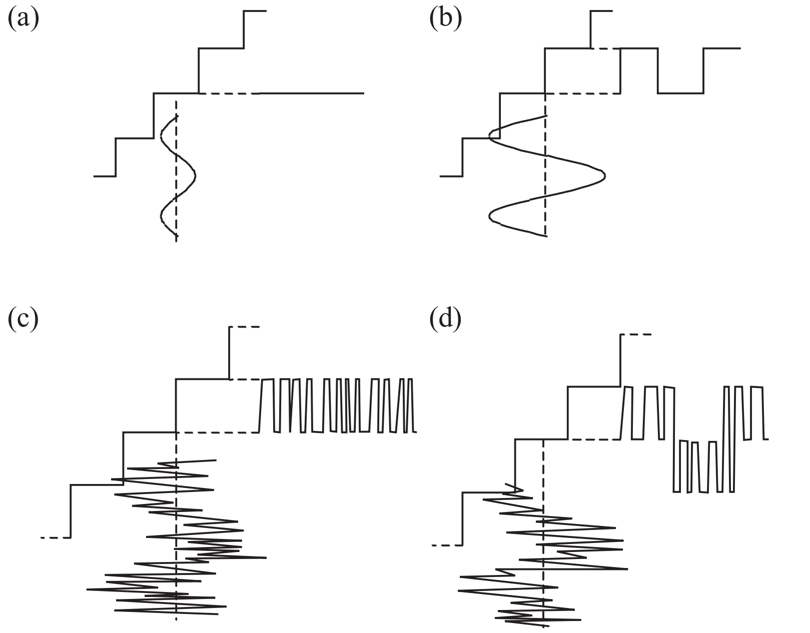

Fig. 2. Effects associated with quantisation of small amplitudes and the influence of the dither noise: (a) "digital deafness"; (b) "binary quantisation"; (c) dither removes the insensitiveness range of the converter; (d) "response blurring" in the case of binary quantisation.

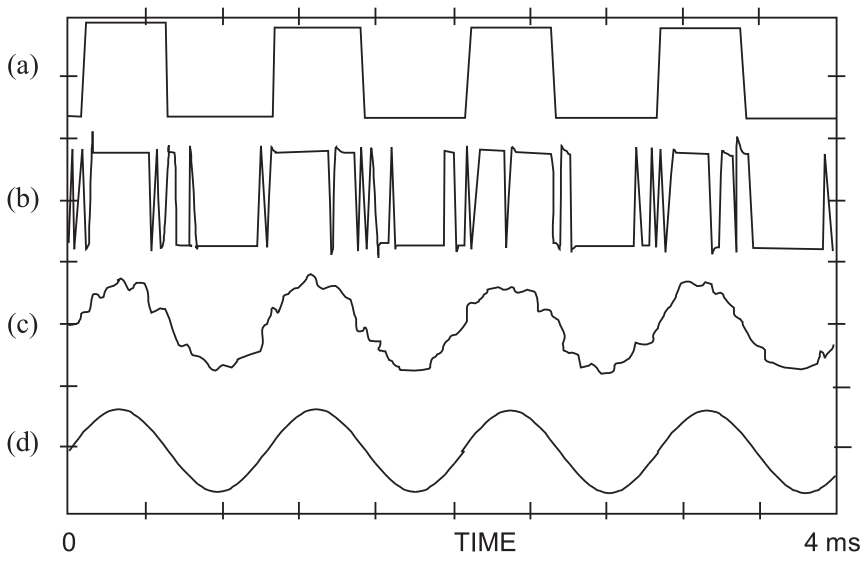

Fig. 3. Effects of quantisation of waveform of amplitude corresponding to quantisation step: (a) harmonic signal directly after quantisation; (b) quantisation using the dither noise; (c) signal from the previous plot averaged over 32 periods; (d) result of averaging over 960 periods.

signal in presence of the dither signal. The same figure (parts c and d) proves that averaging the representation (b) can lead to almost perfect reproduction of the original waveform that was subject to quantisation. One shall note that, as the sense of hearing possesses noticeable integrating properties, similar processes can certainly take place in the hearing track. It is important to observe that the case depicted in Fig. 3b may correspond to the case of spontaneous emission of signals in result of occasional crossing the quantisation threshold level by random acoustical signal (e.g. background noise components).

