Tactile Auditory Sensory Substitution

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Abstract

High frequency hearing loss is a problem common among people of all age groups. People who suffer from this type of hearing loss often lose the ability to hear certain high frequency consonant sounds, which include some of the most common consonants in the English language. Thus, this type of hearing loss, which is the most common type, is detrimental to everyday communication. The goal of this project is to use sensory substitution, a technique for presenting environmental information missing in one sensory modality to another, to help replace this missing high frequency information. Our designed sensory substitution device utilizes a microphone input, digital signal processor (DSP), and handheld vibrotactile unit to inform the user of high frequency patterns in real-time. An acoustic signal is detected by the microphone and transmitted to the DSP which filters the signal into four different channels based on frequency, compares the energy in each band to a threshold value, and then outputs all four channels to an external circuit. The circuit modifies the current and voltage levels from the DSP in order to run the transducers and provides a physical connection to the vibrotactile unit. The four transducers in the handheld unit vibrate in response to the amount of energy in each specific filtered band of high frequency sound. Different fricative sounds will stimulate different vibrotactile transducers based on the frequency of the sound, allowing the user to associate a particular spatio-temporal pattern of vibration with an unheard sound. Therefore, based on which finger is stimulated with the vibration of the motors, the user would be able to tell which high frequency consonant was being communicated. Use of this device, to supplement for the loss of high frequency hearing, should aid the user in daily communications in regard to speech and hearing.

Product Design Specifications Summary

The final design should meet requirements in the areas of performance, safety, cost, and ergonomics. It should help to substitute for high frequency hearing loss to the extent of helping the user with everyday communication. The system should be able to detect speech spoken at the normal 60 dB level and separate it into four tunable frequency bands above 1500 Hz. It should also be able to transmit vibrations at a frequency of 250 Hz resulting in a skin displacement of 1 μm. For clinical testing purposes the device should be comfortable, easy to use, and cause no harm to the user. The device should be safe such that the transducer terminals are insulated from the skin and the electronics powering the system should be isolated from the user. The product design specifications can be viewed in its entirety in Appendix I.

Background

High Frequency Hearing Loss

The number of hearing impaired Americans has more than doubled in the past 30 years with nearly 50 percent of Americans over the age of 65 affected.[1] It not only affects the elderly however, 1.4 million children under the age of 18 also have a hearing condition.[2] The two most common types of hearing loss are conductive and sensorineural. Conductive hearing loss is defined as the condition when sound is not transmitted correctly through the middle ear and into the inner ear. It is often described as having the ears plugged all day. It can be caused by wax buildup or even infection. This type of hearing loss can often be medically cured.

The most common type of hearing loss is sensorineural. About 90 percent of individuals who are hearing impaired have sensorineural hearing loss.[1] This condition, also known as nerve deafness, consists of either damage to the inner ear or the nerves which transmit the messages from the ear to the brain. It is caused by disease, birth injury, or even aging. The most common form of sensorineural hearing loss is high frequency hearing loss. This is where an individual loses the ability to hear certain high frequency constants such as Sh, S, T, Th, P, or F sounds, as seen in Figure 1. Since these are some of the most commonly used consonants in the English language, high frequency hearing loss is truly detrimental to every day communication.

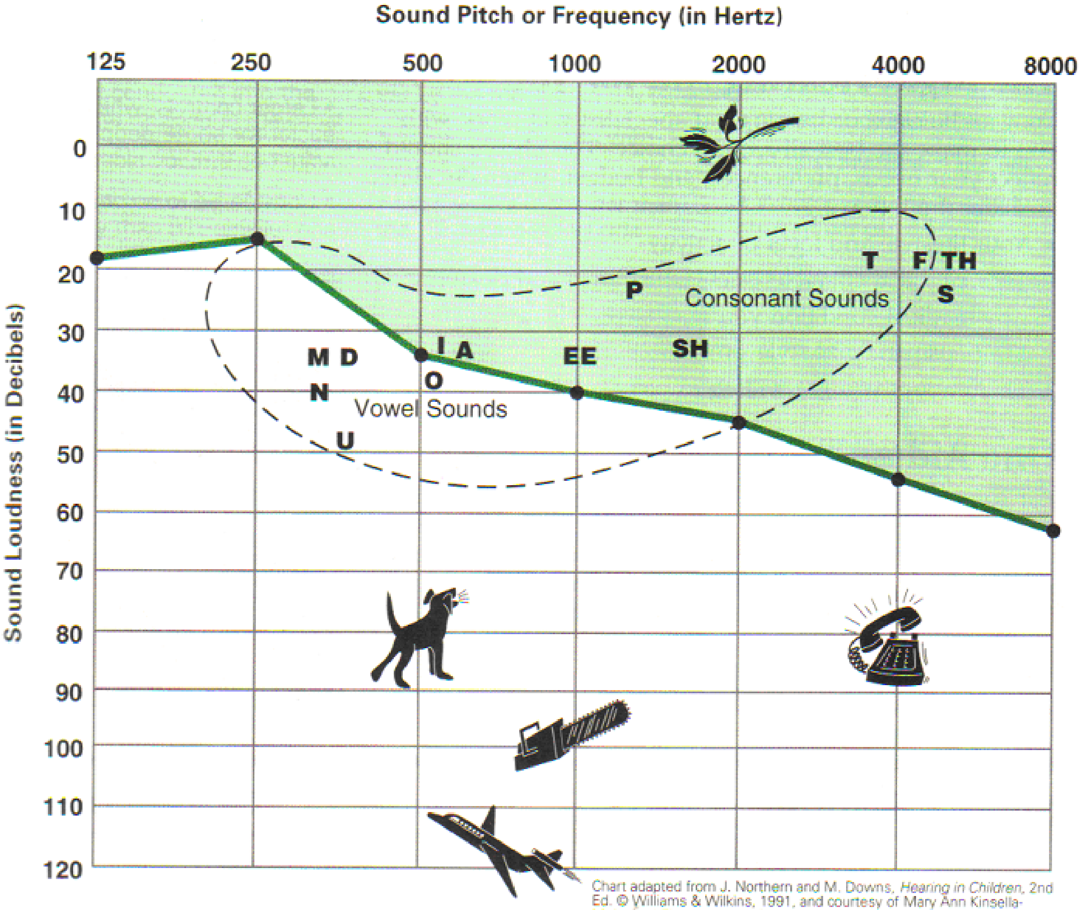

Chart adapted from J. Northern and M. Downs, *Hearing in Children*, 2nd Ed. © Williams & Wilkins, 1991, and courtesy of Mary Ann Kinsella-Meier and Florence Vold, Gallaudet University, 1985.

