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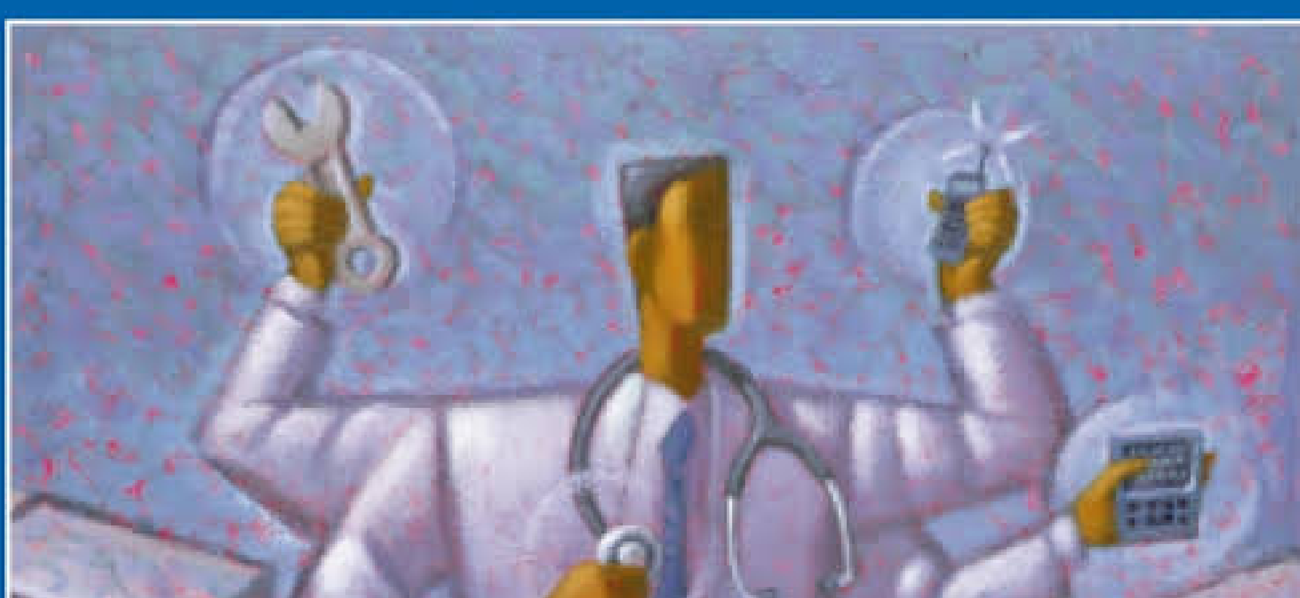
Information Technology Solutions for Healthcare



HEALTH INFORMATICS SERIES

10 IT Applications for the Remote Testing of Hearing

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Telemedicine can play an important role in diagnosing and treating hearing losses. This fact is associated, among others, with the methodology of audiometric measurements and with supporting hearing through hearing aids and cochlear implants. Current problems related to treating hearing impairments and total deafness pose a distinct challenge for science, which must provide ever more effective methods for application in diagnostics and audiology as well as otolaryngology practice. Scientific experiments in these fields also indicate ways of supplementing methods of acoustic signal analysis and processing used in hearing diagnostics and prosthetics with modern technological advances, so that patients can benefit from methods whose history is so short that they may be classified as experimental.

Similar needs for wider introduction of electronics and computer science technologies can be recognised in phoniatry and speech therapy, where the diagnostic process, devoid of advanced signal analysis tools, remains very difficult and therapy is often arduous and laborious.

Choosing hearing aids according to patients' needs and hearing characteristics is yet another problem which has not been solved yet. Experience shows that even the preliminary step, i.e. adjusting the aid's acoustic path to the patient's hearing track, is very important, especially in the case of major hearing losses. The process of regulating parameters of signal track for electronic hearing aids should utilise, insofar as possible, the knowledge on the patient's hearing characteristics, with the aim of allowing him to understand speech. The multimedia hearing aid fitting system presented in this paper is based on research concerning the degree of understanding speech in the presence of noise and on determining (on that basis) the optimum characteristics of the multi-band dynamic compressor, which is an important part of advanced hearing aids. This process utilises so-called fuzzy reasoning, which marks our application as one of the first in audiology to employ soft computing methods.

Advances in teleinformatics as well as its wide employment in recent years have opened new possibilities for conducting mass screening of hearing, tinnitus (ear noises), speech and vision. Diagnostic and recovery systems associated with the interactive medical portal *Telezdrowie (www.telewelfare.com)* designed by the institutions mentioned in the header of this paper serve as an example of how simple diagnostic methods employed in screening tests can be mass-deployed thanks to teleinformatics, thus defining a new quality for widespread diagnostic tests of communication senses. In addition, one cannot neglect the influence of advances in the field of network database systems on epidemiology tests and associated studies on hearing deficiency and vision deficiency.

