A DSP HEARING AID SIMULATOR AND SCREENING TEST

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ABSTRACT - A device to screen hearing impaired people to enable the easier fitting of Hearing Aids is described. It is very flexible and provides a self paced easily controlled method of determining a patient's frequency response preferences. The device has a second function where it can simulate several commercially available hearing aids. This serves as a useful clinical and research tool.

INTRODUCTION

Correctly fitting a hearing aid to a hearing impaired and nontechnical patient is a difficult task. The patient often gives vague and qualitative responses to vital questions, questions intended to determine the correctness of fit of a hearing aid. One method of determining the type of aid to fit, is by using the patient's audiogram from a hearing threshold test. Another method is to determine the patient's preferences with a normal ambient level listening test.

LISTENING TEST

The listening test can take the form of a person reading a prepared text with clear speech at normal listening levels. The patient is then asked to choose between two differently filtered versions of the passage with an A/B test. The patient can switch between the two versions as they like, until they decide which one it is that they prefer. The patient is then presented with another two filtered versions of the same passage, and asked to choose again. This is usually controlled by the audiologist and follows a fixed script. When the test is complete, there is usually an obvious preference for one of the versions. The test can be varied if a trend is clearly seen, or if a result needs to be verified. There can be four different versions of the passage. The first is a lowboost (or emphasis) of frequencies below 2kHz with a slope of -6db per octave, and flat above this frequency. The second is a lowcut characteristic with a slope of +6db per octave below 2kHz and flat above this frequency. The third is another lowcut characteristic with a slope of +12dB per octave below 2kHz and flat above this frequency. The fourth is a flat frequency response. See Figure 1 for the frequency response characteristics. The passage is recorded on four tracks of an audio tape recorder with a different characteristic on each track. The audiologist selects two of the four tracks using the script, and allows the patient to choose between them using an A/B switch. Once the audiologist has finished the script, the patient has been exposed to every permutation, and hopefully will have established a clear preference. This will highlight those aids with a similar characteristic and thus make the selection of the correct aid easier.

NEW DEVICE

The Speech Hearing and Language Research Centre at Macquarie University has developed a similar listening test, but using a new device with enhanced features. The new device performs the same function but uses a novel method. It is based on a computer and uses commercially available enhancements. The computer replaces the four track tape recorder and the prefiltered versions of the passage. The passage can come from an ordinary reel to reel tape recorder, cassette recorder, from a microphone, or any audio source. It can also be prerecorded on the computer's hard disc. The passage is recorded unfiltered. The computer provides the filtering and selects between the different filter characteristics. The audiologist selects the two channels as before (but by using the computer) and the patient chooses between them as before, with an A/B comparison by pressing one of two keys or buttons.

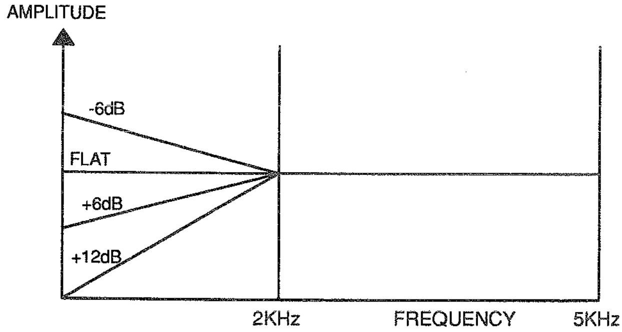

Figure 1. Screening Test Frequency Responses

EQUIPMENT

The heart of the device is a Digital Signal Processor (DSP) from AT&T, the DSP32. This is on a plug in card from Burr Brown (type ZPB32HS) with 64 Kilobytes (64K) of zero wait state memory. It has with it a 16bit Analog to Digital converter (A/D) with anti aliasing filter and a 16bit Digital to Analog (D/A) converter with an interpolating filter. The sampling rate can be changed to several different rates, where 8kHz, 10kHz and 12kHz are the most suitable. The DSP card is plugged into an IBM AT computer which acts as the host. The IBM computer provides the control, and the DSP32 provides the filtering of the audio signal in real time.

PROGRAM

The host IBM computer has a controlling program which selects the two characteristics to be used for the A/B test. This program also allows the patient to switch between the two characteristics with keys or buttons. The control program is written in the C language. It starts and stops the DSP and tells the DSP which filter characteristic to use. The control program monitors the audiologist keyboard in order to assign the selected filter characteristics to the patient buttons. It also monitors the patient buttons to see which of these it will command the DSP to use. It also starts and stops the DSP. A future enhancement to this will be an optional data logging feature, which when enabled, will record the patient's selections as they occur, and the duration of each selection up until the session close. Statistical analysis of this data may help in the audiologist with the selection of the aid, and also may help to determine the effectiveness of any future test evolution.