Noise power at output for static input signal can be defined as:

$$P_n^2(x) = \int_{-\infty}^{\infty} [y(x+\upsilon) - \overline{y}(x)]^2 p_\upsilon(\upsilon)\mathrm{d}\upsilon. \tag{6}$$

In the case of the dither noise of Gaussian distribution defined as:

$$p_\upsilon(\upsilon) = \frac{1}{\sqrt{2\pi}\sigma_\upsilon} \exp\left(\frac{-\upsilon^2}{2\sigma_\upsilon^2}\right), \tag{7}$$

$$\overline{\nu} = \sum_k \nu_k p_\nu[\nu_k], \tag{8}$$

$$\sigma_\nu^2 = \sum_k (\nu_k - \overline{\nu})^2 p_\nu[\nu_k]. \tag{9}$$

Introducing the "masking" dither noise at a proper power level lets us obtain the desired results of eliminating the low-level signals quantisation effects and minimising distortions occurring for very low amplitudes of the signal being quantised. Audibility of the introduced noise can be decreased by preliminary shaping its spectrum, so that noise energy rises towards higher frequencies. The same principles remain valid for masking tinnitus, due to visible similarity of phenomena occurring in electronic and biological signal-transmission systems.

Possibilities resulting from the above statements were employed practically in the process of creating a Web-based application for people suffering from ear noises, which is accessible at www.telezdrowie.pl (or www.telewelfare.com). This application includes a system of electronic questionnaires and a database of signals useful for masking ear noises. It allows a patient to diagnose his own ear noises using a computer. Next, after consulting a doctor, the patient can download a masking noise appropriate for him and store it in a portable mp3 player, which allows for an effective treatment by tinnitus masking. Recently, we extended this concept to the range of ultrasound frequencies [7]. It enabled us to propose and to design the ultrasound tinnitus masker device.

An application of the ultrasound transducer in this device (see Fig. 4) enables to obtain advantageous characteristics of the masker because [5]:

1. It is possible to employ bone conduction for the transmission of masking noise.
2. It is possible to increase the noise level without making it audible and invoking hearing tiredness.
3. A ultrasound converter of small dimensions makes a further significant miniaturization possible.

The experimental results obtained employing above methods were presented in some of our recent papers [7, 8].

a) b)

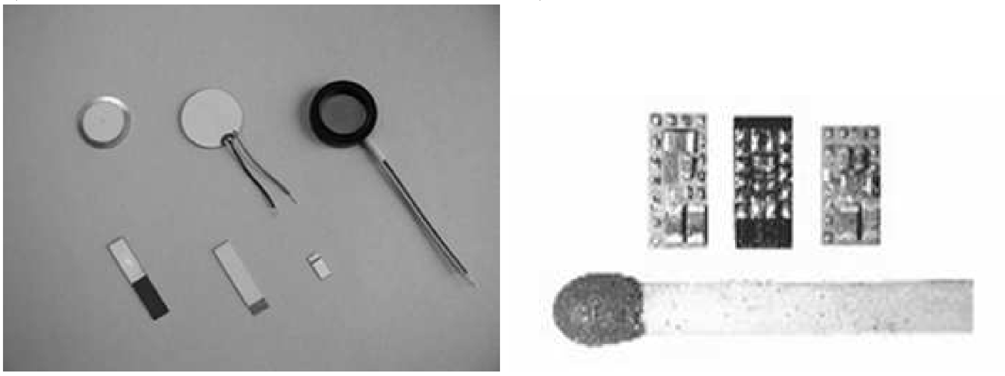

Fig. 4. (a) Piezoelectric transducers utilized in various versions of ultrasound bone stimulator; (b) highly miniaturized signal processors used to generate dither noise at 30 kHz frequency (b).

