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November 9, 2007

Dr. Andrew Rawicz
School of Engineering Science
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Re: Design Specification for an Anti-Snoring Pillow (ASP)

Dear Dr. Rawicz:

Attached is the design specification for LifeX Technology's most recent innovation, the *Anti-Snoring Pillow*. LifeX's goal is to design an *Anti-Snoring Pillow* that uses active noise cancellation technology to lower the background noise, so that everyone can enjoy peaceful and quality sleep every night, even with a noisy snorer.

The design specification is a set of detailed specifications regarding Anti-Snoring Pillow's overall implementation. The design specification will be used as a guide by LifeX for the rest of the development phase of the product.

LifeX Technology is comprised of four highly motivated undergraduate engineering science students from SFU: Camillia Lee, Stanley Yang, Simon Wong and Raymond Lee. If you have any questions or comments, please do not hesitate to contact us by phone at (604) 594-6816, or email us at *KHL2@sfu.ca*.

Sincerely,

Raymond Lee

Chief Executive Officer
LifeX Technology

Enclosed: Design Specification for an Anti-Snoring Pillow (ASP)

DESIGN SPECIFICATION:

ANTI-SNORING PILLOW (ASP)

PROJECT MEMBERS:

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ISSUED DATE:

NOVEMBER 8, 2007

VERSION

3.0

Executive Summary

The design specification is a set of detailed specification regarding Anti-Snoring Pillow's (ASP) overall design and it will be followed closely as a guide by LifeX's project team for the rest of the development phase of the ASP. This design specification is specifically tailored to be used for our proof-of-concept model. As a result, we will only be discussing I and II functional requirements of the *Functional Specification for an Anti-Snoring Pillow (ASP).*

The design specification provides a set of detailed implementation of the ASP, as well as justifications for our design choices. This document includes both hardware and software design of the ASP. For hardware, it specifies the overall system design, system setup, as well as details regarding all the major components of the product, which includes digital signal processor (DSP), speaker, microphone, and pillow. For software, this document provides a list of possible algorithms that are suitable for use with active noise cancellation technology. The most suitable system setup and algorithm will be chosen and implemented during the design & development phase of the project.

The design specification also includes possible candidates as well as justification and reasoning for the final selection of a suitable DSP. A detailed module and system test plan is also provided at the end of this document.

We are currently in our late stages of the design & development phase and will soon be moving to our last phase of the project, which is the integration, testing and optimization phase. The four month R&D cycle of the ASP prototype is expected to be completed by the December 15, 2007, upon which the prototype is expected to conform to this proposed design specification.

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Acronyms

ANC:	Active Noise Cancellation/Control
DSK:	Development Software Kit
DSP:	Digital Signal Processor
EMIF:	Memory interfaces used by DSP
CPLD:	Complex Programmable Logic Device
SDRAM:	Synchronous Dynamic Random Access Memory
ADC:	Analog to Digital Conversion
DAC:	Digital to Analog Conversion
DIP:	Dual In-line Package
SPI:	Serial Peripheral Interface
FIR:	Finite Impulse Response
IIR:	Infinite Impulse Response
LMS:	Least Mean Square
FxLMS:	Filtered-X Least Mean Square
RLMS:	Recursive Least Mean Square
DRAM:	Dynamic Random Access Memory
LED:	Light-Emitting Diode

Glossary

AIC23: An audio codec acquired by Texas Instrument for A/D conversion during audio communication

Frequency Response: A measure of what frequencies can be produced and how accurately they are produced.

Flash Memory: A type of memory chip that can retain data after that system has been turned off.

Control Source: The device responsible for outputting the shifted and inverted signal for noise cancellation

Error Sensor: The sensor responsible for picking up remaining un-cancelled “noise” at the end of system

Feedback Control: The process in which part of the output of a system is returned to its input in order to regulate its further output.

Omni-Directional: Having equal sensitivity to sounds or signals coming from any direction.

1. Introduction

The Anti-Snoring Pillow is an electronic system that uses Active Noise Cancellation technology to reduce the noise level of snoring at the non-snorer’s pillow. Using microphones, speakers and a DSP controller, an adaptive noise prediction system will produce an anti-noise which will suppress the unwanted noise. Our aim is to minimize the noise of snoring at low frequencies (0 to 500 Hz) at the non-snorer as much as possible. The design specification describes the detail design of each component of the Anti-Snoring Pillow.

1.1 Scope

The design specification describes in detail of the design of the Anti-Snoring pillow and explains how the design satisfies the functional requirements described in *Functional Specification for the Anti-Snoring Pillow.* This document will focus on the requirements for the proof-of-concept system only. The appendices include algorithms and flow charts to help facilitate the implementation of the Anti-Snoring pillow.

1.2 Intended Audience

The functional specification will be used by all members of the Anti-Snoring Pillow team in LifeX Technology. This document will help design engineers with the detailed design during the design and implementation phase. Test engineers will use this document to design test plans and to make sure that the final product behaves correctly.