The DSP has a program which reads the input data from the A/D and performs a Fast Fourier Transform (FFT) on it to determine the frequency content. This is then convolved with the desired filter characteristic, as selected by the host, from one of the four stored in the DSP. The modified frequency content is then turned back into a time domain signal with an Inverse FFT (IFFT) and then output to a speaker, headphone or earpiece with the D/A converter. The DSP program is written in Assembly language. Refer to Figure 2 for a functional block diagram. The audio input signal is sampled at 10khz and placed in an input buffer by the DSP32 Direct Memory Access (DMA) feature. When this buffer is full a new input buffer is filled while the old one is processed. The input buffers contain 128 samples which requires 12.8 milliseconds (mS) to fill. The buffer must be processed within this time, to allow the next one to be processed. The total buffer processing procedure takes 7.198mS of which the FFT and the IFFT are the largest time consumers, 3.209mS and 3.478mS respectively. This leaves plenty of time for control port monitoring. The Helms "overlap adding" buffer processing method is used. When a filter command appears in the control port register, the new filter characteristic is loaded and processing continues.

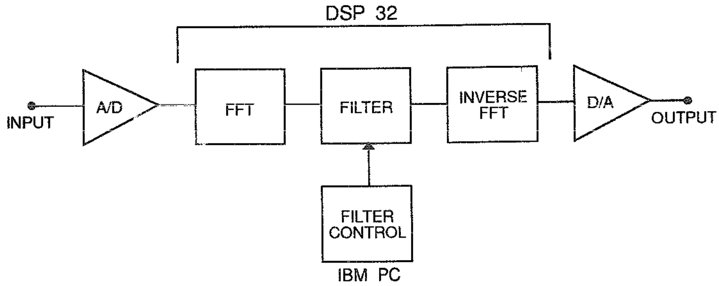

Figure 2. Block Diagram

HEARING AID SIMULATOR

The same hardware can perform other useful functions. With appropriate programs, the device can behave like a hearing aid and simulate its characteristics. Another DSP program, similar to the previous one, performs a close simulation of a hearing aid. This is meant to be used as a clinical tool to further refine the search for the correct hearing aid to fit to a patient. It is also intended to be a research tool, as it can act as a master hearing aid and simulate commercially available aids as well as allowing the audiologist to experiment and test new ideas or combinations.

The DSP hearing aid simulation program loads the filter characteristic from the host. The host has the frequency profiles of commercially available aids, and can simulate different settings and different tone hooks. The frequency profiles and the modifiers used for different settings and different arrangements are stored on disc. These can be retrieved by the host program and sent to the DSP for simulation. The host program allows new profiles to be added to expand the library of commercial units. The new profiles are entered and displayed on the screen so that any corrections can be made. They are then tested and saved. The range of possible settings are stored with them. The characteristics can be added to, modified or combined, and new and experimental features tested. Figure 3 shows a hearing aid profile with different settings. The frequency profiles are entered as gain values verses frequency. These are sent by the host to the DSP, through the parallel command port. The dB values are converted to floating point numbers to be used as the new filter profile.

A basic profile or frequency response is stored for each desired hearing aid model. Each tone control setting, and applicable tone hook are stored as well. Some Phonak aids have five preset gain settings, six tone hooks, and three tone presets. These settings are selected by the control program and then added to the basic stored profile. The total frequency response profile is then sent to the DSP for execution.

The two channel hearing aids can be simulated just as easily as the tone control settings for the simple hearing aids by sending the two profiles to the DSP.

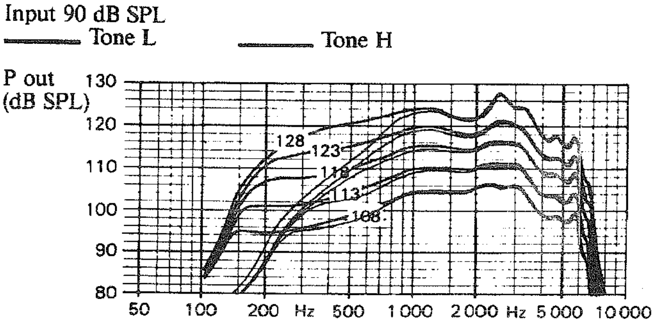

Figure 3. Hearing Aid Frequency Responses

COMPRESSION

Level compression has not been simulated yet. The simulation of input and output compression is merely the adding of two functional blocks. The compression blocks will also be subject to host control. They will be turned off when a simple aid is being simulated. They will be able to be turned on, and supplied with attack and release times as appropriate, and compression ratios.

