CHAPTER 12
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VOICE PITCH ANALYZER

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INTRODUCTION

A portable device for monitoring voice pitch was designed for patients with speech impairment. The portable device assists therapists and their patients in acquiring information on episodes of exceedingly high pitch in everyday conversations.

SUMMARY OF IMPACT

Speaking with unusually high volume and/or high pitch is believed to cause pathological conditions to the vocal cords, such as vocal nodules. Chronic hoarseness, breathiness, or complete loss of voice are the most common symptoms of vocal nodules. Abnormal voice qualities such as these can be devastating to one's job performance and can have profound psychological effects.

The primary method of treating patients with voice disorders related to vocal abuse is the monitoring of vocal habits. Typically, patients are asked to self-monitor their speaking habits throughout the day and make adjustments in their phonatory behaviors as indicated. This treatment strategy is often unsuccessful (although some additional techniques, not described, here) have proven successful in many cases.

A continuous monitoring device may be more effective in objectively determining the individual's natural voice pattern and providing feedback (through alerting sounds) when excessive high pitch or high intensity is detected. Whenever the normal pitch range is exceeded, the portable unit produces an audible tone alerting the patient. In addition, a total daily count is registered to further assist the clinician in monitoring the patient's progress towards recovery. The portable speech analyzer meets the needs of many clients. The unit will ideally lead to fewer office visits and in turn to a reduction in healthcare costs for the patient.

TECHNICAL DISCRIPTION

The portable device is battery operated and can be worn around the waist. Signal from either a throat microphone or lapel microphone is sampled and amplified with an adjustable gain through the onboard A/D converter. Upon acquiring the sampled speech signal, digital signal processing algorithms coded in TMS320C50 assembly language perform voice pitch level calculations. The DSK board will generate a beeping sound if the voice pitch exceeds a preset threshold recommended by the clinician. The pattern of the audible sound is programmable based on the preference of the patient.

The main design requirements for this project were that it be: 1) portable and as small as possible for the patient to wear around the waist without any discomfort; and 2) microprocessor based for ease of entering the patient-dependent parameters and for maintaining a log of episodes.

The design is carried out on the Texas Instruments TMS320C50 digital signal processor (DSP) DSK board. The fixed point TMS320C50 DSP provides 10 K of on chip RAM with an instruction cycle of 50 nsec. The DSK board comes with the Texas Instruments TLC32040C Analog Interface Circuit (AIC). The AIC is a highly integrated component that combines the functions of a 14-bit A/D, a 14-bit D/A, input anti-aliasing filter, output reconstruction filter, and a serial CPU interface. The AIC can be programmed for various sampling rates, anti-aliasing frequencies, and input gains.

For the portable speech analyzer, the input speech signal from the microphone pre-amplifier is sampled at 8K Hz by the AIC. The algorithm for the portable speech analyzer uses two buffers of size 240 points (one frame). While one buffer is being filled, the other is processed. Four buffer status flags are used to control processing. The RBUFSEL flag is set to controls which of the two input buffers are currently being filled. PBUFSEL controls which buffer is being processed. BUF1RDY and BUF2RDY indicate that the corresponding buffer is full and ready to be processed. The main processing loop continuously checks for a full buffer. Upon finding a full buffer, the subroutine for the power calculation is called. During the processing, the other buffer is filled by the interrupt service routine (ISR). In the power subroutine, the power associated with the 240 samples in the buffer is calculated. The calculated power value is then added to a running sum that calculates the averaged power of 10 consecutive frames. The averaged power is then compared with a set threshold. If the averaged power exceeds the threshold, then a warning tone is generated. If desired, the output warning could also be realized in the form of a mechanical vibration. Data in the buffer are analyzed for the largest sample in the first and last third of the buffer to determine a clipping value. Two-thirds of the smaller of the two peak values is used to clip the data prior to pitch estimation routine as to remove components due to formant frequencies.

The Autocorrelation Method and Average Magnitude Difference Function (AMDF) method are two methods used for calculation of vocal pitch. Auto-correlation is a technique used to emphasize the periodic peaks and de-emphasize the non-periodic portions of a signal by taking the windowed sum of lagged products of the signal. Auto-correlation shows a maximum at a first non-zero lag equal to the phonatory pitch. The AMDF method takes the absolute value of the windowed difference between lagged signal samples. The non-zero lag with the deepest value of ΔMDF indicates the pitch period. The pitch information is averaged over several frames. If the averaged pitch value exceeds a preset pitch threshold a warning sound or a mechanical vibration is produced.