2. System Specification

The Anti-Snoring Pillow system will generate an anti-noise to suppress the unwanted snoring noise using an active noise cancellation system. It will perform the noise suppression automatically upon powering up through the push of the start button. The physical arrangement of the system components must be fixed to ensure proper performance.

3 Overall System Design

3.1 Cushioning

Cushioning of the pillow will also be a very important aspect of our design. The anti-snoring pillow will be filled with cotton fibers.

Using the cotton fibers we will create a hypoallergenic pillow that can be used by anyone. Also, cotton is proven to be incredibly durable and will last several years without to loosing its form.

Other forms of filling where considered such as wool, down, memory foam and polyester. Down fill is very durable and firm but can be uncomfortable due to its firmness. With a down fill it could be to stiff and not contour to the structure of the sleeper’s head. In addition, the hypoallergenic down pillows are significantly more expensive to buy. A cotton fill would be much softer and more comfortable to use.

The cotton fill would be evenly disturbed throughout the pillow for the perfect balance between softness and firmness. The firmness of the pillow will help maintain alignment and the position of the head. The durability of the cotton fill will provide several years of support. The cushioning will be covered with a pillow case that can be easily washed for increased longevity of the pillow.

Cotton fill is affordable and comfortable thus providing overall performance for a good nights rest. In the future, if the cost to produce our product decreases an alternative material may be used.

3.2 High level system design

Outlined in Figure 1, is the over is the simplistic system arrangement and their relations to each other.

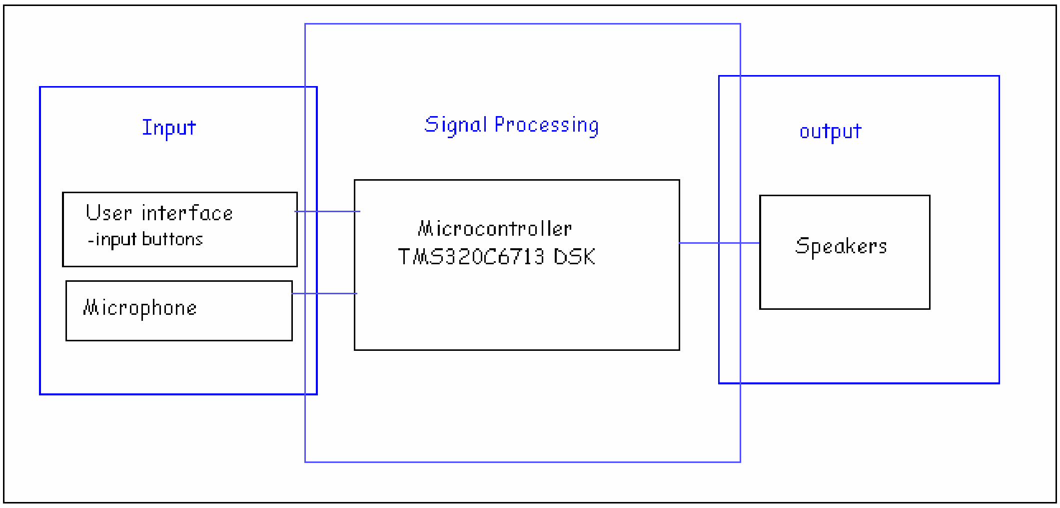

Figure 1 System Overview

For the system to turn on the on button needs to be pressed on the user interface. This button will activate the microcontroller. Once the microprocessor is activated it receives the recorded signals from the microphone input. The signals are then processed in the microcontroller. The microprocessor used is the TMS320C6713 DSK which is controlled by the provided software. This software is programmed to manipulate the input signal to produce an output signal at the speakers. The output signal produced is adjusted and modified at a specific amplification for ideal noise cancellation.

3.3 Microphone and Speaker Arrangement

The arrangement of the microphones and the speakers in order to achieve the optimal noise cancellation involves using finite element analysis and genetic algorithm. The positioning of the microphone and speaker will be described in detail in System Setup.

3.4 Electrical System

The Anti-Snoring Pillow basically has four components that require power to operate: DSP controller, error microphone, reference microphone and output loudspeakers. The DSP controller will be running on a 5V supply voltage and 3A supply current. The microphones will be powered by a 1.5V-5V power source directly from the DSP controller. The loudspeakers will be powered directly off a regular electrical outlet of 110-120V.

There will be cable connection going from the speakers and microphones to the DSP controller. Cables shall be insulated and sheathed and installed in a heavy duty non-metallic enclosure to protect the users from any electrical danger. Other safety issues regarding electrical power supplies are not concerned in this project as protection circuits are already integrated within each components. Figure 2 shows the overview diagram of the electrical connection for the Anti-Snoring Pillow.