The aspects mentioned above will be briefly illustrated by sample applications deployed under the supervision of the head of the present project. For practical reasons, the presentation is limited to selected applications associated with hearing. Information on other applications can be found in [4], [6], [8], [2].

1 Methodology of Hearing Screening

Screening methods employed thus far can be divided into three groups: The first one includes methods which only use questionnaires for the person being diagnosed or for people from his/her vicinity (e.g. parents). The second group employs physiological and audiometric measurements. The third group of methods is comprised of tests employing both questionnaires and measurements. Several measurement methods can be distinguished among screening tests. Screening can utilise audiometric tests. According to the standard proposed by ASHA (American Speech-Language-Hearing Association) in 1985, an audiometric screening test should determine whether a child can hear in both ears three 20-dB HL tones of frequencies of 1000 Hz, 2000 Hz and 4000 Hz, respectively. The above procedure may be extended by supplying an additional 500-Hz tone of the same level of 20 dB HL. If the child cannot hear at least one of the tones presented, he or she must undergo detailed audiometric tests. In the case of adults, the test signal levels are increased, usually by 5 dB.

Screening tests associated with the project show that speech audiometry can also be employed for screening tests. The main advantage of speech audiometry is that, unlike tonal audiometry, it allows not only for measuring receptive or conductive hearing, but also for assessment of the whole complex hearing-perception mechanism. That is why speech audiometry is proposed for use in screening children and teenagers. 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 50% of the time. The second threshold type is *speech reception threshold* (SRT), defined as the lowest signal level for which speech is understood 50% of the time. These thresholds are presented in Figure 10-1.

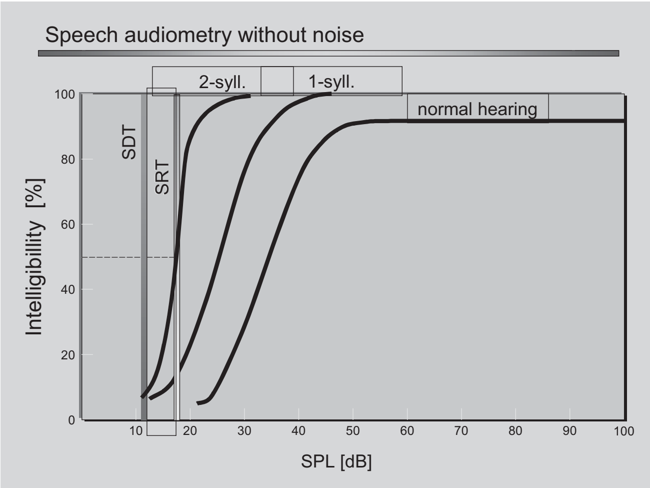

Figure 10-1 Sample speech audiograms.

Sound material presented during audiometric tests may vary in type. For example, it may involve logatoms, monosyllabic words, disyllabic words or whole sentences. The degree of understanding is directly influenced by the type of verbal material and, more importantly, acoustic pressure level. This effect is presented in Figure 10-2, where curves marked "1-syll." and "2-syll." correspond to monosyllabic and bisyllabic (spondey) tests, respectively. In the case of regular hearing the speech audiograms are rising curves, which describe an increase in the understanding of speech, coupled with increasing sound exposure. However, for certain hearing impairments increasing the speech signal level causes the degree of understanding to decrease. Analysis of this phenomenon may be facilitated by the so-called audiogram curvature index defined by the following formula:

$$RI = \frac{(Y_{\max} - Y_{\min})}{Y_{\max}} \qquad (10\text{–}1)$$

where: RI is the audiogram curvature index,

Y_{max} is the maximum understanding value,

Y_{min} is the minimum understanding value for the sound levels presented at the uncomfortable hearing level.

A value of the audiogram curvature index exceeding 0.45 may indicate hearing damage.

When designing tests for speech audiometry, one must take into account the number of words in the test, since it influences test reliability. For example, Figure 10-2 presents a graph of standard deviation as a function of the number of words in the test and of speech understanding. This graph was constructed on the basis of the following relation:

$$\sigma = 100\sqrt{\frac{p(1-p)}{n}} \tag{10–2}$$

where: σ is the standard deviation,

p is the speech understanding ratio,

n is the number of words in the test.

As can be seen from the graph, test reliability decreases (i.e. standard deviation grows) with a decrease of the number of words used in the test. Tests using 10 or even fewer words are particularly unreliable.

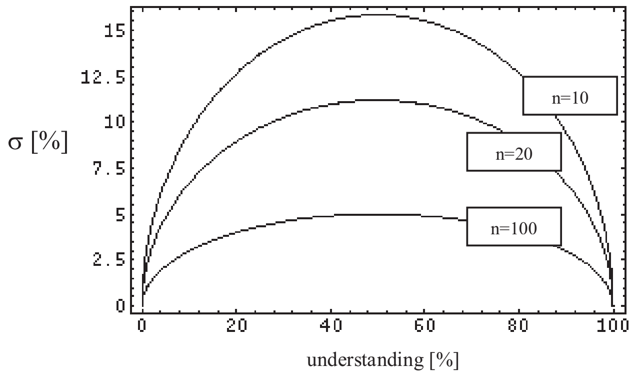

Figure 10-2 Standard deviation σ as a function of the number of words in the test and of speech understanding.