Audiogram of a person with a hearing loss that is worse in the higher frequencies: This person heard none of the sounds in the green area above the line.

Fig. 1: Audiogram of missing consonant frequencies[3]

Most elderly Americans suffer from some form of high frequency hearing loss, along with 14.9 percent of children.[1] High frequency hearing loss is not easily medically fixed. Hearing aids do not do an adequate job of resolving the problem of high frequency hearing loss because they only amplify the sound. This amplification is only useful when there is a deficit but not total loss, as in high frequency hearing loss, since the actual nerve that receives these high frequencies is dead. Consider a piano with no strings. No matter how hard the keys are hit, there is still not going to be any sound. Similarly, no matter how loud the hearing aid makes the sound, a person with high frequency hearing loss cannot hear sounds in those frequencies. Obviously, something more than hearing aids needs to be considered in order to help the rising number of people with high frequency hearing loss.

Sensory Substitution

Sensory substitution is presenting environmental information absent in one sensory modality to another. Sensory substitution can be seen in many actions people perform throughout their everyday lives. For example, a person substitutes the sense of touch for sight when reaching into their pocket to retrieve an object. Other common examples are the use of sign language to substitute vision for hearing or Braille which substitutes the sense of touch for sight (specifically for reading). In this project, the device will substitute high frequency sounds with vibrotactile stimulation, thus substitute touch for part of the hearing spectrum.

Existing Sensory Substitution Products

There are a few existing devices which aim to use sensory substitution by substituting for hearing using vibrotactile or electrotactile stimulation, but these products are made specifically for people with complete hearing loss. For example, the Tickle Talker™ uses vibrotactile stimulation on the fingers of the user through the use of a glove. Each finger receives stimulation from a different range of frequency and based on the pattern, strength, and duration of the vibration, the user can pick out a certain frequency range. Since this form of sensory substitution covers the entire spectrum of hearing, it is not sensible for a person with only high frequency hearing loss to use the device. They would have to sift through way too much information to get the signals they needed. With such a large learning curve, the user must spend hours with the device to learn minimal amounts of words. For example, after more than 40 hours of training, a certain user could only identify 70 words.[4] The user with only high frequency hearing loss could adapt at a much faster rate to only a high frequency stimulator because they are only missing certain sounds, not the entire spectrum.

Fig. 2: *The Tactaid VII* [6]

Other similar devices include the Tacticon 1600 and the Tactaid VII. The Tacticon 1600 uses electro stimulation by putting electrodes on a belt around the user's abdomen.[5] This device also covers the whole spectrum of hearing and has a steep learning curve. In similar ways, the Tactaid VII, seen in Figure 2, uses vibrations covering the entire range of human hearing.[5] The vibrators are attached to the user's sternum, each corresponding to a certain frequency range. Since each existing device attempting to substitute hearing for either vibro or electrotactile stimulation attempts to substitute for the entire range of hearing, it is not sensible for use of helping the most common type of hearing loss – high frequency hearing loss. Thus, a device which only substitutes for these high frequencies would be very useful and in high demand.

Final Design

Design Overview

The final prototype consists of a microphone, DSP with custom daughterboard, and a handheld vibrotactile unit. The microphone utilizes an analog input on the DSP to convert the inputted sound into a digital signal. The signal is then filtered into four separate bandwidths, 1.6 – 2 kHz, 2 – 3 kHz, 3 – 3.5 kHz, and 4.5 – 8.0 kHz, using real-time digital filters. The amount of energy in each bandwidth is compared to a threshold value, which is calculated as a percentage of the maximum value of the current input signal. If the energy in a band is greater than the threshold it triggers one of the four digital outputs on the DSP to jump from 0 V to 3.3 V. This digital output voltage is then modified on an external circuit board in order to be able to drive the vibrotactile transducers. The circuit then outputs the voltage via a parallel cable to four transducers housed in a mobile handheld unit. The transducers are embedded in rubber foam which is attached to memory foam in order to dampen the vibrations along the unit itself and increase the user's ability to discern the vibration patterns. The user is then able to incorporate the vibration patterns with the lower frequency sounds they are able to hear in order to hopefully increase their word recognition ability.

Channel Frequencies

The current channel frequency ranges were selected after analyzing segments of recorded speech sounds and inspecting the high frequency components they contain. Speech samples were obtained from a variety of sources including recordings of male and female speakers as well as the prerecorded California Consonant Test. The samples were then analyzed using COLEA, a Matlab software tool designed specifically for speech analysis. Using COLEA, consonant sounds in speech samples were located and isolated. Power spectral plots were then made over the duration of single consonant sounds to analyze the frequency components they contain. The spectral plots of the consonant sounds were then analyzed by inspection to determine the general frequency nature of each sound.

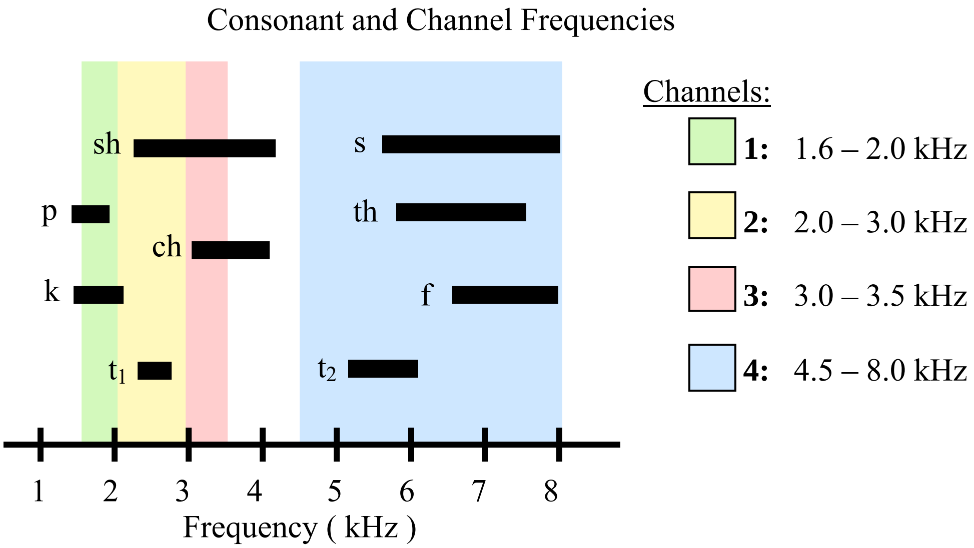

Fig. 3: A plot of experimental consonant frequencies overlaying frequencies of the four channels.