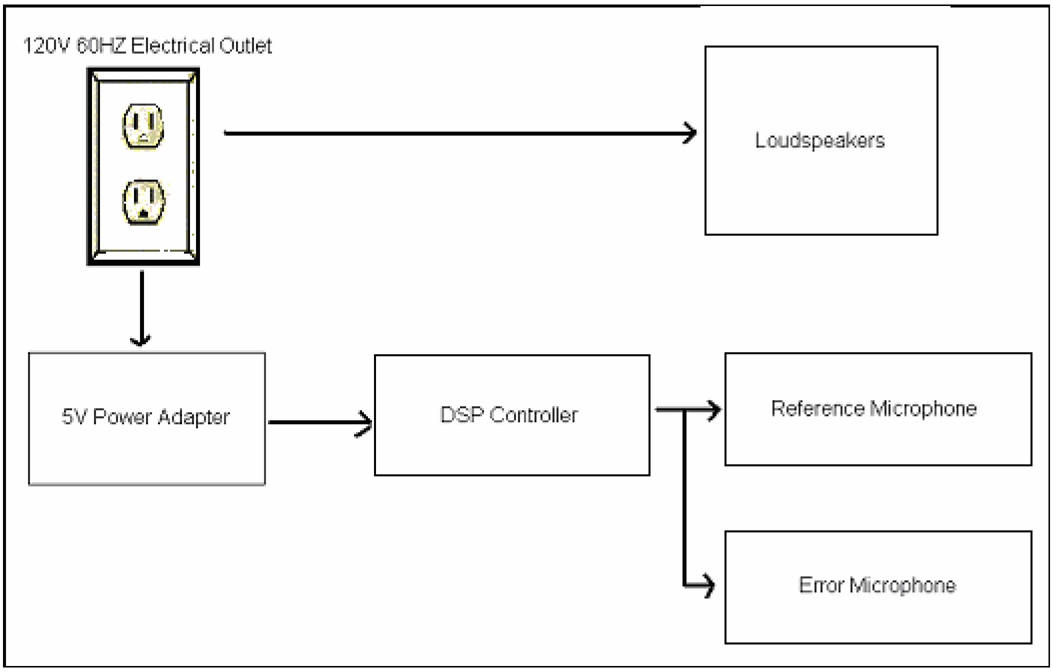

Figure 2: Electrical Connection of the System

4. Microphone

The microphone we are going to use for the ASP will be Frontech's JIL-3452 microphone. This microphone is ideal for use with the Anti-Snoring Pillow mainly because its frequency range is low enough for sensing low frequency sounds and it is omni-directional. Note for the prototype, we are currently not taking the size into consideration. Figure 3 show's an image of Frontech's JIL-3452.

Figure 3: Frontech JIL-3452 [8]

The microphone's specifications are listed in Table 4.1 [8].

Specifications	
Impedance	Low impedance
Frequency Range	50~16000Hz
Sensitivity	>= -58dB =+- 3dB
Directivity	Omnidirectional
S/N	>=60dB @1KHz
Operation Voltage	1.5~9V
Plug / Cable	3.5mm stereo / 1.5m (dia)

5. Speakers

The speakers we are using to implement the ASP will be Logitech's X-540. Logitech's X-540 meets the functional specifications for our ASP prototype. The main reason for choosing these speakers is because the frequency response of the speakers are low enough to play back the low frequency anti-noise of snoring sounds. Note for the prototype, we are currently not taking the size into consideration. Figure 4 show's an image of Logitech X-540.

Figure 4: Logitech X-540 [9]

The main specifications are listed in Table 5.1 [9].

Specifications	
Total RMS Power	70W RMS
● Satellite	45W
● Subwoofer	25W
Total Peak Power	140W
Frequency Response	40Hz ~ 20kHz

6. DSP

The digital signal processor we are going to use to implement the ASP is Texas Instrument's TMS320C6713 DSK.

6.1 Hardware Components

6.1.1 Functional Overview

The DSP on the 6713 DSK interfaces to on-board peripherals through 32-bit wide External Memory InterFace (EMIF). The SDRAM, Flash, CPLD are all connected to the bus. EMIF signals are also connected to daughter card expansion connectors which are used for third party add-in boards.

The DSP interfaces to analog audio signals through an on-board AIC23 codec and four 3.5mm audio jacks (microphone input, line input, line output, and headphone output). The codec can select the microphone or the line input as the active input. The analog output is driven to both the line out and headphone connectors.

The CPLD is used to implement glue logic that ties the board components together. The CPLD has a register based user interface that lets the user configure the board by reading and writing to the registers.

The DSK includes four LEDs and a four position DIP switch to provide the user with interactive feedback. These can be accessed by manipulating the registers on the CPLD.

The DSK operates from a single 5V external power supply connected to the main power input. On-board switching voltage regulators provide the +1.26V DSP core voltage and +3.3V for DSP's I/O buffers and all other on-board chips. The power connector is a 2.5mm barrel-type plug.

The DSP communicates with the USB host interface through an embedded JTAG emulator. The DSK can also be used with an external emulator through the external JTAG connector.

6.1.2 Main Board Components

The main components of the DSK include DSP, CPLD, flash, SDRAM, and AIC23 codec. Figure 5 is a block diagram of the C6713 DSK.