4. Method of computer-aided adjustment of hearing aids

Widely-used systems of adjusting hearing aids allow for setting parameters of a hearing aid, which usually does not mean however that the optimum characteristics of speech perception in hearing impaired is obtained. As in a general case the problem of adjusting a hearing aid can be approximately brought down to the problem of adjusting the wide dynamics of speech to the narrowed dynamics of impaired hearing, we and our Ph. D. student [14] focused mainly on the problem of determining the characteristics of impaired hearing and then on deriving the dynamics of the sought hearing aid that would compensate for the damage. This is certain simplification, because in a real hearing aid the quality of hearing loss correction is also influenced by other factors, such as: noise reduction algorithms, equalizing of the processed signal or acoustic elements of the hearing aid [1, 12, 13].

Properties of hearing dynamics can be determined on the basis of results of a loudness scaling test. The designed system includes an implementation of the loudness scaling test based on an algorithm for evaluating sensation of Loudness Growth in half-Octave Bands (LGOB) [10]. The test utilizes signals in the form of white noise samples filtered in half-octave bands of central frequencies of 0.5 kHz, 1 kHz, 2 kHz and 4 kHz. A person being diagnosed during the test is exposed to randomly presented test signals of various sound levels. The person being diagnosed evaluates the loudness sensation using a seven-degree scale of categories of loudness sensation represented by descriptive labels (from "nothing is heard" to "very loud"). This is a standard procedure serving as an initial step to further processing of results obtained in this way, because the developed system utilizes a special module for converting results of loudness scaling to hearing dynamics representation. This module utilizes fuzzy processing to map the results of scaling from categories of loudness sensation evaluation into the objective scale of input sound level expressed in dB. The category scale is represented by 7 fuzzy sets described by membership functions. These membership functions were derived from statistical analysis of loudness scaling for 20 regular-hearing persons. The output of the fuzzy logic system is described by 13 membership functions representing the difference between the given result of scaling and the results for regular hearing [5].

As there are four frequency bands of centre frequencies of 500 Hz, 1000 Hz, 2000 Hz and 4000 Hz tested in the LGOB test, four sets of membership functions are required. For determining such membership functions it was needed to perform the LGOB test on several dozens of regular-hearing persons. At the stage of method designing we prepared the required membership functions on the basis of generally accepted approximation of LGOB test results for regular hearing.

One of the basic methods of approximation that come to mind after analyzing typical fuzzy logic systems and the set of theoretical membership functions (as in Fig. 5) is approximation of result-containing boundary of the set with triangle-shaped functions. It can be performed e.g. on the basis of mean-square approximation. One should determine the equations of two straight lines approximating the triangle sides in the mean-square sense for this purpose. The algorithm used by the authors to determine the triangular membership functions involves the following steps [5]:

- Finding the value of the first element belonging to the given fuzzy set (value of the first argument, for which the factual membership function takes a non-zero value).
- For determining the position of the first arm of the triangle one considers all the elements of the membership function MF fulfilling equation:

$$x : \left\langle \underset{x_i}{\forall} \left(\mathrm{MF}\left(x_i\right) - \mathrm{MF}\left(x_{i-1}\right)\right) > 0 \right\rangle, \tag{10}$$

where i – subsequent indices of arguments of membership functions MF fulfilling condition (10).

- Calculating coefficients a_1 and b_1 of the straight line equation $y = a_1 x + b_1$
- For determining the position of the second arm of the triangle one considers all the elements of the factual membership function MF fulfilling equation:

$$x : \left\langle \underset{x_i}{\forall} \left(\mathrm{MF}\left(x_i\right) - \mathrm{MF}\left(x_{i-1}\right)\right) <= 0 \right\rangle, \tag{11}$$

where i – subsequent indices of arguments of membership functions MF fulfilling condition (11).