The analysis of high frequency consonants revealed that most of the consonants span a wide frequency range, but usually have much narrower frequency range where the energy is most concentrated, appearing as a spike in spectral plot. These spectral peaks were used to define the consonant sounds for design purposes as well as targets for detection in a speech signal. Also, there was little variation observed from speaker to speaker. This is most likely a result of how high frequency consonant sounds are formed by the speaker. Spoken vowel sounds rely mostly on the resonance of the speaker's voice to generate a sound that is then "shaped" by the speaker's mouth and lips. In contrast, consonant sounds are generated by a speaker forcing air though a closure formed by their mouth or lips, so there is little contribution from the vocal resonance that naturally varies from speaker to speaker, most notably for speakers of different sex.

Fig. 3 shows the frequency spans of the high frequency consonant sounds determined through experimentation. Also displayed are the frequency ranges that were selected for the four channels of the device. These ranges were selected in an attempt to allow as many consonant distinctions as possible. The main goal was to ensure the ability to distinguish and "s" from an "sh," since this distinction is considered to be the most important for speech intelligibility.

Digital Signal Processing

Unlike in previous designs where the signal processing of the system was performed on commercial software loaded onto a laptop, the current prototype utilizes the real-time signal processing capabilities of a Texas Instruments© TMS320C6713 Digital Signal Processing Starter Kit (DSK). Whereas in previous semesters, audio files needed to be prerecorded, filtered using the Cool Edit Software, and then saved into four separate files before being played back, the system with the DSK is standalone and real-time.

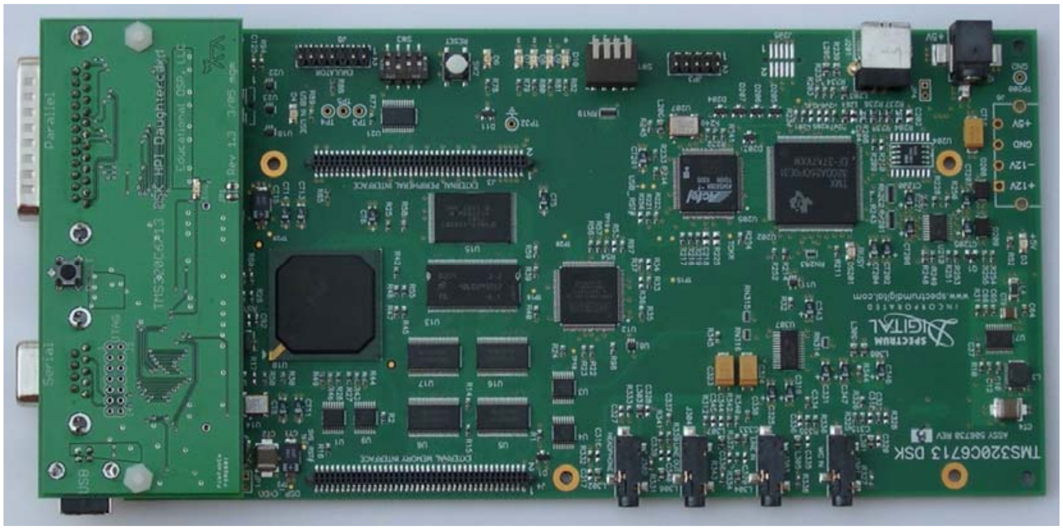

Fig. 4: Top view of the TI TMS320C6713 Digitial Signal Processing Starter Kit

The DSK 6713 is a fully programmable interface with 16 megabytes of SDRAM where programs can be stored and run. Development for the DSK is traditionally done in the C programming language, but environments have been developed to program the board with Matlab and LabVIEW languages. In addition to the on-board memory, the DSK also features 2 analog inputs through 3.5 mm audio jacks, 2 analog outputs through 3.5 mm audio jacks, 4 DIP digital input switches and 4 digital LED outputs. The board is powered by its own DC power supply that is rated for 5 V, 3A at maximal power. Once the program is written, interpreted and compiled on a PC, it is exported through a USB connection to the DSK. The programming environment chosen to program the DSK for this project was the LabVIEW programming environment because of its relative ease of use in programming algorithms that require virtual object blocks that interconnect with multiple other blocks.

To specifically only use the functions that are available on the DSK chip when programming in LabVIEW, the LabVIEW DSP module needed to be installed first. Once the DSP module targeted the DSK 6713, LabVIEW recognized the options that were available for the board and the program could be written.

To make a program that ran continuously, the algorithm was created within a while loop construct that forced the inputs to continuously sample new data. The microphone input of the DSK board was used as the input into the program and the parameters were set to sample at 48000 samples/s. This prototype is only interested in frequencies up to 8 kHz, and as such, only a 16 kHz sampling frequency is required in accordance with the Nyquist sampling theorem, but this would have required a low pass antialiasing filter before sampling. Depending on the characteristics of the antialiasing filter, important data could be lost before reaching the signal processing stages. A higher sampling rate requires more computational power by the DSK, which was found to be able to slow down the system to a state that was no longer real-time, however, no important data would be lost. A 48 kHz sampling rate was discovered to be the best compromise between performance and functionality.

The analog input was sent into a bank of filters that were designed to separate out important high frequency consonant sounds such as the “s” and “ch” into four distinct channels. The filtering functions that were used in this bank were finite impulse response (FIR) digital bandpass filters that were chosen because of their computational efficiency.