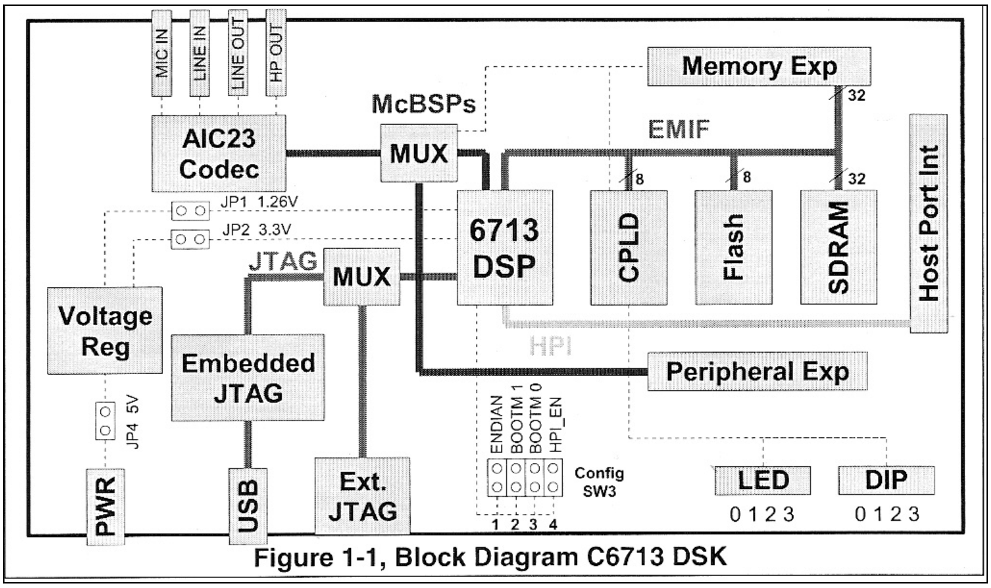

Figure 5: Block Diagram 6713 DSK [4]

6.1.3 Digital Signal Processor Core

The TMS320C6713 DSP is a floating-point DSP generation in the TMS320C6000 DSP platform. The C6713 device is based on the high-performance, advanced very-long-instruction-word (VLIW) architecture developed by Texas Instruments (TI), making this DSP an excellent choice for multichannel and multifunction applications.

Operating at 225 MHz, the C6713B delivers up to 1350 million floating-point operations per second (MFLOPS), 1800 million instructions per second (MIPS), and with dual fixed-/floating-point multipliers up to 450 million multiply-accumulate operations per second (MMACS).

6.1.4 Complex Programmable Logic Device

The C6713 DSK uses an Altera EPM3128TC100-10 Complex Programmable Logic Device (CPLD). The EPM3128TC100-10 is a 3.3V, 100-pin QFP device that provides 128 macrocells, 80 I/O pins, and 10ns pin-to-pin delay. The device is EEPROM-based and is in-system programmable via a dedicated JTAG interface.

The CPLD logic is used to implement the ANC-enabled Anti-Snoring Pillow (ASP). The four CPLD memory-mapped registers allows users to control CPLD functions in software. On the 6713 DSK the registers are primarily used to access the LEDs and

DIP switches and control the daughter card interface.

USER_REG Register

USER_REG register is used to read the state of the four DIP switches and turn the four LEDs on or off to allow the user interact with the DSK. The DIP switches are read by reading the top four bits of the register and the LEDs are set by writing to the low four bits.

DC_REG Register

DC_REG register is used to monitor and control the daughter card interface.

VERSION Register

Version register contains two read only fields that indicate the BOARD and CPLD versions. This register will allow your software to differentiate between production releases of the DSK and account for any variances. This register will not be expected to change often, if at all.

MISC Register

The MISC register is used to provide software control for miscellaneous board functions. On the 6713 DSK, the MISC register control how auxiliary signals are brought out to the daughter-card connectors.

6.1.5 Flash Memory

The DSK uses a 512Kbyte external Flash as a boot option. The Flash is wired as a 256K by 16bit device to support the 16-bit boot option. However, the software that ships with the DSK treats the Flash as an 8-bit device, as a result only 256Kbytes are available without software changes.

6.1.6 Synchronous DRAM

The DSK uses a 128megabit synchronous DRAM (SDRAM) on the 32-bit EMIF. The total available memory is 16 megabytes. The integrated SDRAM controller is part of the EMIF and must be configured in software for proper operation. The EMIF clock is derived from the PLL setting and should be configured in software at 90MHz.

6.1.7 AIC23

The DSK uses a Texas Instrument AIC23 stereo codec for input and output of audio signals. The codec samples analog signals on the microphone or line inputs and converts them into digital signals so it can be processed by the DSP. When the DSP is finished with the data it uses the codec to convert the samples back into analog signals on the line and headphone outputs so the user can hear the output.

The codec communicates using two serial channels, one to control the codec's internal configuration register and one to send and receive digital audio samples. The control channel is unidirectional and is used to send a 16-bit control word to the AIC23 in SPI format. This channel is only used when configuring the codec, and is usually left idle when audio data is being transmitted. The data channel is bi-directional and it supports many data formats based on the three variables of sample width, clock signal source, and serial data format.