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, although in this case it is done on purpose. This effect can be seen on audiograms as classic noiseless speech audiograms "shifted" to the right by a value equal to the masking noise level (Figure 10-3).

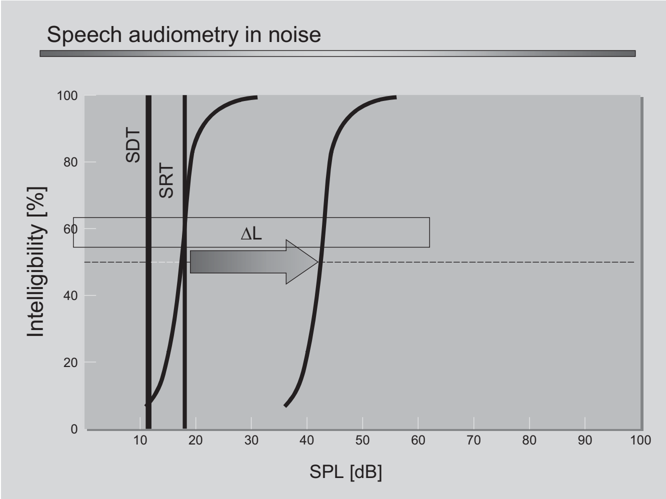

Figure 10-3 Example of audiogram being "shifted" due to influence of masking noise of level ΔL.

Widely-used measurement methods are based only on three-tone tonal audiometry. Within the presented project a new method, based on additional use of speech-in-noise audiometry, was proposed. Besides simple tones this method utilizes a speech signal mixed with noise in appropriate proportions. This method, in conjunction with the three-tone test and with a system of electronic questionnaires, forms the basis for a mass hearing screening system, which has so far been used to test above 150,000 school pupils. The patronage of MENiS (Polish Ministry of Education and Sport) over this system allows us to expect that the tests will become ever more widespread, covering schools in the whole country, similar to screening ear noises, speech [4] and sight [7]. Figure 10-4 illustrates typical results obtained from tests performed in schools. A criterion of min. 70% correct answers in the speech-in-noise test causes up to 20% of children to require clinical diagnosis in order to exclude hearing difficulties. Performing the full set of tests, i.e. analysing the electronic questionnaires and the results of the three-tone test, often reveals that over 15% of children have difficulties with attaining satisfactory results of the screening test. A standard audiometric test allows us to detect hearing problems in up to 20% of the population being tested. This result confirms the frightening extent of hearing impairment, even in young people.

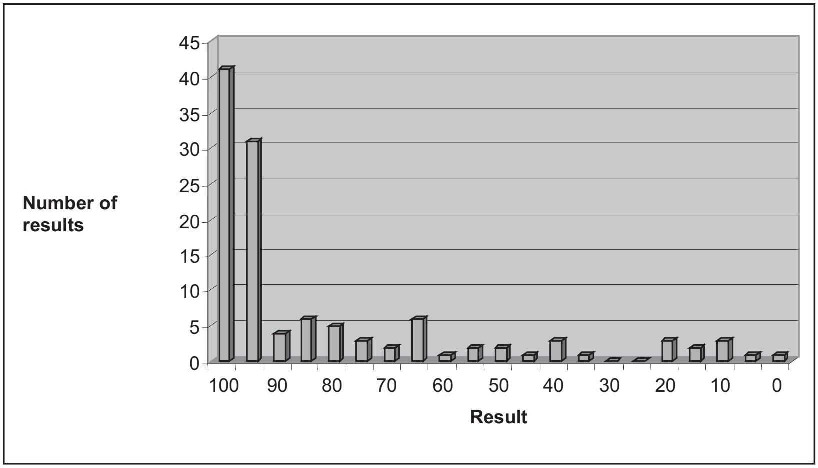

Figure 10-4 Summary of results of the picture test (utilizing speech in noise). The x-axis represents the percentage of correct answers.

2 Method of Computer-Aided Adjustment of Hearing Aids

Widely-used systems for adjusting hearing aids allow for setting optimum parameters of a hearing aid, which usually does not mean that the optimum characteristics of hearing loss correction are obtained. The idea of a multimedia system for adjusting hearing aids (developed within the scope of a Ph.D. thesis by Piotr Suchomski [12]) was to create a computer utility allowing for unrestrained shaping of the characteristics of the prospective hearing aid which would optimally compensate for the given hearing damage. Since in a general case the problem of adjusting a hearing aid can be boiled down to the problem of adjusting the wide dynamics of speech to the narrowed dynamics of impaired hearing, the presented system focusses mainly on determining the characteristics of impaired hearing and then on deriving the dynamics of the sought-after hearing aid that would compensate for the damage. This is, however, a simplification, because in a real hearing aid the quality of hearing loss correction is also influenced by other factors, such as noise reduction algorithms, equalising of the processed signal and acoustic elements of the hearing aid itself [3], [12].