FIR digital filtering is a method of filtering a discrete time input by multiplying the current value of the input by a certain weighting constant, and the previous values by other weighting constants to overall create the effect of passing only selected frequencies of the signal.[7] The resulting output to input relationship, or transfer characteristic of each filter system is: $H(z) = b_0 + b_1z^{-1} + b_2z^{-2} + \ldots + b_Nz^{-N}$ where z^{-N} is the Nth delayed term and b is the weighting constant. With the increased number of weighting constants, the filter cutoff frequencies become sharper and greater stopband attenuation is reached within a smaller frequency range outside of the cutoff frequencies. With LabVIEW implementation of the digital FIR filters, processing time increased with the number of weighted constants used in the filters, so a compromise was made to maximize the number of weighted constants (101 weights/channel) while still maintaining real-time operation. FIR filters were chose instead of infinite impulse response (IIR) filters because the DSK had problems computing the rational functions necessary to implement an IIR transfer characteristic. Even though, sharper cutoff frequencies could be obtained with less weighting constants by implementing the rational transfer characteristic given by Equation 1. The DSK was more easily able to compute more weighting constants in a polynomial transfer characteristic (FIR) compared to less weighting constants in a rational transfer characteristic (IIR).

$$H(z) = \frac{\sum_{i=0}^{n} a_i z^{-i}}{1 - \sum_{i=1}^{n} b_i z^{-i}} = \frac{a_0 + a_1 z^{-1} + a_2 z^{-2} + \ldots + a_n z^{-n}}{1 - b_1 z^{-1} - b_2 z^{-2} - \ldots - b_n z^{-n}} = \frac{Y(z)}{X(z)} \qquad \text{Eq. 1}$$

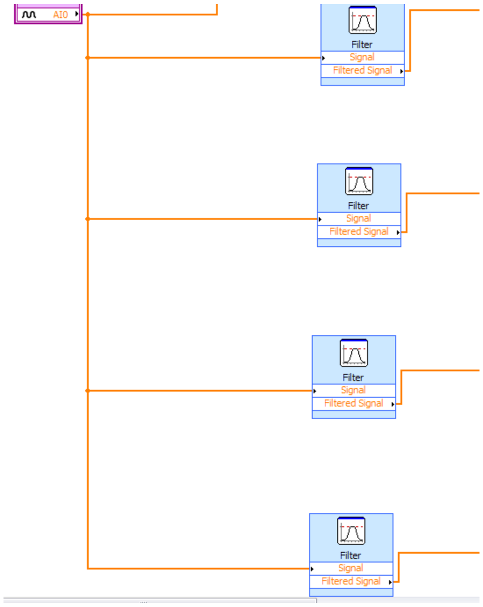

Fig. 5: The analog input shown leading to the four FIR bandpass filter banks

The output of each filter is wired to its own comparator function which compares the amplitude of the filtered output to a threshold value to determine if there is enough energy within the channel to warrant activation of the corresponding motor. Unlike in previous prototypes where the threshold value was constant, this system utilizes an adaptive threshold that is dependent on the amplitude of input, as shown in Fig. 6.

The adaptive thresholding function operates by first rectifying the input using the Absolute Value function and then looking at the rectified input. The maximum rectified input of the last several values were calculated by using the Array Max Min function. This value is multiplied by 15 percent, which was found to give the most effective indication of the presence high frequency consonant sounds. A minimum threshold value was set at an offset of 50 amplitude units to always give a nonzero threshold and prevent constant activation of the motors by low amplitude noise. The greater thresholding value is then determined by another comparator function and fed through a switching function to use as the comparison level for each of the four channels. This thresholding algorithm is a great improvement over the constant threshold used in previous designs because of its ability to account for volume differences in the input. The system no longer activates motors by simply speaking loudly into the microphone.

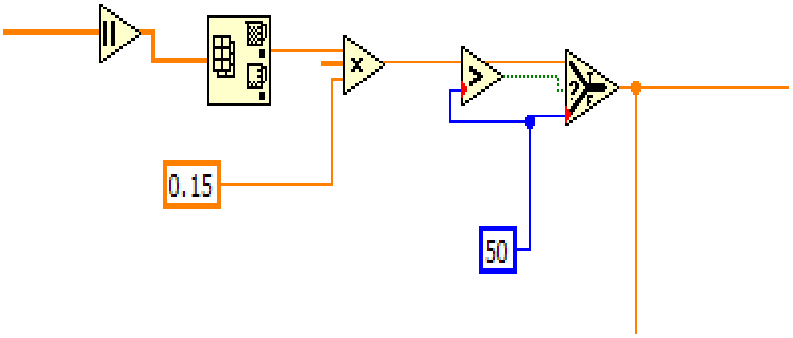

Fig 6: Implementation of the adaptive algorithm using an Absolute zero, Array Min Max, Multiply, Comparator and the Switch functions, respectively.

Ultimately, the output of each comparator, after comparison with the dynamic threshold, gives a Boolean constant output indicative of whether there was enough spectral energy in that channel. This Boolean varies with the consonant sound input and therefore, is appended to an array for storage. The Array Element function parses the most recent value of this stored Boolean array and uses it to drive the LED outputs on the DSK board.

The LED logic algorithm necessary to run the corresponding motor when there is significant energy in the channel is inverted to give the proper voltage outputs. A disadvantage of using LabVIEW is that it cannot directly interface with serial and parallel port outputs that can be added through a commonly used daughterboard of the DSK 6713. Consequently, the LED digital outputs of the board had to be used off the board to drive the motors. The logic operates in the following way: when there is not significant energy in the channel, the LED remains on giving a 0 V signal to the motor circuit (off) and when there is significant energy in the channel, the LED turns off giving a 3.3 V signal to the motor circuit (on). This reverse logic was implemented using if else statements in the LabVIEW code, as shown in Fig. 7. For loops were embedded within the case statements that turned the motor on in order to keep the motor on for an extended number of clock cycles (20 000) in order to give a longer duration on signal to the motors.