The codec has a 12MHz system clock which corresponds to USB sample rate mode, because many USB systems use a 12MHz system clock. The internal sample rate generated subdivides the 12MHz clock to generate common frequencies such as 48 KHz, 44.1 KHz, and 8 KHz.

6.2 Audio Codec

The TMS320C6713 evaluation kit uses the AIC23 stereo codec for input and output audio signals. Analog signals are sampled through either the line in or microphone input of the DSK, then through the AIC23, gets converted to a digital signal so it could be converted for digital processing. When the signal is processed, the data is converted back into analog signals by the codec, in order to be played at the headphone or lineout audio jacks.

AIC23 uses two serial channels, one to control the codec's internal register, and one to transfer data back and forth. A port called McBSP1 is used for unidirectional data control. It is always programmed to send a 16-bit control word to the AIC23 in SPI format. The 7 most significant bits are used to specify which register, while the other 9 specifies the register content. McBSP1 is mostly idle, since it is only used during the configuration of the codec.

Audio data are transferred via a bi-directional port called McBSP2. Data transferred must have the proper sample width, clock signal source and serial data format in order to be received by AIC23. In general, MCBSP2 carries a 16-bit sample width in master mode so it generates the frame sync and bit clocks at the correct sample rate without the need of the DSP. Usually the DSP are designed to operate with McBSP ports.

AIC23 uses a 12MHz system clock. The sample rate is set by the codec's SAMPLERATE register.

The maximum level of input signal to be converted is determined by the specific ADC circuitry on the codec, which is 6V pk-pk with the onboard codec. The captured signal is then processed. An output filter smooths out or reconstructs the output signal. ADC and DAC are all apart of the AIC23 chip.

Functions involved in the AIC23 include lowpass filtering, oversampling, ADC, DAC and so on. Only words with length of 16, 20, 224 and 32 bits could be properly transferred by McBSP2.

The AIC23 codec contains a 12 MHz crystal oscillator to be used as a clock. The sampling rate is adjustable and can be configured to either 32, 44.1 48, 88.2 or 96kHz.

The ADC converts the input signal to discrete digital words in the format of 2's-complements, which represents an analog signal level. The DAC is representing an interpolation filter and a digital modulator. A decimation filter is needed to reduce the digital data rate to match the sampling rate.

6.3 ANC Algorithms and Implementations

The Active Noise Cancellation algorithms and implementation is the heart of the Anti-Snoring Pillow. The ANC system basically produces an "anti-noise" source. The anti-noise is then introduced back into the system usually through the means of loudspeaker to achieve noise reduction. In this section, the design of the electronic control system will be described in detail here. First the type of ANC system will be discussed, then control digital filter types followed by the algorithms used for tuning them.

6.3.1 Types of ANC System

There are two major types of active noise control system: Adaptive filtering and waveform synthesis. Waveform synthesis is a type of feed forward control that is only suited to periodic noise. Snoring noises are generally non-periodic in nature and so adaptive filtering is the obvious choice here.

Adaptive filtering system can be classified into feed forward and feedback system. They differ from each other in the manner in which the control signal is attained. Feed forward systems use some predictive measure of the unwanted disturbance to generate the anti-noise. On the other hand, feedback systems try to attenuate the unwanted disturbance after it has passed. Compared to a feed forward system, feedback systems basically do not have a reference signal of the primary noise source.

In scenarios where it is not possible to sample the primary noise sufficiently early for a feedforward system, feedback systems are the ideal type of controller. However, the performance of a feedback system lacks the feedforward system and a feedback system is inherently unstable. When a reference signal is available, like in the case of the Anti-Snoring Pillow system, a feedforward system is always preferred over a feedback system.

Feed forward systems usually consist of a reference sensor, a control source, an error sensor, and an electronic controller as shown in Figure 7.

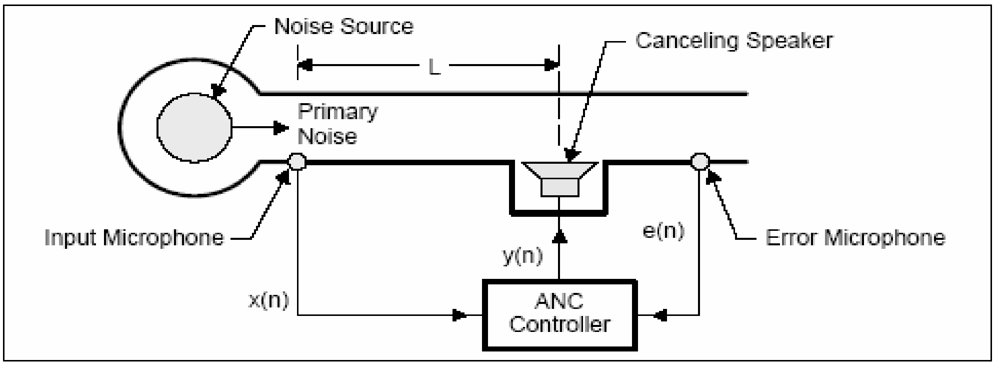

Figure 6: Example of a feed forward system. [2]

Systems for feedforward ANC can be further classified into two categories:

• Adaptive broadband feedforward control with an acoustic input sensor
• Adaptive narrowband feedforward control with a nonacoustic input sensor

The second one is ideal for applications where the primary noise is periodic and is produced by rotating machines. However, as stated before, snoring is not periodic and not produced by rotating machines. Therefore, an adaptive broadband feedforward control with an acoustic input sensor will be the type of ANC system used for the Anti-Snoring Pillow.