- Calculating coefficients a_2 and b_2 of the straight line equation $y = a_2 x + b_2$.
- Calculating the point of intersection of the straight lines $y = a_1 x + b_1$ and $y = a_2 x + b_2$.
- Calculating zeros of both lines.

An example of a set of membership functions for the frequency band of 500 Hz obtained by approximating the real values is illustrated in Fig. 6.

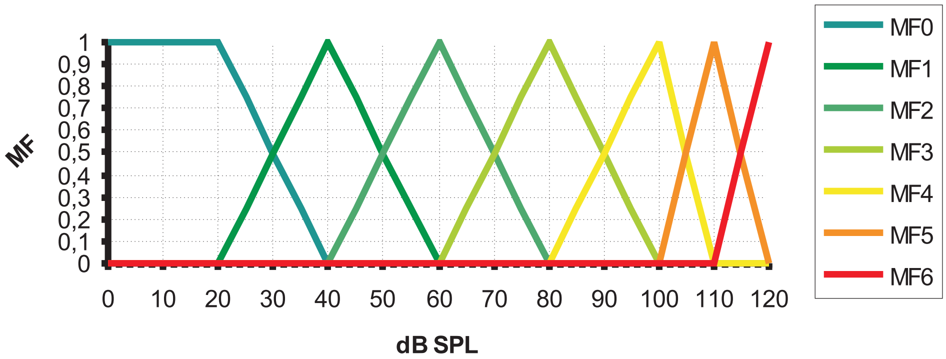

Fig. 5. Set of membership functions describing loudness sensation for regular hearing, derived on the basis of theoretical data.

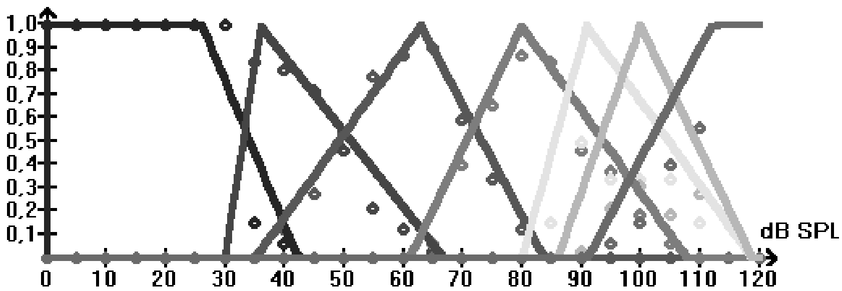

Fig. 6. Approximation of values of membership functions with triangles performed on the basis of real data.

The next step of the described method involves defining the system output. As the aim of the designed method is determining the dynamics of impaired hearing, the designed system should compute the difference among the given loudness sensation evaluation and the correct loudness sensation evaluation corresponding to the given test signal. This difference should be expressed in dB.

An analysis of a typical plot of LGOB test results reveals that between seven categories of loudness sensation evaluation one can define six differences pointing to hearing loss (area below the LGOB curve for regular hearing) and six differences pointing to hypersensitivity (area above the LGOB curve for regular hearing). No difference is a special case of difference. The above analysis leads to a conclusion that the output of the described fuzzy system can be described by a set of thirteen membership functions (Fig. 7) expressing the difference between the factual loudness sensation evaluation and the evaluation for regular hearing. Fuzzy sets obtained in this manner can be described with the following labels (describing the difference size): none, very small, very small+, small, small+, medium, medium+, big, big+, very big, very big+, total, total+. Labels marked with "+" sign denote positive difference (hypersensitivity).

A fuzzy processing depends on properly defined rule basis. Fuzzy logic rules have the following shape:

If <*premise1*> **AND** <*premise2*>. . . **AND** <*premise_n*> **THEN** *decision*.