This paper presents the general principles of operation of the designed system as well as the results of experiments utilising implementations of this system. Several important stages of operation can be distinguished in the designed system (Figure 10-5). First, dynamics characteristics of the diagnosed hearing impairment are determined. These are then used to determine dynamics characteristics of the desired hearing aid. On the basis of the derived dynamics characteristics of the desired hearing aid, hearing training is conducted, utilising an extensive base of speech signals (100 logatoms, 200 words, 630 sentences).

The 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 the sensation of Loudness Growth in half-Octave Bands (LGOB) [1]. The test utilises signals in the form of white noise samples filtered in half-octave bands, with central frequencies of 0.5 kHz, 1 kHz, 2 kHz and 4 kHz respectively. During the test the person being diagnosed 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 (Figure 10-6).

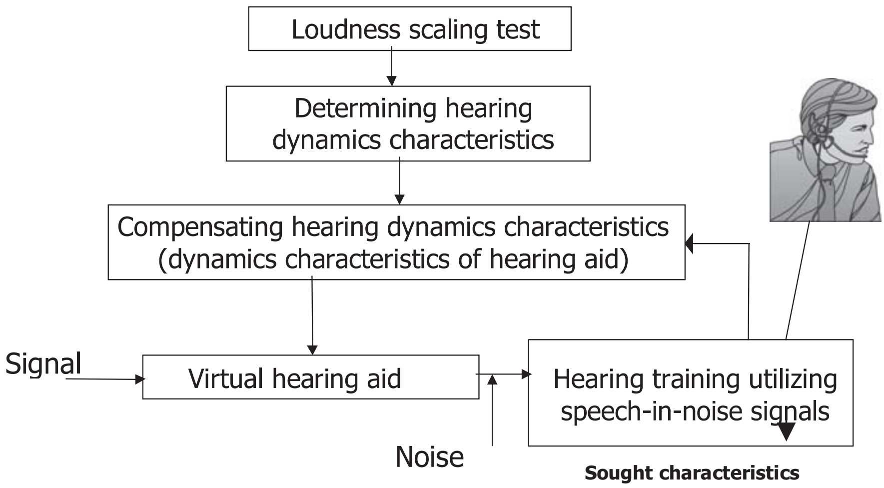

Figure 10-5 Scheme of adjusting hearing aid parameters.

The presented system utilizes a special module for converting results of loudness scaling into hearing dynamics. It uses fuzzy processing to map the results of scaling derived from the categories of the loudness sensation evaluation onto an objective scale of input sound levels, expressed in decibels. The category scale is represented by 7 fuzzy sets described by membership functions. These membership functions have been derived on the basis of statistical analysis of loudness scaling for 20 persons with no hearing impairments. The output of the fuzzy logic system is described by 13 membership functions which represent the differences between any given result of scaling and analogous results for unimpaired hearing [9], [11], [10].

As there are four frequency bands, with central frequencies of 500 Hz, 1000 Hz, 2000 Hz and 4000 Hz, which are tested in the LGOB test, four sets of membership functions are required. In order to derive these membership functions one has to perform the LGOB test on several dozens of regular-hearing persons. During the method design phase we prepared the required membership functions on the basis of generally-accepted approximation of LGOB test results for normal hearing (Figure 10-7). In order to obtain a much more reliable and realistic set of membership functions, one had to test loudness scaling using the LGOB test on several dozens of healthy individuals.

Figure 10-6 Scale of loudness sensation evaluation (graphical interface of a developed computer program).

One of the basic methods of approximation that come to mind when analyzing typical fuzzy logic systems and the sets of theoretical membership functions (as in Figure 10-7) is the approximation of the result-containing set boundary with triangle-shaped functions. It can be performed e.g. through mean-square approximation. In order to perform this task, one should determine the equations of two straight lines approximating triangle sides in the mean-square sense. The algorithm used by the authors to determine the triangular membership functions involves the following steps:

hearing, the designed system should "calculate" the difference between the given loudness sensation evaluation and the correct loudness sensation evaluation corresponding to the given test signal. This difference should be expressed in decibels.

Analysis of a typical plot of LGOB test results reveals that in between the seven categories of loudness sensation evaluations, one can define six differences which point to hearing loss (area below the LGOB curve for regular hearing) and six differences which point to hypersensitivity (area above the LGOB curve for regular hearing). No difference is a treated as a special case of difference. The above analysis leads to the conclusion that the output of the described fuzzy system can be expressed by a set of thirteen membership functions (Figure 10-9) representing the difference between the factual loudness sensation evaluation and the evaluation for regular hearing. Fuzzy sets obtained in this fashion can be described with the following labels (indicating the size of the difference): *none*, *very small*, *very small*+, *small*, *small*+, *medium*, *medium*+, *big*, *big*+, *very big*, *very big*+, *total*, *total*+. Labels marked with "+" sign denote positive difference (hypersensivity).