6.3.2 Digital Filter

The control filter is a part of the ANC controller shown in Figure 6. It is responsible for generating a control signal output which is then fed to a loudspeaker to generate a canceling sound. A digital control filter uses samples of reference inputs, multiplies them by filter coefficients, and sums the results to produce the control output.

The most common type of control filters are the finite impulse response filter (FIR) shown in Figure 7 and the infinite impulse response filter (IIR) shown in Figure 8.

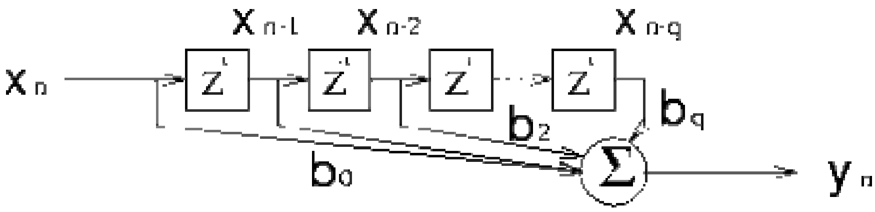

Figure 7: Example of a FIR filter. [3]

FIR filter produces an output that is the weighted sum of the current and past inputs.

$$\begin{aligned} y_n &= b_0 x_n + b_1 x_{n-1} + b_2 x_{n-2} + \ldots + b_q x_{n-q} \\ &= \sum_{j=0}^{q} b_j x_{n-j} \end{aligned}$$

[3]

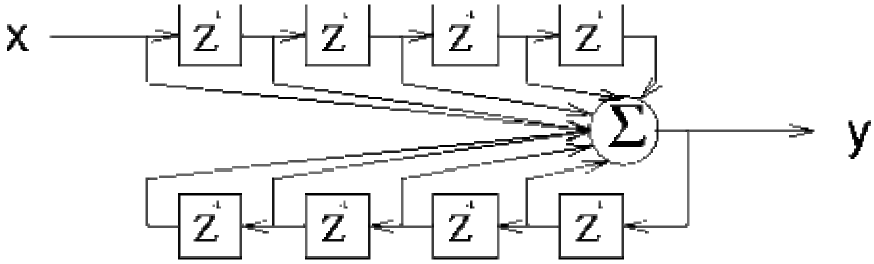

Figure 8: Example of an IIR filter. [3]

IIR filter produces an output that is the weighted sum of the current and past inputs and past outputs.

$$y_n = \sum_{i=1}^{p} a_i y_{n-i} + \sum_{j=0}^{q} b_j x_{n-j}$$

[3]

FIR filters are ideal solutions to tonal noise problems and where the control signal does not interfere with the reference signals. In the case of the Anti-Snoring Pillow, the noise is not tonal and the loudspeaker signal will indeed contaminate the signal at the reference microphone.

For scenarios where the noise is broadband in nature, IIR filters are preferred over FIR filters. In addition, IIR are preferred in feedforward systems where the control source can pollute the reference microphone through acoustic feedback. Although the problem can be reduced through modifying physical characteristic of the system, it can never be entirely eliminated as long as the reference signal is captured through a microphone.

Another major advantage of IIR filter is that it can accurately model complex systems with much fewer filter coefficients compared to FIR. The result is that IIR requires much less computational load. However, the tradeoff of this advantage is its inherent instability, slower convergence and the possibility of convergence to a local minimum error instead of a global minimum. Despite the disadvantages, the more appropriate choice of digital filter for the Anti-Snoring Pillow would still be the IIR filter.

The three major parameters that affect the performance of the digital control filters are the type of filter, the filter weight values and the number of weights. For random broadband noise, up to several hundred filter weights are needed [7]. The exact number of weights will be determined by trial and error for optimum performance during design.

The digital control filter weight values are tuned by adaptation algorithms and will be discussed in the next section.

6.3.3 Adaptation Algorithms

A part of the ANC controller shown in Figure 10 is the adaptation algorithms, which is responsible for updating the digital filter's weight coefficients. A number of algorithms presented in [2] will be discussed here and they are based on one of the most common algorithm, the least mean-squared (LMS) algorithm. A few algorithms are to show basic concepts but only the one used for IIR digital filter implementation procedure will be discussed in detail.

Figure 10 shows a simplified high level block diagram of the ANC system, where P(z) is the unknown primary acoustic path, W(z) is the adaptive filter, H(z) is the secondary acoustic path and LMS is the adaptation algorithm. The goal of all adaptation algorithms for ANC system is to minimize the error signal, *e(n),* using the anti-noise produced by W(z).

