Adaptive Stimulation Method Used to Improve the Auditory-Verbal Education Procedure

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Abstract: - The paper presents an adaptive stimulation algorithm designed to be implemented in an automated software application that is used in auditory-verbal education. The algorithm is able to adapt in real-time the distribution of image-sound stimuli emission according to the performance of the subject during the training sessions. This procedure is an improvement of the classical unbiased training methodology, in which all the stimuli are even (uniform) generated regardless the auditory knowledge cumulated by the subject during the training stages. The proposed methodology generates the stimuli in a differential (not uniform) manner in such a way to emphasise the learning process for the incorrect associated image-sound inputs and to force the subject to memorize it faster and, on the other side, to reduce the stimulation of sounds that were already learnt, for which we try only to refresh (to keep warm) the information already cumulated.

Key-Words: - Adaptive algorithm, auditory-verbal education, e-learning, software design, software implementation

1 Introduction
The ability to perceive sound is a fundamental requirement in the correct development of the human being. Correctly perceiving and understanding sounds is the basis of interaction between the world and an individual. From a very early age the process of discovering everything that surrounds us implies correctly assessing and correlating sound and visual stimuli. Therefore, the malfunctioning of the auditory system determines the human to be subject to a complex restructuring process of the acting strategies and of the sensorial and psychic compensation strategies. The sense of orientation and in particular the development of the verbal language (and the performing of its functions) will be disturbed [1].

Auditory-verbal education is responsible with the training of persons in mastering their hearing (which has been partly regained after a medical intervention) [1], [2]. The classic approach in auditory therapy implies regular visits to specialized cabinets. This can be an exhausting and time consuming activity as the number of specialized centers is very small and therefore, in many cases, one should travel a lot to reach the cabinet. Also, the expertise the specialist possesses is crucial in obtaining good results and therefore the number of properly prepared specialists can also be important with regard to the availability of good treatment.

A solution like an e-learning type computer application [3] capable of supplementing and to some extent replacing the specialist could be useful in cases where the lack of access to a cabinet is an impediment [4]. Furthermore, if the solution is build upon web based e-learning principles then the availability to the end-user is enhanced and also the application becomes platform independent, the only requirement for the personal computer being the existence of a web-browser capable of correctly running the content. [5], [6]

This paper presents further principles of designing such a computer application, in addition to the ones presented in [7]. We present now our efforts on developing an algorithm to be used in automatically adapting the sound stimuli emission. This is an important factor in the architecture of such a software system due to the fact that subjects have different learning abilities that need to be focused on accordingly

2 Problem Formulation
The auditory education which prepares the auditory-verbal education was and remains a priority in the recovery methodology panoply for the hearing
impaired persons in general and for hearing impaired children in particular. It is known the fact that one cause of the lacking of response to word stimulation, besides the auditory deficiency degree, is represented by the inability of freely detecting meaningful verbal material. These lacks are added to the causes determining the low intelligibility of hearing impaired persons’ language, decreased scholar performances, and difficult social integration, revealed by the literature [8], [9], [10].

The main issue encountered when developing an e-learning type application represents an incorrect development process, where different stages are mixed or skipped. In order to correctly associate a sound with the visual information, the child first has to hear it, listen to it and therefore consciously and actively accept it. Following the established methodology, and as presented in [2] there are some distinctive stages that the child should run over in order to successfully complete an auditory-verbal education process:

1) Stimulating the attention towards auditory acceptance;
2) Detecting, differentiating, identifying and recognizing the nonverbal sounds in the nature (objects, natural phenomena, animals);
3) Identifying and differentiating, recognizing and reproducing the verbal sounds; comprehension and articulation of the simple and complex verbal structures.
4) Imposing practice of the prosodic traits of language: rhythm, accent and intonation.

It is therefore mandatory to follow these theoretical stages in order to build a software application with the intended goal of offering a correct training session (therapy).

3 Problem Solution

3.1 Using IT Applications
Following the above mentioned steps is important in building an efficient e-learning type computer application for auditory education. It should be underlined that the decision to research in this field is not generated by the inefficiency of the classic methodology but by the lack of specialized centres and in some cases to the lack of expertise of the personnel. Also, almost no research exists in this field in our country or, the little work completed, is either obsolete or discontinued [11]. Therefore one should know that the intention is to offer the real life methodology – one that proves to be efficient – in a virtual environment, accessible to anyone that owns a personal computer.

As presented in much more detail in [7] and [12], the computer application should present an appealing interface that attracts the attention of the hearing impaired and keeps him focused for the duration of the training process. The interface and specifically, the graphics should be adapted to the age of the impaired.

Also it is important to build a large database of both sound and images. Images and sounds are going to be presented to the impaired, training him in this way to associate the subject of the images with the sound he hears in the computer speakers. The database should be well organized on the type of the stimuli (e.g. sounds of nature elements, animals, machines, etc.) and should also present the possibility of regular update.

After several sessions of training the impaired should be tested to see how well he assimilated the presented combinations of sound / images. It is important at this point to remember the results of such tests in order to adapt further training accordingly.

The problem of presenting a method of verbal training is not treated in this article but will be reviewed by our team in future studies.

At this point we would like to concentrate on developing automated methods of correctly training a hearing impaired in associating sounds with their emitters. This is a very important element in the entire software architecture as it is responsible for adapting the parameters of the training sessions for a given person (e.g. the number and type of the images / sounds presented) in accordance to information extracted from previous tests. By doing so the application emphasizes those stimuli that have not been correctly assimilated in previous training sessions. The adaptive behaviour can also be used in the future development of the verbal language when the subject will be presented different words, on the same principles of adapting the new stimuli in accordance with the past performance of the impaired.