Fuzzy processing depends on a properly defined rule basis. Fuzzy logic rules take on the following form:

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

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 the 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 between the fuzzy sets associated with individual premises, labels of fuzzy sets for the first premise use lowercase letters while those of fuzzy sets for the second premise utilize capital letters.

Generally speaking, the rule base is designed on the basis of expertise. In our case, such expertise can be derived from the analysis of the LGOB test. The analysis of LGOB test results for healthy persons shows that the loudness scaling is linear in character, albeit the factor of proportionality increases from 1:1 to 2:1 (the loudness sensation rises twice as fast) for test signals exceeding 100 dB SPL. Based on this information, a rule base was prepared. It is presented in Table 10-1. Categories associated with regular hearing are marked in lowercase (e.g. "loud"), while categories written in capitals (e.g. "LOUD") represent hearing impairment.

The last stage of the system's operation is defuzzyfication, i.e. conversion of the obtained categories to a numerical value. After performing this process (using the centre of gravity method) we obtain the relations between sound levels expressed in decibels and the expected subjective assessment of loudness sensation for a given patient. This relation is the sought-after characteristic of hearing dynamics).

In order to determine the entire scope of dynamics characteristics for a given patient we had to create an algorithm which would calculate the desired characteristics on the basis of LGOB test results using the designed method of determining the difference between regular and impaired loudness scaling. The general scheme of the procedure is shown in Figure 10-10.

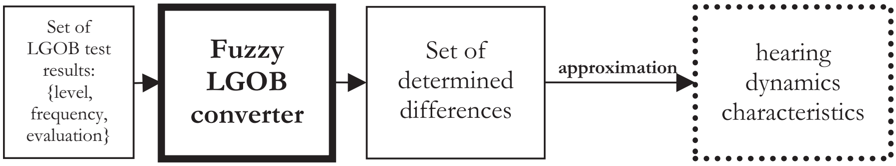

Figure 10-10 Diagram of fuzzy processing-based algorithm for determining characteristics of hearing dynamics.

Table 10-1 Rule base for the fuzzy system (the decision attribute is the difference between regular perception and perception associated with 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

The system is fed a stream of results of the given LGOB test and produces a stream of consecutive differences for each result of the test. An LGOB test result consists of three parameters (level, frequency, evaluation), where "level" is the level of the given test signal (expressed in decibels), "frequency" is the frequency band encompassing the given test signal (expressed in hertz) and "evaluation" is the loudness sensation category used to evaluate the loudness sensation caused by the given test signal.

A fundamental advantage of the presented 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 hearing dynamics utilizes all the available information on regular loudness scaling in the LGOB test, while standard methods of adjusting hearing aids rely on averaged data only. In the course of the standard determination of loudness scaling (LGOB test) the diagnosed person is presented with multiple filtered noise samples and the obtained results are subsequently processed in a statistical manner, while the proposed method only requires 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 audiologic practice.

Figure 10-11 presents a sample plot of loudness scaling results with the LGOB test, obtained for test signals from the frequency band centred around 500 Hz.

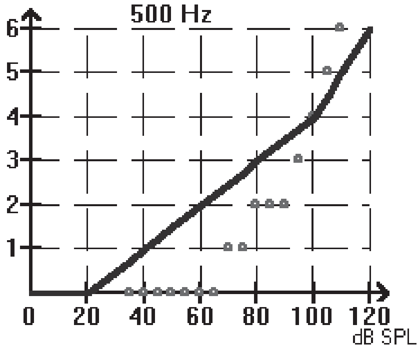

Figure 10-11 Sample results of LGOB test (Y axis represents category scale).

The derived dynamics characteristics of impaired hearing can be additionally used by the presented system for approximate simulation of hearing loss. In order to derive the dynamics characteristics of any hearing aid, the system compensates for the characteristics of impaired hearing. This compensation is based on flipping the hearing dynamics characteristics around the $y = x$ straight line (Figure 10-12).

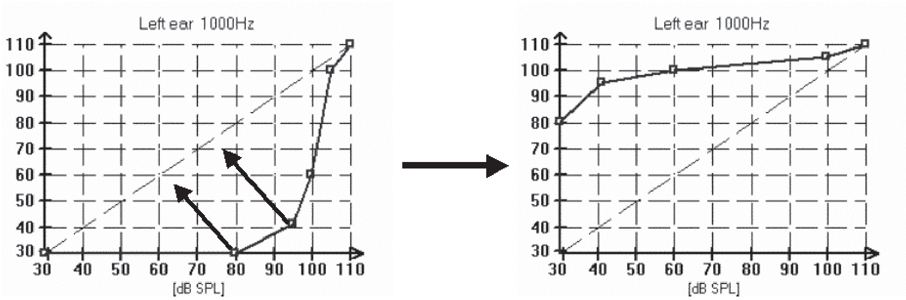

Figure 10-12 Compensating for the dynamics characteristics of impaired hearing.