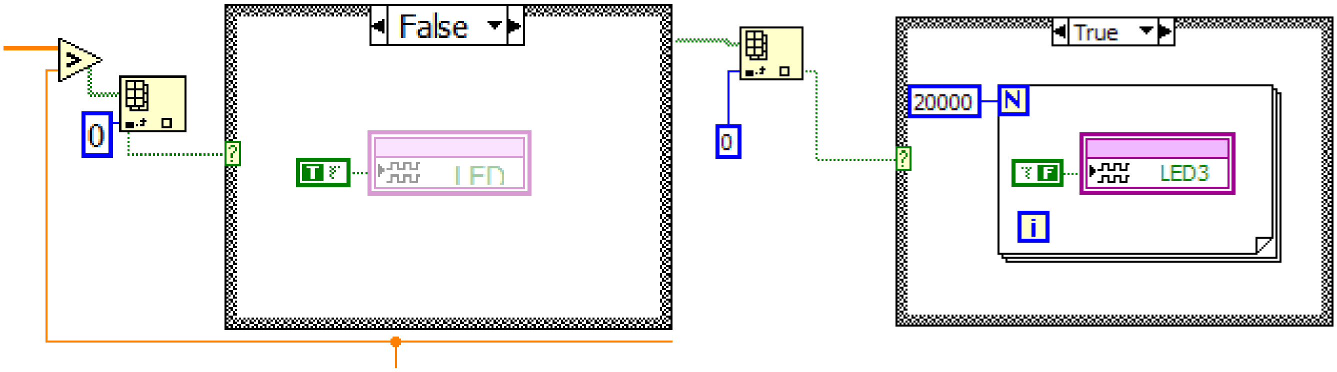

Fig. 7: LED logic implementation with case statements for when the motor should be off (left) and when the motor should be on (right). The "on" case statement also includes a for loop to extend the duration of the motor "on" time.

Once the proper logic is established by the if/else case statements, the signal is routed to a custom-built daughterboard that amplifies the current to a level that is necessary to run the vibrating motors, a circuit diagram of which is shown in Fig. 8. Also, the transistors used for the current amplification (NPN 2904) protect the LEDs on the DSK board from sinking any of the current from the motor power supply. The motors were powered directly from the 5 V power supply of the DSK board so that the system was completely self-contained and no external battery was required. A parallel port was created to serve as the interconnect between the DSK board and the housing unit which allows for simple attachment and detachment of the housing. Also, future modifications to the motor housing can be made without having to modify the daughterboard.

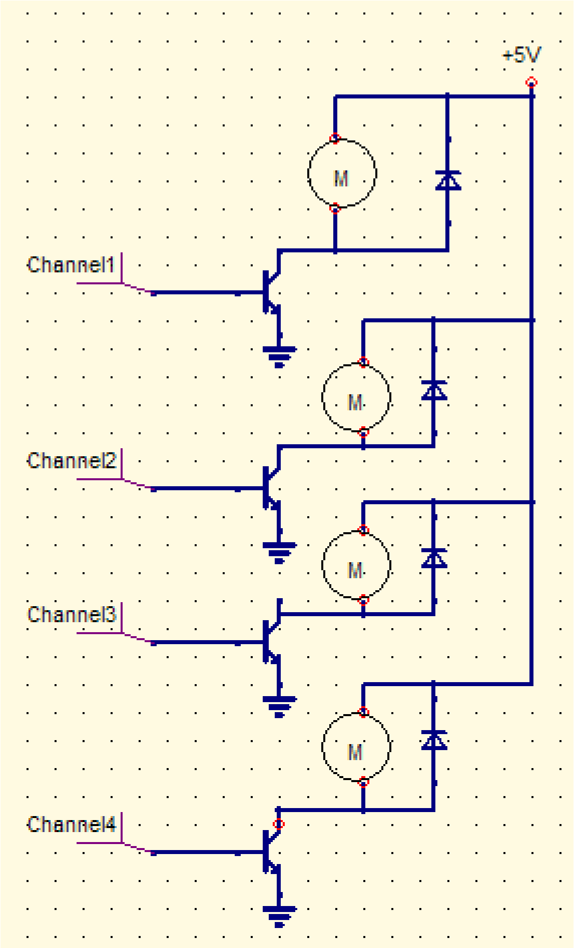

Fig. 8: Circuit schematic of custom daughterboard with transistor current amplification and protection diodes.

Vibrotactile Transducers

Fig. 9: Solarbotics VPM2 Motor[9]

Two different areas were researched for the tactile stimulation part of the prototype, vibrotactile and electrotactile. Vibrotactile stimulation involves generating vibrations that activate mechanoreceptors in the skin while electrotactile stimulation passes current through the skin which directly stimulates afferent nerves. For our final prototype, four Solarbotics VPM2 flat disk motors were chosen to alert the user of missing high frequency information, these motors are shown in Fig. 9. Vibrotactile transducers were chosen because of their ease of acquirement and implementation with the tradeoff of having higher power consumption.

The amount of vibration that is appropriate for the user is determined by the characteristics of mechanoreceptors known as Pacinian Corpuscles. These mechanoreceptors are sensitive to gross pressure changes and higher frequency vibrations applied to the skin. They are also classified as fast afferent type II (FA II), referring to their ability to adapt after sending an action potential.[8] It has been shown that the minimum threshold amplitude needed to stimulate these receptors varies with the frequency of vibration. In one study, the minimum threshold for the average subject was measured to be 1 mm at 5 Hz and 1 μm at 200 Hz. These values are indicative of a trend that at higher frequencies, less amplitude of skin displacement is needed to activate

Pacinian Corpuscles. The optimal range of sensitivity, however, was discovered to be in the range of 60 Hz to 250 Hz.[8]

The Solarbotics VPM2 flat disk motors exhibit several key features which made them advantageous for implementation in the final design. The sensation that the user perceives is more consistent with a mechanical stimulus because the transducer is only displacing the skin as opposed to sending currents through the skin in the case of electrotactile stimulation. The small size of the motors, 12 mm in diameter and 3.4 mm thick, lends to increased discrimination between separate motors. Also, the motors vibrate at a frequency of approximately 200 Hz which is in the ideal perceptive frequency range presented earlier. Another positive aspect of these motors is their relatively low cost, $4.00 each, coupled with there ease of implementation. One minor disadvantage was the relatively large power consumption, drawing 40 mA at 2.5 V for a power rating of approximately 100 mW.[9]

Placement of Transducers

Since the Pacinian corpuscle senses slight pressure and vibrations, the transducers must be able to stimulate the Pacinian corpuscles, which are of different prevalence in different parts of the body. Two point discrimination threshold tests are often performed to determine the sensitivity of a certain area of the body, thus determining the amount of mechanoreceptors in that particular region. For example, in order to feel separate stimuli, the stimuli must be 45 mm apart on the calf, 35 mm apart on the forehead, 10 mm apart on the palm, and 2-3 mm apart on the fingers. Two point discrimination tests were used to acquire this data.