3.2 Adaptive Stimulation algorithm
We therefore propose a new method of adapting the sound stimuli emission. Details about this method are presented below.

Building such an algorithm, at very first, presumes recording all the answers from the periodical test sessions the subject submits. This can simply be done by knowing at each question in the test, the correct type of stimuli and increasing a
counter associated with that type every time the user answers correctly (the very same can be done when counting incorrect answers). Also, knowing the total number of questions in the test (due to the total number of questions per type of stimulus) a simple statistics can be computed which can be used to determine the performance of the user.

For a database that contains $N$ types of stimuli, $2N$ counters will be used to estimate the correct associated probability $p_c(t)$ of each input at the current moment $t$ of the training process: $N$ for the correct answers $k_i(t)$ and $N$ for the total number of questions for each type $q_i(t)$.

$$p_{correct}(t) = \frac{k_i(t)}{q_i(t)}, \quad i = 1...N \quad (1a)$$

And, therefore, the incorrect associated stimuli probability will be:

$$p_{incorrect}(t) = \frac{k_i(t)}{q_i(t)}, \quad i = 1...N \quad (1b)$$

Consequently, to be able to differentiate the learning process of the correct and incorrect associated stimuli, the number $n_i$ of the input pairs image–sound in the following training session $(t+1)$ for each category will be changed proportional with the incorrect answered above estimated probability, as follows:

$$n_i'(t+1) = n_i(t) + \left(1 - \frac{k_i(t)}{q_i(t)}\right)n_i(t) \quad , i = 1...N \quad (2)$$

In such a way, the total number of stimuli in the next training session will increase by a factor of $F$, where $F$ is

$$F = \frac{\sum_{i=1}^{N} n_i'(t+1)}{\sum_{i=1}^{N} n_i(t)} \quad (3)$$

Due to the subject concentration limited time interval, it is necessary to have a constant training procedure, consisting of similar learning session and, therefore, to preserve theirs features (number of stimuli, duration, etc.), the normalization of formula (2), in respect with the total number of the inputs in the training session, is necessary:

$$n_i(t+1) = \frac{n_i'(t+1)}{F} \quad (4)$$

On the same principle, the idea can be elaborated for every stimulus in a category, if a more detailed view on the performance is needed to build a more accurate adaptive algorithm. This latter case is particularly important when the database consists of images associated with more than one sound or vice-versa. In such a case, the subject has the added difficulty of recognizing the sound emitter in different images and also of recognizing different sound stimuli associated with the same emitter. More generally presented, the idea behind the algorithm is to modify the probability density function of the stimuli emission (in the training session) in accordance with the probability density function of the incorrect classification rate of each category in test sessions. If we consider that the initial setup of the application emits stimuli according to a uniform distribution – Fig. 1 – then this distribution is modified according to

$$p_{stimulation}(i,t+1) = \frac{p_{stimulation}(i,t)(1 + p_{incorrect}(i,t))}{p_{total}(t)} \quad (5)$$

where:

$$p_{total}(t) = \sum_{k=1}^{N} p_{stimulation}(k,t)(1 + p_{incorrect}(k,t)) \quad (6)$$

the initial uniform distribution is:

$$p_{stimulation}(i,1) = \frac{1}{N} \quad (7)$$

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Fig.1 - Initial probability distribution of stimuli emission.
The resulting distribution is presented in Fig. 2. What can be seen is that the probability of presenting a certain category of stimuli (or stimulus) in the training session is higher for the cases where the error rate is also higher and vice-versa – Eq.(5). Implementing the above algorithm do not change the initial computer requirements, the only change is that the data for a given subject can be stored on the disk and be loaded in the program every time the subject resumes his training.

As a ‘stopping’ condition, once the subject performs, on average, better than a given limit the training is considered to be complete. Yet this mark and the amount of training sessions between tests or the number of total questions in a test / slides in a training session can be dependent on the specialist opinion and is a problem that still needs our attention.

There are several options the application presents:
- three category of stimuli (animals, home, nature)
- the possibility to select the number of questions in the test
- the possibility to select the number of answers per question
- the path of the output file
- the possibility to submit data about the current subject

The example presented in [12] was implemented using ActionScript [13], [14] code and is accessible on the internet at http://ai.pub.ro/education.html yet it had the major disadvantage of not being able to store a statistics in disk files. For this purpose we started developing a second application, this time written in C using the Windows API [15] for the purpose of implementing the adaptive algorithm presented above. The work is still in progress and it also includes updates to the previous application as recommended by our collaborators. Some screenshots with the new application are presented in figures 3 - 8.

In the figures presented some of the differences from application in [7], [12] are observed. First of all, the menus contain fewer graphic as the specialists considered that images on the configuration screen would only distract the attention of the subject. (In [7] it was described that the application was tested in an auditory training center in Bucharest).

Secondly, the configuration screen requests information about the subject currently on test in order to build the statistics it needs for the adaptive behavior – Fig. 3, right side. Also there is an option that let’s the user (or the subject’s supervisor) save the files in a predefined folder – Fig. 6. Finally a more obvious feedback was required after the impaired has chosen the answer – Fig. 5, Fig. 8. Test pages remained unchanged in design – Fig. 4, Fig. 7.
4 Conclusion

During the conducted experiments in [7], [12] it was observed that the novelty of learning through a computer program stimulated the young children’s attention and improved their ability to correctly identify sounds and use the computer. Therefore this study continued the work and improved the application in order to resume the advance towards a complete solution that would eventually prove an efficient therapy tool.

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