The obtained hearing dynamics characteristics have been verified on the basis of training with speech-in-noise signals, processed according to the virtual hearing aid module algorithm (Figure 10-13). The speech signal is first filtered into 4 octave-wide bands with central frequencies identical to those used in the loudness scaling test. For each band, the signal dynamics are processed in a way adequate for the dynamics characteristics of the hearing aid, for the given band. When the dynamics have been processed, signals from individual bands are added to form a common channel. The module also allows for adding arbitrary noise, at both input and output. Moreover, we can equalize the level of the processed signal in individual bands, critical for speech.

The obtained degrees of speech understanding for individual listening tests are presented in Table 10-2. The degree of speech understanding is expressed as percentage values. The first value denotes the degree of speech understanding without dynamics processing, while the second value (to the right) denotes the degree of speech understanding following dynamics processing, based on the obtained dynamics characteristics of the hearing aid.

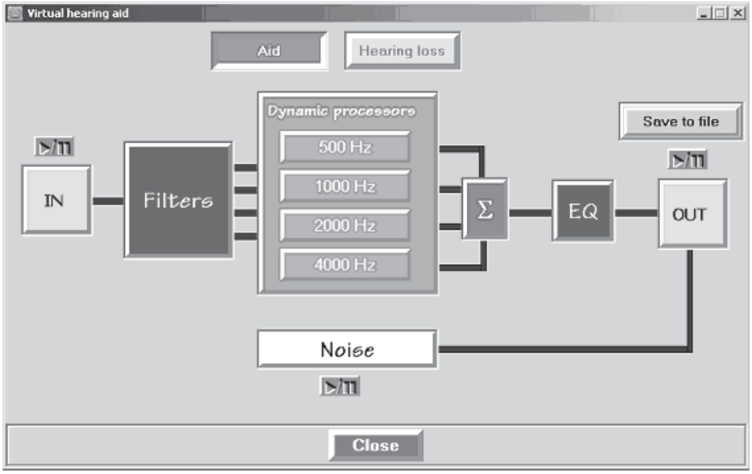

Figure 10-13 Algorithm for signal processing in a virtual hearing aid.

Table 10-2 Averaged degrees of speech understanding for patients suffering from severe hearing deficiency (without compression)-> with compression adjusted on the basis of investigation of loudness discrimination)

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

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 a condition may be caused by degeneration of external hair cells, causing neurons to be activated by stronger-than-normal signals. In such cases, an elevated activation threshold is present. However, before such elevated threshold appeared for the given patient e.g. due to a disease or development of otosclerosis, signals had been received and interpreted as hearing stimuli at higher levels of the hearing track. This fact results in the introduction of an additional mechanism of threshold quantisation of weak acoustic stimuli, which in turn relates to the elevated activation threshold of nerve cells. Existing audiology theories attempting to explain this phenomenon do not directly take into account the mechanisms of signal quantisation occurring due to the presence of threshold characteristics in the transmission system. Such interpretation becomes possible only with the knowledge of electric signal processing developed in other fields of science, e.g.

in digital signal processing. Using this approach, we have proposed a new interpretation of the phenomenon of tinnitus generation on the basis of the audio signal quantisation theory [4]. Moreover, using digital signal processing concepts we have proposed a theoretical explanation of a method to eliminate threshold quantisation-related tinnitus, basing on the dithering technique. Generally speaking, this technique involves supplementing low-level useful signals with certain levels of noise, which has the effect of stopping the process of spontaneous noise generation in the hearing track (resulting from threshold characteristics). One can easily notice that a similar approach is used to reduce tinnitus in audiology, where masking noise generated by a special masker device is utilised. The widely known effectiveness of such techniques for reducing both ear noises and quantisation noises generated spontaneously in electronic circuits indicates that ear noises can indeed be interpreted as a direct consequence of quantisation of weak signals in threshold systems. At this point it may be useful 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 [5].

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

$$Q(x) = \Delta\left[\frac{x}{\Delta} + \frac{1}{2}\right] \tag{10–5}$$

or:

$$f(x) = \Delta\left[\frac{x}{\Delta}\right] + \frac{\Delta}{2} \tag{10–6}$$

where: x – value of the sample before quantization (at input)

Δ – height of quantization 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. The distribution of error probability density for a quantiser of the transition function described by formula (10–6) is a rectangular window function:

$$p_\delta(x) = \begin{cases} \dfrac{1}{\Delta} & dla\,|x| \le \dfrac{\Delta}{2} \\ 0 & dla\,|x| > \dfrac{\Delta}{2} \end{cases} \qquad (10\text{–}7)$$

For complex input signals the maximum error is equal to the least significant byte (LSB) and, given a good approximation, 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 low-level input signals 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 [-Δ/2,Δ/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 constitutes a disadvantageous phenomenon associated with quantisation.