In the final design of last semester, it was determined the motors would be placed behind the ear. This was done for a number of reasons. This position would allow for the device to be nearly completely concealed from outsiders looking from the front or the opposite side. Another advantage of this placement is due to the decreased mobility of the region, the user’s everyday movements would not be affected by the device placed behind the ear. Vice versa, the user’s movements would not interfere with the sensing of vibrations. A complication was caused by bone conduction of the mastoid process which made spatial discrimination more difficult. Another disadvantage of this placement was that its attachment might have been impeded by the hair of the user, and some sort of adhesive had to be used which could have been an annoyance to the user.

After choosing the area behind the ear to house the transducers last semester, the sensitivity of the region was tested to find the optimal distance between the tactors, as well as the exact locations in that region that each transducer would be placed. This data was found by using a two point discrimination test. It was found that there should be two tactors on the mastoid process and two on the skin below it. The most advantageous distance between each tactor was also determined. It was determined that stimulations needed to be 25 mm apart on the mastoid process and tactors on the skin below the mastoid process also needed to be 25 mm apart. The last tactor on the mastoid process and the first one on the skin below it needed only to be 20 mm apart. The design from last semester can be seen in Fig 10.

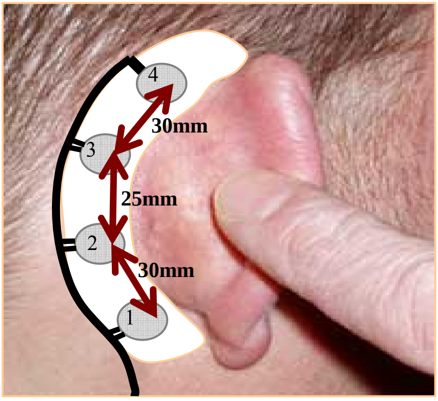

Fig. 10: The housing unit for the transducers and the distances between them.

However, after the prototype was built and tested, it was discovered that it was very difficult to discriminate between each motor. For example, if Tactor 1 were to fire, the user should not be able to feel the entire device vibrate and think it might be Tactor 4 firing. Also, the user should be able to easily tell which tactor is vibrating based on the sensitivity of the skin. This idea, which is paramount to the project idea itself, was not held up with the motors placed behind the ear. This could have been due to the bone conduction through the mastoid process, or simply that there was not enough space behind the ear to keep the motors the necessary distance apart because of the insensitivity of the skin in that region. Also, the two point discrimination test results show the placement of two nonmoving stimuli. This could mean that the results could not be generalized to vibrating motors placed on the skin. Because the tactors could not be distinguished from one another, placement needed to be reevaluated since if the user cannot discriminate between the motors, they then would not be able to discriminate frequency bands, and the efficacy of the design would therefore be in question.

Another important aspect of placement this semester was the concept of invasiveness. Since the main purpose of this semester was to have the prototype be able to be used in a testing setting, it needed to be as noninvasive as possible. This would extremely benefit the subject as well as be advantageous for getting IRB approval for human testing. This is because the less invasive a test is, the less likely the patient will become uncomfortable, and therefore the more likely the testing will gain IRB approval. With the device located behind the ear, some kind of adhesive would have had to of been used. Adhesives are very invasive since they have to be applied, washed off, and then that same device which was stuck to one person's head would then have to be applied to another person's head. The adhesive could also hurt the user when being peeling off since in a testing environment there could not be a more permanent adhesive so it would be needed to be pulled off after testing each subject. Also, a place on the body which does not make the subject feel uncomfortable is needed for placement of the tactors. Lifting up the shirt to put the device on the abdomen, or strapping something around the user's thigh or upper arm would make the subject feel uncomfortable in this type of setting.

Putting together all of these important concepts, the hand was found to be the optimal place to use for testing purposes. First, the fingers are one of the most sensitive parts of the human body. The pad of the finger has 4.2 times the density of mechanoreceptors as the palm, which is also quite sensitive in comparison to behind the ear for example. With regards to the two point discrimination test, the fingers are 10 times more sensitive than the area behind the ear. Since the pads of the fingers are not directly connected to each other, a separate channel can be easily sent to each finger. Taking advantage of the fingers great sensitivity to vibrations allows for easy discrimination between the motors, as long as vibrations are not transmitted through the device itself. Secondly, this is a very noninvasive part of the body. There would be no need for adhesive, nothing to strap around any part of the body, no need to lift up any clothing, or in anyway make the subject feel uncomfortable. Also, if the subject were to feel uncomfortable because of the vibrations for any reason, he or she could simply lift their hand off of the device.

Housing Unit for Transducers

Fig. 11: Handheld device.

The vibrotactile transducers need to be in a vibration dampening environment in order to decrease the chances of interference from one transducer on another. The material that the motors are embedded in must also be flexible so it can bend with the natural curve of the finger and be comfortable to hold in the hand. Also, it must be tough and long lasting. To satisfy these requirements, sponge rubber was chosen to house the vibrators. It is flexible so it can curve around the cylinder, yet it is firm and tough so it will be long lasting. It is also shock absorbing to dampen vibrations. However, after initially building the prototype with only sponge rubber, vibrations were being transferred through the sponge rubber and throughout the device. Thus discrimination of the tactors was nearly impossible. In order to fix this problem, viscoelastic memory foam was used. This foam consists of billions of high density, viscoelastic memory cells which not only are soft and conform to curvature, but are excellent at absorbing vibrations. A layer an inch and a half thick was placed between the motors embedded in the sponge rubber and the plastic cylinder. Furthermore, there are cuts in the cylinder beneath the memory foam in order to further reduce the transfer of vibrations. With the hand resting on the cylinder and the finger pads resting on the motors, there is very good discrimination between the motors, thus the user can tell which frequency range is present in communication.