CONCLUSION

The novel use of a DSP and computer host has led from the improvement of the listening test, to the simulation of commercially available hearing aids. This points to further refining of the hearing aid simulation, and the development and addition of other features. The clinical application will benefit from a more accurate first fitting for the hearing impaired.

ACKNOWLEDGMENTS

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A SPEECH ANNOTATION AND PHONOLOGICAL ANALYSIS PROGRAM

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ABSTRACT - A software system for phonetic annotation and phonological analysis of speech samples is described for application to a longitudinal study of sound change in second language learning.

INTRODUCTION

This paper describes the development of a system for phonetic annotation and phonological analysis of speech samples and discusses methodological issues that arise in connection with its application to a longitudinal study of sound change in second language learning involving Vietnamese immigrants acquiring Australian English. In recent years, a variety of inexpensive microcomputer-based systems for speeech analysis have become available, which meet many of the signal processing requirements for descriptive phonetic and applied linguistic research. More or less contemporaneous with these developments, a variety of multi-tiered text manipulation programs for linguistic analysis have appeared (CHILDES, IT, KWIC-MAJIC, etc.), some of which are specifically designed for phonological applications (e.g.: Pye & Ingram, 1988).

However, for much applied phonetic and linguistic research, the easy manipulation and retrieval of speech signal files and the textual data-base manipulation of their corresponding annotations are equally important and complementary operations. The recoverability of relevant acoustic segments of the original signal files is required for establishing transcription reliability or for investigating signal parameters that correlate with the use of particular annotation symbols. Existing microcomputer-based speech analysis programs have limited annotation facilities that do not meet the needs of phonological analysis, and text-oriented linguistic programs do not permit recovery of the primary speech data. The system reported here provides an **integrated environment** for acoustic phonetic and phonological analysis of speech samples. In this, we believe, it is unique, at least for systems based upon a PC-DOS platform.

The system comprises:

(1) a speech file annotation and segmentation facility, which is part of a sound manipulation hardware/software package (Ultrasound: ULS).

(2) a data base facility for phonological analysis of annotation files (Ultrasound data base: UdB).

THE ANNOTATION FACILITY

Annotation of speech files is performed with a waveform editor, incorporating standard audio-playback and zoom-windowing facilities. Three levels of annotation are provided for:

1. Segment annotations: Single case-sensitive ascii keyboard characters, used to encode I.P.A. symbols. These symbols may be associated with acoustic segments of the signal file.

2. Feature annotations: One to five character feature codes, which, for present purposes, are interpreted as phonetic or phonological process codes, representing connected speech processes or phonetic feature changes caused by second language phonological interference effects. Feature annotations are link-listed to segment annotations.

3. File annotations: A text line of annotation or commentary that is associated with the whole file.

Figure 1, which shows a screen dump of the annotation display and its corresponding pull-down help menu provides an idea of the functionality of the annotation editor.

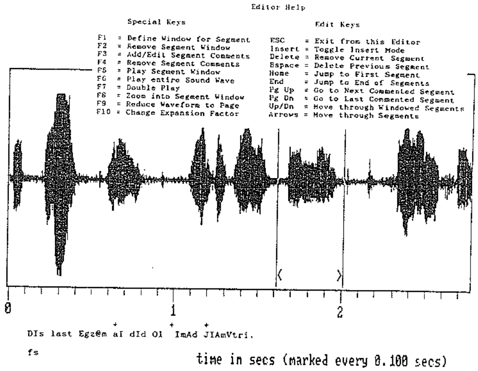

Figure 1: Screen dump of annotation editor and help menu

The annotations are saved as text files, with the same path and filename as their parent signal files, but with a uniform extension (.SEG). The SEG file annotations were originally intended to capture both the conventional **phonological form** of a word, and its **phonetic form** or pronunciation in a given instance or token. Basically, the phonological target of a word is conveyed by its conventional phonemic transcription in IPA symbols, in accordance with entries in the Macquarie dictionary. Phonetic forms are represented as featural annotations, augmenting or modifying the pronunciation indicated by the phonemic transcription.

A conventional phonemic representation corresponds to a broad phonetic transcription, or the approximate pronunciation of a word spoken carefully in isolation, according to generally accepted standards in the speech community. It may therefore be interpreted as an appropriate 'target' pronunciation for language learners and phonetic depatures from it, in terms of