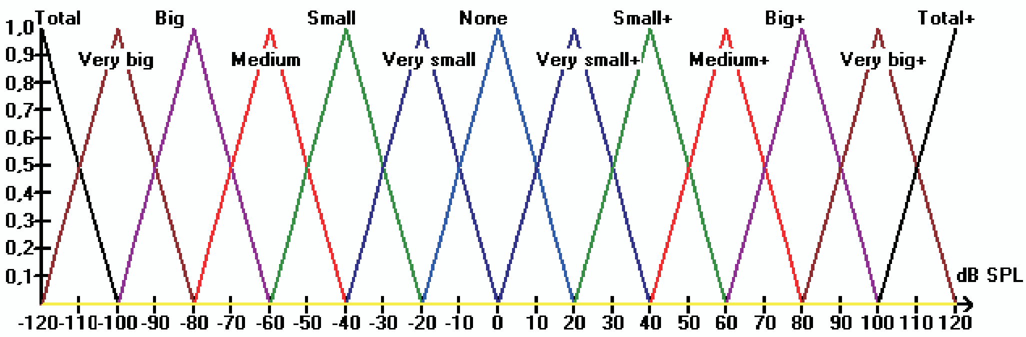

Fig. 7. Initial membership functions describing the difference between regular and abnormal loudness scaling.

In the discussed case there are two premises. The first one is associated with information on regular loudness scaling. The other premise is associated with the investigated results of LGOB test. Since both premises apply to the results of the LGOB test, they both use the same categories describing loudness sensations. In order to differentiate the fuzzy sets associated with individual premises, labels of fuzzy sets associated with the first premise use lower case letters while those of fuzzy sets associated with the second premise utilize upper case letters.

Generally the rule base is designed on the basis of audiologists' expertise. In this case such expertise can be derived from the analysis of the LGOB test. The analysis of LGOB test results for regular-hearing persons shows that the loudness scaling is linear in character, however the factor of proportionality rises from 1:1 to 2:1 (the loudness sensation rises twice faster) for test signals of level exceeding 100 dB SPL. On the basis of above information a rule base was prepared. It is presented in Table 1. Categories associated with regular hearing are marked with lower case (e.g. "loud"), while categories marked with upper case (e.g. "LOUD") are associated with hearing impairment.

The last stage of the system's operation is defuzzyfying, i.e. conversion of the obtained categories to a numerical value. After performing this process (using the centre of gravity method) one obtains the characteristics relating the sound levels expressed in dB to the expected subjective assessment of loudness sensation for the given patient; this is the sought characteristics of hearing dynamics.

In order to determine the whole dynamics characteristics of investigated hearing we had to create an algorithm which would calculate the desired hearing dynamics characteristics on the basis of LGOB test results using the designed method of determining the difference between regular and impaired loudness scaling.

A fundamental advantage of the designed method is the mechanism of automated mapping from the category scale to the scale of sound levels expressed in dB. Moreover, the designed method of determining the hearing dynamics utilizes full information on

Table 1. Rule base for the fuzzy system (the decision attribute is the difference between regular perception and perception taking place in cases of hearing impairment.

	I DON'T HEAR	VERY SOFT	SOFT	MCL	LOUD	VERY LOUD	TOO LOUD
I don't hear	None	V.small+	Small+	Medium+	Big+	V.Big+	Total+
very soft	V.small	None	V.small+	Small+	Medium+	Big+	V.Big+
soft	Small	V.small	None	V.Small+	Small+	Medium+	Big+
mcl	Medium	Small	V.Small	None	V.small+	Small+	Medium+
loud	Big	Medium	Small	V.small	None	V.small+	Small+
very loud	V.Big	Big	Medium	Small	V.small	None	V.small+
too loud	Total	V.big	Big	Medium	Small	V.small	None

regular loudness scaling in LGOB test, while the standard methods of adjusting hearing aids utilise averaged data only. In the course of the standard determination of loudness scaling (LGOB test) the diagnosed person is presented multiple times with filtered noise samples and the obtained results are subsequently processed statistically, while the designed method requires only presenting the diagnosed person with filtered noise samples corresponding to seven loudness levels in four frequency ranges. The resulting difference in test-related effort and duration is of fundamental importance for audiology practice.