The size of the portable unit is 4 5/16 by 2 11/16 by 2 1/16 inches. The housing contains the DSK board, the microphone preamplifier, the beeper unit, and the batteries. The portable unit consumes three 9-volt alkaline batteries. The DSK board itself runs from two 9-volt batteries (+9 and –9 volts) and the microphone preamplifier also requires a separate 9-volt supply. The current consumption during the loading of code from PC to the DSK is 240 mA with a minimum required voltage of 6.00 volts. During processing, the current consumption drops to 100 mA with a minimum required voltage of 4 volts. In order to save on battery life, in the absence of any input voice signal, the processor is put into the idle mode (via software) where the current consumption can be further dropped to 50 mA. The overall continuous running time for the portable unit is approximately 3 hours.

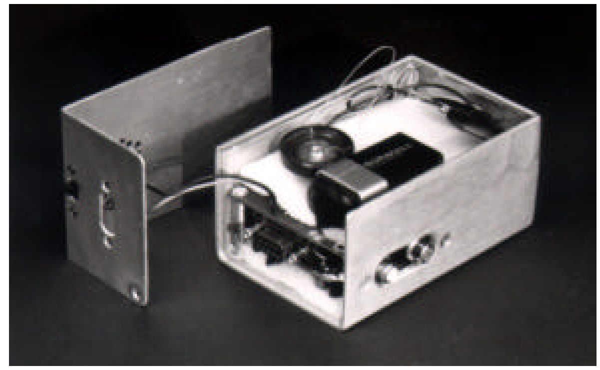

Figure 12.1. Portable Speech Analyzer Internal View.

A three-digit display can also be incorporated to the design to display the number of times excessive pitch or high intensity occurred. A programmer BCD counter with asynchronous RESET (74HC160) is used. The BCD output is directly fed to the BCD-to-Seven Segment latch/decoder/display driver (74HC4511). A common cathode 7 segment LED display is directly connected to 74HC4511. This arrangement is identical for all the three digits. However, the ripple-carryout output of the counter driving the least significant digit must be given as the clock input to the next counter. The RESET and LOAD pins of the counters are tied to the supply.

The device was tested for detecting various levels of high pitch or voice signals, and in each case the device was successful.

The final cost of the project is approximately $200.

A DSP-BASED WIRELESS INFANT MONITORING DEVICE FOR INDIVIDUALS WITH HEARING IMPAIRMENT

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INTRODUCTION

A portable wireless device was designed to enable an individual with hearing impairment to monitor an infant by detecting when the infant is crying. The receiver of the device can be housed like a pager and, in effect, page the individual with hearing impairment each time the infant cries.

SUMMARY OF IMPACT

Persons with hearing impairment are often unable to use intercom-type infant monitoring devices. Therefore, alternative devices are essential. This infant-monitoring device relies on a digital signal processor to detect infant crying sounds. Once crying is detected, a paging device alerts the user by vibrating and activating a light emitting diode. With such a device, users may conveniently monitor an infant and still complete other household activities. The wireless DSP-based infant-cry-recognition system serves as a cost effective and convenient device for enabling an individual with hearing impairment to monitor an infant.

TECHNICAL DISCRIPTION

Digital signal processing and wireless transmission are the two primary technologies employed in the development of this device. Digital signal processing is used to recognize the sound of an infant crying. Sound recognition is accomplished in real time through the use of a digital signal processor (DSP). The device utilizes the TMS320C50 DSK board, a low-cost DSP board equipped with 14-bit input/output analog- to- digital (ADC) and digital-to-analog (DAC) converters. The DSP receives a signal from a microphone located near the infant. Employing digital signal processing algorithms, the DSP determines if the microphone signal has the same properties as those of an infant crying. The DSP then sends a control signal to a wireless transmitter. The transmitter sends a digital code to a receiver using frequency shift keying modulation technique. Upon detecting the code, the receiver vibrates and activates a light emitting diode.

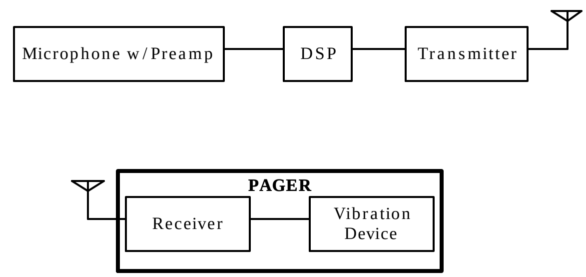

Figure 12.2. Block Diagram of the Wireless Infant Monitoring Device.