As can be seen in Fig 11, the sponge rubber tape embedding the motors is cut with grooves for better comfort for the fingers leading up to the motor. The motors meant for the pinky and pointer fingers are located slightly before the motors meant for the middle and ring fingers in order to make the device a more comfortable fit for human hand. Beneath the sponge rubber is the yellow layer of memory foam which is connected to the plastic cylinder. Wires run through the memory foam into the cylinder where they are soldered and formed into one cable in order to compact all of the speaker wire which is used to connect directly to each motor.

Testing

IRB:

In order to perform testing on human subjects we must receive IRB (Initial Review Board) approval. Once we receive IRB approval, we will test the efficacy of our device by conducting a discrimination test and a California Consonant Test on our subjects, which have both normal and high frequency hearing loss.

Subject Selection:

The subjects of this study will be individuals with high frequency hearing loss (hearing loss greater than 30 dB above 1 kHz). The study is expected to enroll approximately 10 subjects with high frequency hearing loss. Subjects with high frequency hearing loss will be recruited through referrals from Dr. Veronica Heide, Au.D.. Dr. Heide will ask her patients if they are interested in participating in our study, there will be no affiliation between Audible Difference and our project. However, these subjects will be from the clinical population she regularly sees at Audible Difference. Audible Difference is a private company which provides complete audiological services in Madison, WI. Dr. Heide will initiate the discussion with a potential candidate, and if the individual expresses an interest, they will be given the recruitment flier (attached), and invited to contact the investigators. This is intended to ensure that potential subjects do not feel compelled to volunteer at the time of their visit with Dr. Heide.

The initial recruitment for this study will be the only interaction that our subjects will have with Dr. Heide. In our study, we will be excluding people with communicable diseases that could be passed on to another subject and any problems that affect their ability to use the device. Health status questionnaire questions are asked to determine if they should be excluded for one of these reasons. Subjects will serve as their own control. In addition, all subjects will be at least 18 years old, and will be cognitively intact with no other known hearing problems.

Experimental Procedure:

Subjects with high-frequency hearing loss (HFHL) will be presented with dynamic vibrotactile patterns on their fingertips. These patterns will correspond to combinations of the different frequency bands in which large amplitude high frequency sound is present. The first step will be to train the subject to recognize a specific consonant’s vibration pattern. Presentation of the consonants in the form of vowel-consonant-vowel “nonsense syllables” such as "A"-"CH"-"A" will hopefully teach the user the distinct pattern which identifies the consonant sound, in this case –"CH". The distinct pattern of this consonant sound is vibration of the second and third tactile stimulators, corresponding to the middle two frequency bands, from 2-3.5 kHz and 4-6 kHz, in the filtered 1.6 – 8 kHz region, where most of the consonant’s high frequency information is present.

After learning the vibration patterns associated with specific consonant sounds the subject will hopefully be able to integrate this new information in order to increase word recognition ability, as measured by the California Consonant Test (CCT). The California Consonant Test or CCT is an established test for subjects with high frequency hearing loss, by testing their word recognition ability. The CCT is a one hundred multiple choice question exam that involves playing an audio recording of words each stressing high frequency consonants sounds. Each word will be presented to the subject at 65 dBA from a single computer speaker while a filtered version of the word is presented simultaneously via four tactile stimulators placed on the fingertips. The subject will be asked to identify the word presented from a list of four possible choices. It is hoped that the subject will be able to properly identify a higher percentage of target words with use of the tactile substitution device. There is an established grading rubric for the CCT that will be used in our analysis. If the subject selects the correct answer then they will be awarded one point, however if the answer is wrong then they will receive zero points.

Data Analysis:

The majority of our analyses will be conducted "within-subjects". That is, each subject is exposed to all experimental conditions, and differences between the experimental conditions across subjects are statistically analyzed—we are interested in whether or not it allows subjects to discern and integrate high frequency consonant sounds with practice. Our study will employ standard word recognition tests which present the stimuli in random order multiple times. We believe that a relatively small subject population will yield the necessary data to demonstrate efficacy of the device in increasing phoneme recognition, improving word discrimination, and potentially improving the speech comprehension ability of HFHL subjects. Previous studies assessing the ability of various hearing aids to improve high frequency word recognition have used approximately 10 subjects.

Future Work

Prototype Testing:

Once we have received IRB approval, which we aim to get by the beginning of the spring semester, we will begin our human testing using both a discrimination test and the CCT. The data from the tests will be analyzed and recorded, to determine the effectiveness of our device in assisting individuals with HFHL in everyday communications. In addition, we need to make modifications to the housing of our device such that it will be an effective testing prototype. To accomplish this, the device will be made sturdier by eliminating the exposure of the soft foam. A Solidworks picture of how we intend to accomplish this can be seen in Fig 12.

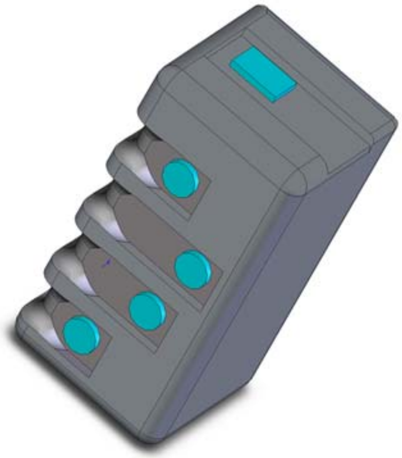

Fig. 12: Proposed prototype for future testing

Commercial Prototyping:

In order to go into production, three main modifications need to be implemented to the device design. First, the device must be capable of running on battery power, so that it can be portable. In addition, the device must miniaturized, the DSP and the circuit components. The goal is that these components will eventually be able to fit within the housing component chosen for the device. Finally, the location that the device will be placed on the subject's body is under additional consideration. While we feel that a handheld device would be best for testing, we do not necessarily feel that this is an effective way for people with HFHL to utilize our device in everyday communications.

References

[1] American Speech-Language Hearing Association (ASHA). The Prevalence and Incidence of Hearing Loss in Children. 1997-2006. http://www.asha.org/public/hearing/disorders/children.htm

[2] Better Hearing Institute (BHI). Hearing Loss- The Prevalence of Hearing Loss. 2005-2006. http://www.betterhearing.org/hearing_loss/prevalence.cfmC. J. Poletto and

[3] Krames Communications. (1995). Hearing Aids. [Brochure]. San Bruno, CA.