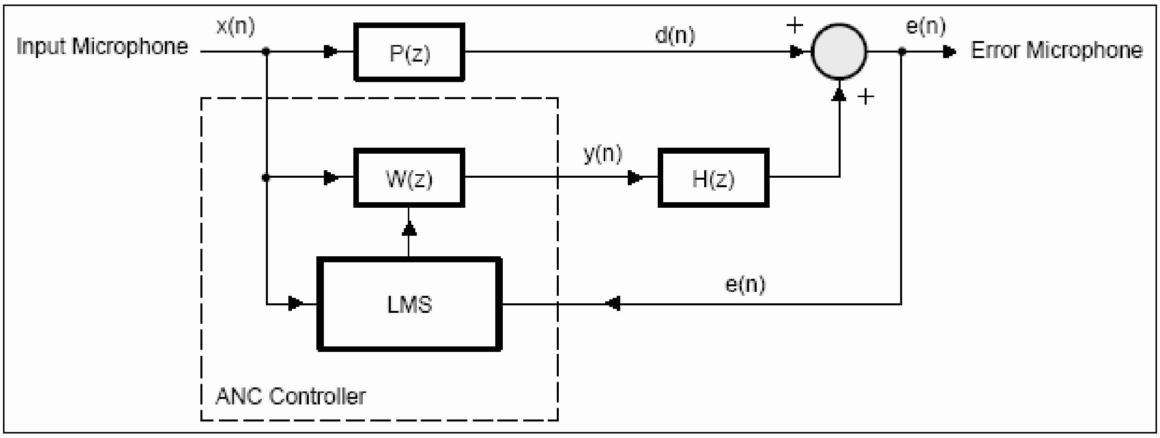

Figure 9: Block Diagram of ANC System. [2]

The z-transform of the error signal in Figure 10 is:

$$\mathbf{E(z) = X(z)\ P(z) + X(z)\ W(z)\ H(z)}$$

After the convergence of the adaptive filter, ideally the e(n) should be zero. This implies that:

$$W(z) = -P(z) / H(z)$$

Therefore, the adaptive filter has to model the primary path and the inverse of the secondary path. According to this equation, the system can become unstable and ineffective if H(z) = 0 and P(z) = 0 respectively.

Secondary Path Estimation and FXLMS Algorithm

To account for the effect of the secondary path and to ensure convergence of the algorithm, the input is filtered by a secondary-path model C(z). The resulting algorithm is known as the FXLMS Algorithm and its block diagram is shown in Figure 11.

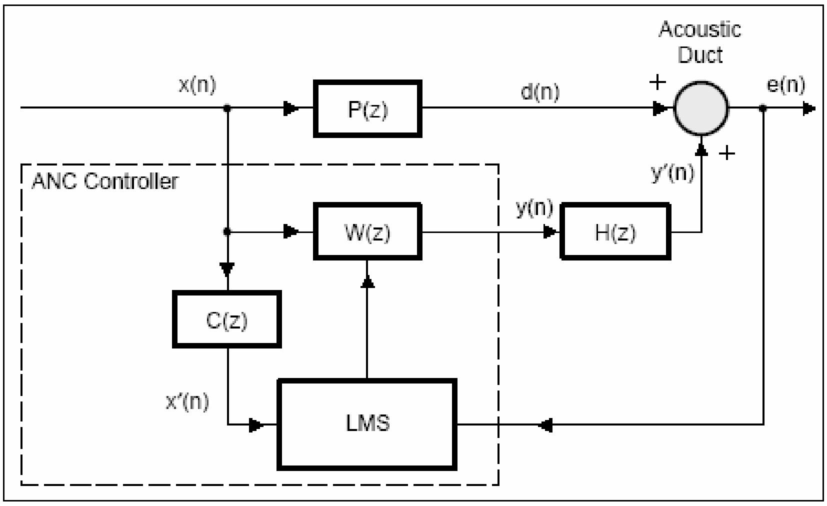

Figure 10: Block Diagram of FXLMS Algorithm. [2]

The FXLMS algorithm can be expressed as:

$$w(n+1) = w(n) - \mu e(n)x(n)h(n)$$

where μ is the step size of the algorithm that determines the rate of convergence. In practice, H(z) is unknown and estimated by C(z). So:

$$w(n+1) = w(n) – \mu e(n)x'(n)$$

where

$$\underline{x}'(n – 1) = [x'(n – 1)\ x'(n – 2) \ldots x'(n – N + 1)]^T$$

$$x'(n) = c^T x(n) = \sum c_i\, x(n – i)$$

and the coefficient vector of the secondary-path estimate, C(z) is:

$$\underline{c} = [c_0\ c_1\ \ldots . C_{M-1}]$$

Feedback path F(z) is the feedback path from the loudspeaker upstream to the input microphone. The concept of estimating F(z) is similar to the estimation of H(z) discussed.