The design requirements for the wireless monitoring device were that it: 1) be able to recognize infant crying sounds; and 2) consistently alert the caretaker of infant crying via a wireless system.

The infant crying sound detection is carried out on the Texas Instruments TMS320C50 digital signal processor (DSP) DSK board. The input speech signal from the microphone pre-amplifier is sampled at 8K Hz by the AIC. One can assume that the environment in which the infant-monitoring device operates will be relatively quiet. Sounds such as doors opening and closing and people moving past the device will probably be the extent of the common noises in the environment.

Speech recognition relies on the speaker to vocalize clearly and at a reasonable volume. Since infants do not make precise crying sounds at distinct volumes, the methods used to identify the crying sound are based on short-term energy and zero-crossing analysis. Short-term energy is the energy contained in a finite length of the signal. It is defined as the sum of the squares of the samples in the 20 msec segment sliding on a sample-by-sample basis. Once the short-term energy exceeds a preset threshold, the algorithm indicates that an infant crying sound may be present. The threshold is chosen so that very quiet or distant sounds will not cross the threshold.

To make use of short-term energy practical, the microphone preamplifier was designed with adjustable gain. This gives the user some degree of freedom as to how far the microphone can be placed from the infant simply by increasing the gain with distance. The second part of recognition is based on zero-crossings. Zero-crossings are the number of times the signal crosses the zero axis. There was an obvious trend in zero-crossing any time the infant started to cry. As a result, it was determined that using zero-crossing in conjunction with short-term energy could provide a suitable method for determining if the sound was the infant crying. The only concern was that other common sounds would also share these characteristics. It was determined through real time testing that most sounds such as talking in a normal tone and doors closing were not recognized as the infant crying.

The firmware for this project consisted of a Texas Instruments TMS320C50 Digital Signal Processor mounted on small evaluation board called a DSK board. The DSK consists of the DSP, a power supply, an analog interface, two RCA sockets for analog input and output, and an RS232 connection to communicate with a personal computer. The entire evaluation kit also included a debugger, an assembler, an instruction manual, and sample programs that were used to develop the DSP program. A Texas Instruments TMS320C50 digital signal processor was used. It is operated at approximately 40 MHz and is only capable of fixed-point operation. The arithmetic logic unit (ALU), the accumulator, and its buffer are 32 bits. The TMS320C50 has 2K x 16-bit on-chip ROM, 9K x 16 bit on-chip RAM, as well as 1056 x 16 bit on-chip data RAM. It also contains 64K I/O ports and two serial ports for input and output.

The DSP was programmed to fill a buffer of samples with a length of 20ms. Since the analog to digital converter was programmed to sample at 8000 Hertz, the buffer contained 160 samples when filled. For each subsequent input sample, the oldest sample in the buffer is overwritten by the new input. Each time the buffer is updated with a new sample, its short-term energy is calculated. The energy is then compared to the threshold number stored in memory. The threshold level of the signal was determined in the simulation. If the energy exceeds the threshold, the DSP calculates the zero-crossings in the buffer. The number of zero-crossings found in the buffer is then compared to two values specifying the valid range of infant crying zero crossing. These values were also found by analyzing the simulation results. If the number of zero-crossings falls between these two numbers, an output signal is sent from the DSP to the transmitter. The transmitter sends this signal, using frequency shift keying modulation technique, to the receiver, which activates the pager. The final cost of the project is approximately $300.

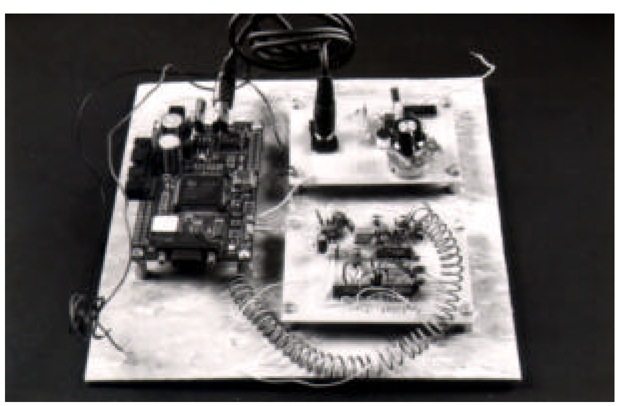

Figure 12.3. DSP-Based Wireless Infant Monitoring Device for Hearing-Impaired.