Figure 8 presents an example plot of loudness scaling results with the LGOB test, which were obtained for test signals from the frequency band centered around the frequency of 500 Hz.

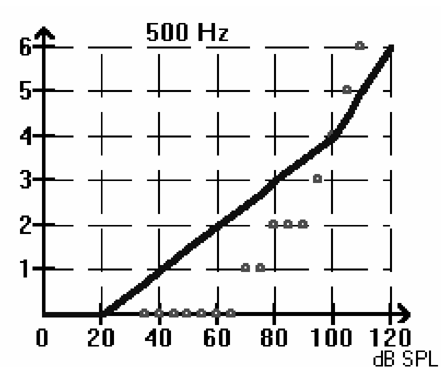

Fig. 8. Sample results of LGOB test (category scale on Y-axis).

The derived dynamics characteristics of impaired hearing can be additionally used by the presented system for approximate simulation of the hearing loss. In order to derive the sought dynamics characteristics of hearing aid, the system compensates the dynamics characteristics of impaired hearing. This compensation is based on flipping the hearing dynamics characteristics by the $y = x$ straight line (Fig. 9).

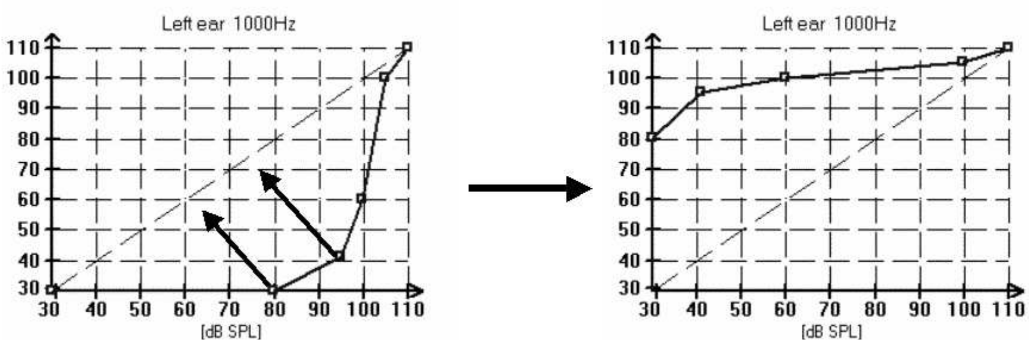

Fig. 9. Compensating the dynamics characteristics of impaired hearing (sound level on Y-axis).

Obtained degrees of speech understanding for individual listening tests are presented in Table 2 [14]. The degree of speech understanding is expressed in percent. The first value denotes the degree of speech understanding without dynamics processing, while the second (after the arrow) denotes the degree of speech understanding after dynamics processing based on the obtained dynamics characteristics of hearing aid.

Table 2. Averaged degrees of speech understanding for patients suffering from severe hearing deficiency [14].

Patient	Logatom test	Word test + CCITT noise	Sentence test + cocktail-party noise
Patients 1–10	55% › 66%	11% › 33%	55% › 77%

values (without compression) mapped (›) to values (with adjusted compression).

5. Conclusions

We reviewed in this paper three examples of our common research devoted to applications of multimedia technology to hearing impaired people.

Results of over 500,000 tests performed so far show that multimedia computers running appropriate software are effective tools for performing hearing screening tests. In this approach it is important to use tests adapted for individual age groups and use speech in noise as test material.

Computers can be used effectively also in tinnitus diagnosing and digital ultrasound devices are applicable to the therapy process. Moreover, the approach aimed at exploiting the analogies between nature and technology helped to search for tinnitus origins and to formulate hypotheses concerning the process of its generation.

Fitting hearing aids according to patients' needs and hearing characteristics is yet another problem, which so far has not been finally solved. As we demonstrated this process may utilise soft computing (fuzzy reasoning) in the course of optimising compression settings in hearing aids.

Acknowledgments

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