The dither technique is aimed at modifying statistical values of the 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 amplitude to the quantisation step, the error depends strongly on the input signal and it introduces audible distortion and modulation noise. Use of a dither signal with appropriately-shaped statistical properties may cause the audible distortions to become similar in character to stationary white noise.

Modern digital audio tracks use the dither technique, with noise described by a triangular function of probability density and peak-to-peak values of 2 LSB. The dither noise is therefore additive in nature, and it is usually introduced into the signal before the quantisation step. The averaged response obtained at the conversion system output is the following function of the input signal:

$$\overline{y}(x) = \int_{-\infty}^{\infty} y(x+\upsilon)\,p_\upsilon(\upsilon)\,d\upsilon \qquad (10\text{–}8)$$

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

$$p_\upsilon(\upsilon) = \begin{cases} 1/\xi, dla\ |\upsilon| \le \xi/2 \\ 0 \end{cases} \qquad (10\text{–}9)$$

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

Figure 10-14 illustrates the fundamental phenomena taking place when a signal of amplitude comparable to the quantization threshold is fed to the A/C converter input.

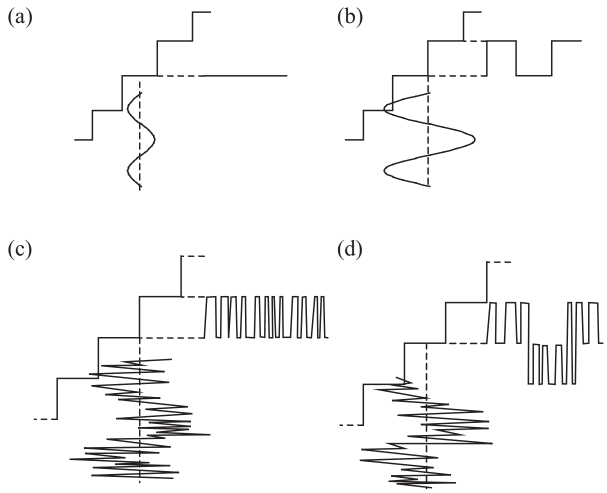

Figure 10-14 Effects associated with quantisation of small amplitudes and the influence of the dither noise: (a) "digital deafness"; (b) "binary quantisation"; (c) dither removes the insensitiveness range of the converter; (d) "response blurring" in the case of binary quantization.

Analysis of the influence of the dither noise on quantisation within the first part of the quantiser's characteristics (see Figure 10-15) shows that if the continuously-present dither noise is related to the quantiser's characteristics in a certain fashion, for properly adjusted noise level its steps become blurred and it approximates linear characteristics, therefore reducing the quantisation error and the related noise decrease.

FIGURE 10-16 a shows the result of quantization of a sinusoidal signal obtained without introducing dither noise, while FIGURE 10-16 b shows the result of quantisation of the same signal in the presence of the dither signal. The same figure (FIGURE 10-16 c and d) proves that averaging the representation (b) can lead to almost perfect reproduction of the original waveform that was subject to quantisation. We can also note that, as the sense of hearing possesses noticeable integrating properties, similar processes can certainly take place in the hearing track.

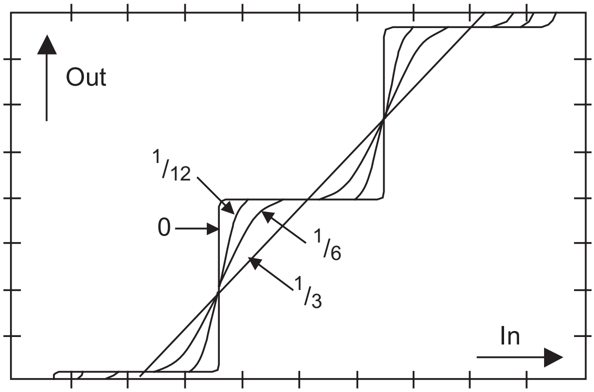

Figure 10-15 Effect of linearisation of conversion characteristics under the influence of dither noise at various levels corresponding to fractions of the quantisation step.

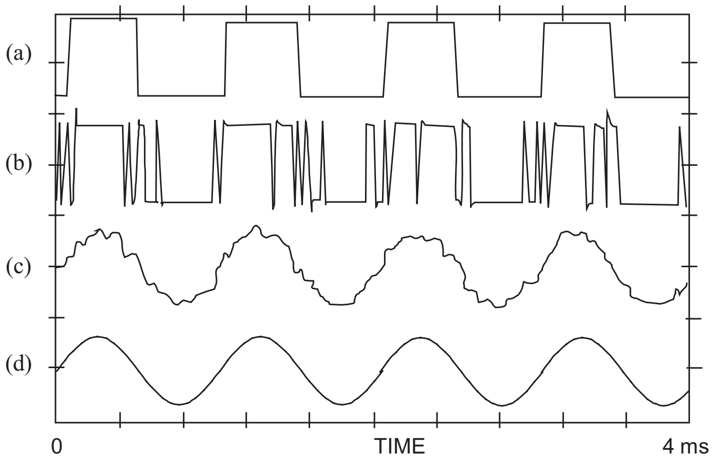

FIGURE 10-16 Effects of quantisation of waveform of amplitude corresponding to quantisation step:
(a) harmonic signal directly after quantisation; (b) quantisation using dither noise; (c) signal from the previous plot averaged over 32 periods; (d) result of averaging over 960 periods.