[4] Galvin, K. et al (Aug 1999). Acquisition of a tactile-alone vocabulary by normally hearing users of the Tickle Talker™. *The Journal of the Acoustical Society of America*, Vol. 106, Issue 2, pp. 1084-1089.

[5] Lynch, M., Eilers, R., (April 1992). Open Set Word Identification by an Adult with Profound Hearing Impairments: Integration of Touch, Aided Hearing, and Speech Reading. *Journal of Speech & Hearing Research*, April 1, 1992, Vol. 35, Issue 2.

[6] Audiological Engineering Corp. (n.d.) Tactaid 7. Retrieved 5 March 2007 from http://www.tactaid.com/tactaid71.html.

[7] Tompkins, W editor. Biomedical Digital Signal Processing. Prentice Hall, New Jersey. pp 100-125. 2000

[8] Kanyo, M. et al. 2005. A Tactile Synthesis Method Using Multiple Frequency Vibrations for Representing Virtual Touch. *IEEE/RSJ*. p 1121 – 1127.

[9] Solarbotics VPM2 Flat Type Vibration Motor Specifications Sheet

Appendix I: Product Design Specifications

Project Design Specifications
Tactile Auditory Sensory Substitution

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Function:

High frequency hearing loss is the most common form of hearing loss experienced. It is caused by damaged nerve ends on the hairs in the cochlea. People with high frequency hearing loss cannot hear high frequency sounds such as ‘s’, ‘f’, ‘sh’, ‘ch’, ‘t’, and ‘p’ sounds. Since these sounds are some of the most commonly used in the English language, high frequency hearing loss immensely inhibits communication.

Individuals with high frequency hearing loss tend to rely on hearing aids, which amplify sounds. However, the hearing loss cannot be fixed with amplification of these high frequency consonants because the person simply cannot hear in that frequency range, the volume does not make a difference. Instead of amplification, these missing consonants can be communicated by sensory substitution. In other words, a sense other than hearing can be used to send the auditory information to the brain. The goal is to design and develop an auditory substitution device that through the use of vibro-tactile stimulation can substitute for regional frequency hearing loss. Individual vibrators will represent certain frequency ranges, and when the user feels the specific vibration attributed to certain frequency range, the user will be able to identify the sound.

After using the device for an extended period of time, the user will no longer have to concentrate on the vibrations but will be able to “hear” the high frequency sounds. Due to the plasticity of the brain, the brain will interpret the vibrations as sounds and fill in the gaps for the user, and communication will be restored.

Client Requirements:

- The device will substitute for high frequency hearing loss to the extent of helping the user in everyday communication.
- The device will use vibro-tactile stimulation.
- The device should be self contained, portable, and discrete.
- Complete a testable prototype for analysis of frequency range and wear-ability.
- Test the prototype, obtain results, and determine efficacy.

Design Requirements:

1. Physical and operational characteristics

a. Performance requirements

- It will increase the user’s quality of communication by allowing the user to recognize high frequency consonants and incorporate them into word recognition through vibro-tactile stimulation.
- This device should use real time analog filtering to recognize and separate certain high frequency sounds and communicate them to the vibro-tactile stimulator.

b. Safety

- A current of more than 5 mA should not pass through the device and into the user.
- The device should not heat to over 43° C (110° F) while in use.
- Testing protocol should comply with all IRB regulations.

c. Accuracy and Reliability

- Be able to detect sound at normal speaking level of 60 dB while rejecting environmental noise.
- Be able to process and substitute for the consonants T, F, S, Th, Sh, and P when coming from a variety of different vocal tones enough to improve scores on standard speech recognition evaluation tests, Word Intelligibility by Picture Identification (WIPI), sentence level Bamford-Kewal-Bench test (BKB), and PLOTT test for vowel and consonant discrimination.
- Human Hearing Frequency Range: 20 – 20,000 Hz
- Speech Frequency Range: 125 – 8,000 Hz
- High Frequency Hearing Loss: above 1,000 Hz

d. Life in Service

- The transducers should last at least 3 years.
- The sound processing unit, along with its microphone, should last 10 years.
- Adhesive should last at least through an entire day.
- On a single battery charge the device should last approximately 5 days, similar to that of a common hearing aid.
- Common hearing aid batteries have an output voltage of 1.4 V and have power ratings between 140 and 640 mAh. With daily use of the device being about 14 hours the device should draw from 2 - 10 mA of current from the battery.

e. Operating Environment

- The device will be located behind the ear.
- Elements such as wind, rain, sun and sweat should not cause the device to vibrate for non-spoken noises, output dangerous levels of current or distort outgoing signals.
- Eventually test the prototype on human subjects in a lab setting.

f. Ergonomics

- The device should not move during normal physical activity.
- Transducer unit should be hidden by the ear.
- Device should be comfortable to the user.
- The controls of the processing unit should be small enough to keep them discrete yet still be able to be manipulated.
- Attach vibro-tactile device with adhesive that will not cause irritation.
- The location of the transducer should be easily repeatable.

g. Size

- The transducer unit should be no more than 5 cm in length, 1.75 cm wide, and 1.25 cm thick.
- The processing unit should be no more than 10 cm in length, 5 cm wide, and 2 cm thick.

h. Weight

- The weight of the processing unit should be no more than 8 oz.

i. Materials

- Adhesive which holds the transducer unit in place should not irritate skin, leave large amounts of residue, or be painful to remove.
- Soft, durable plastic such as vinyl.
- Housing unit made of material to absorb vibrations.

j. Aesthetics, Appearance, and Finish

- Unit should be flesh-colored and not overtly noticeable to others.
- Adhesive attachment used for transducer unit should not leave large amounts of residue and should not be painful to remove.

2. Production Characteristics

a. *Quantity*: The device should be able to be produced in mass quantities.

b. *Target Product Cost*: The device should cost between $300 and $500, which is approximately 5-10% of the total cost of a hearing aid.

3. Miscellaneous

a. Standards and Specifications: FDA approval of a class II device. Must follow regulation code 21 CFR 812 and 21 CFR 50 for testing.

b. Customer: Adjustable frequency ranges depending on what the user needs most

c. Patient-related concerns: Device should not cause discomfort and should not be overly noticeable.

d. Competition:

- Tickle Talker
- Tacticon 1600
- Tactaid 7 – http://www.tactaid.com/tactaid71.html