FIGURE 10-17 shows how adding dither noise affects the reduction of harmonic distortions.

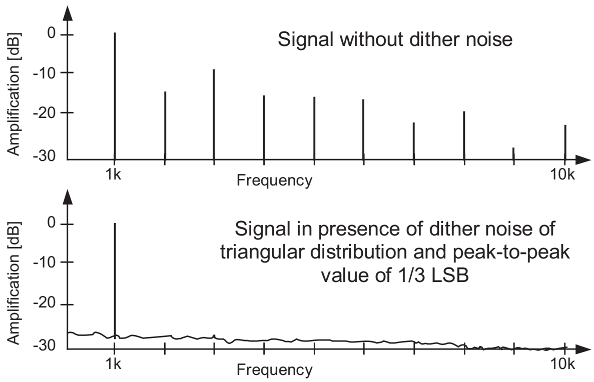

FIGURE 10-17 Spectrum of a quantised harmonic signal with an amplitude corresponding to the quantisation threshold (a) and a spectrum of the same signal in the presence of dither noise added at input of the A/C converter (b)

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

$$P_n^{\,2}(x) = \int_{-\infty}^{\infty} [y(x+\upsilon) - \overline{y}(x)]^2 \, p_\upsilon(\upsilon) d\upsilon \qquad (10\text{–}10)$$

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

$$p_\upsilon(\upsilon) = \frac{1}{\sqrt{2\pi}\sigma_\upsilon} \exp\left(\frac{-\upsilon^2}{2\sigma_\upsilon^{\,2}}\right) \qquad (10\text{–}11)$$

$$\overline{\nu} = \sum_k \nu_k p_\nu[\nu_k] \qquad (10\text{–}12)$$

$$\sigma_\nu^2 = \sum_k (\nu_k - \overline{\nu})^2 p_\nu[\nu_k] \qquad (10\text{–}13)$$

modulation with noise is practically absent. Adding Gaussian noise allows for reducing quantisation errors, while being relatively simple to implement.

Introducing the “masking” dither noise lets us obtain the desired results of eliminating the quantiser’s insensitiveness range and minimising distortions occurring for very low amplitudes of the signal being quantised. Audibility of the introduced noise can be decreased by preliminary shaping of its spectrum so that noise energy rises towards higher frequencies. The same principles remain valid for masking tinnitus, which suggests direct similarities between the phenomena occurring in electronic and biological signal-transmission systems.

The possibilities resulting from the above statements have been 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 the patient to diagnose his/her own ear noises using a computer. Subsequently, after consulting a doctor, the patient can download a masking noise appropriate for him/her and store it in a portable mp3 player, which allows for effective treatment by masking tinnitus.

4 Summary

Results of over 200,000 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 noised speech as test material.

Choosing hearing aids according to patients' needs and hearing characteristics is yet another problem, which has so far remained unsolved. Experience shows that the process of regulating the signal track parameters of electronic hearing aids should utilise knowledge on the patients' hearing characteristics in the maximum degree possible, with the aim of allowing them to understand speech. In order to face this challenge, we had to design new methods of adjusting hearing aids that would involve computers to a greater extent than before. An example of such a method is the multimedia hearing aid selection system presented in the paper, which is based on research regarding the degree of understanding speech in the presence of noise and on determining the optimum characteristics of the multi-band dynamics compressor, which is an important part of advanced hearing aids. This process utilises fuzzy reasoning, which marks the final application as one of the first in audiology to employ soft computing techniques.

Besides hearing deficiency, the problem of ear noise remains one of the principal hindrances for patients seeking audiological aid. The reasons behind this problem are not always understood by the patients, which hinders diagnostic and therapeutic interactions. Nevertheless, modern sound engineering and computer science can significantly improve the situation in both fields. Computers can be used effectively in tinnitus diagnosing and miniaturised recording and playback devices are successfully utilised in the therapy process. Moreover, the biocybernetics-inspired approach to scientific problems aimed at exploiting the analogies between nature and technology lets us search for ear noise origins and formulate hypotheses concerning the process of their generation.

As mentioned in the introduction, projects described in the present paper do not exhaust the scope of the application of sound and vision engineering in biomedicine. In fact, many other systems can be mentioned as well: e.g. a system for vision screening designed by the authors, a diagnostic system and therapeutic devices for correcting speech, and numerous other research projects that either have been or are being conducted at many research facilities around the world. Interdisciplinary applications of sound and vision engineering and multimedia techniques underlying telemedical applications result in increased effectiveness of diagnostic and therapeutic methods and therefore in increasing the effectiveness of detecting diseases and impairments, as well as of their prevention and therapy.

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