Filtered-U Recursive LMS (RLMS) Algorithm

For the Anti-Snoring Pillow system, an adaptive infinite impulse response (IIR) filter will be used. Since IIR structure has the ability to model transfer functions with poles and zeros, this approach removes poles introduced by acoustic feedback by the poles of the adaptive IIR filter. It also dynamically tracks changes in the primary and secondary path in a real time manner. The recursive LMS (RLMS) algorithm for the IIR filter will be used for the Anti-Snoring Pillow.

The standard RLMS algorithm will also require modification to compensate for the secondary feedback path as discussed above. Figure 12 shows the block diagram of the algorithm.

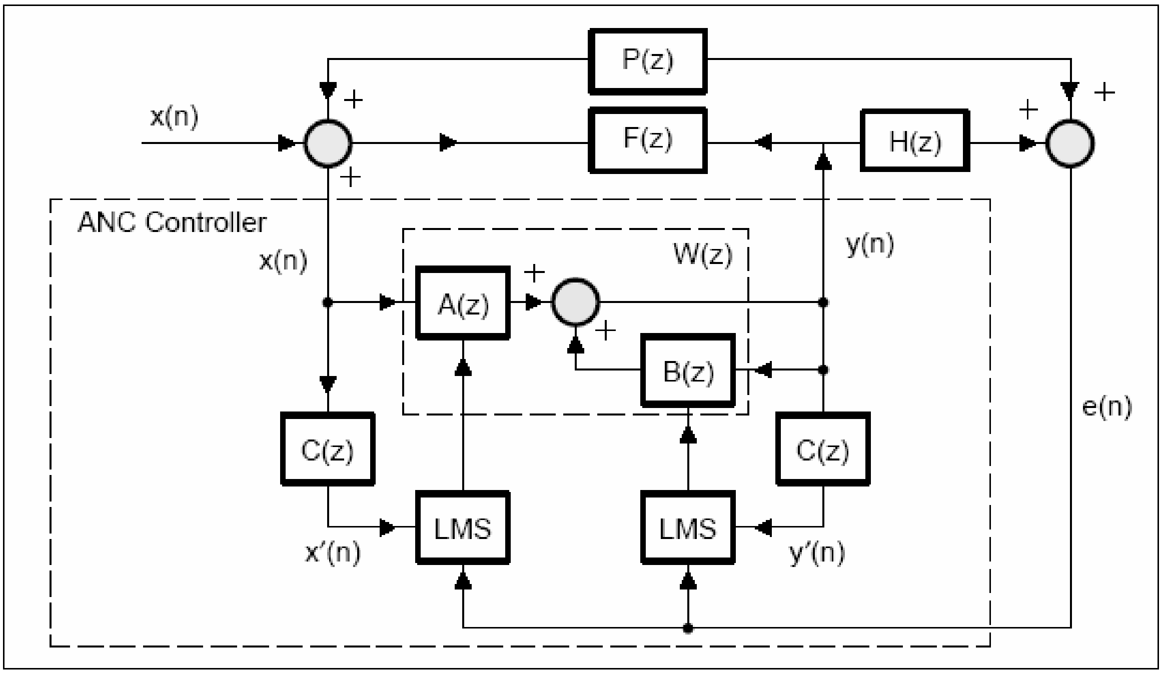

Figure 11: Block Diagram of the RLMS Algorithm. [2]

y(n) is the output signal of IIR filter computed by:

$$\mathbf{y(n) = \underline{a}^T(n)\ \underline{x}(n) + \underline{b}^T(n)\ \underline{y}(n-1) = \sum a_i(n)x(n-i) + \sum b_j(n)\ y(n-j)}$$

where,

a (n) = [a0 (n) a1 (n) … aN – 1 (n)]^T is the weight vector of A(z) at time n

b (n) = [b1 (n) b2 (n) … bM (n)]^T is the weight vector of B(z) at time n
y (n – 1) = [y (n – 1) y (n – 2) … y (n – M)]^T is the signal vector containing output feedback with one delay.
N = order of A(z)
M = order of B(z)

The filtered-U RLMS algorithm can be expressed by two vector equations for adaptive filters A(z) and B(z) as follows:

$$\mathbf{a(n-1) = \underline{a}(n) - \mu e(n)\, \underline{x}'(n)}$$

and

$$\mathbf{\underline{b}(n-1) = \underline{b}(n) - \mu e(n)\, \underline{y}'(n-1)}$$

where:

$$\mathbf{\underline{y}'(n-1) = [y'(n-1)\; y'(n-2) \ldots y'(n-M)]^T}$$

$$\mathbf{y'(n) = \sum c_j y(n-j)}$$

and

$$\mathbf{\underline{x}'(n-1) = [x'(n-1)\; x'(n-2) \ldots x'(n-N+1)]^T}$$

$$\mathbf{x'(n) = c^T x(n) = \sum c_i\, x(n-i)}$$

When A(z) and B(z) has converged, the error signal e(n) is equal to zero. Then:

$$W(z) = \frac{A(z)}{1 - B(z)} = \frac{-P(z)}{H(z) - P(z)\; F(z)}$$

The solutions for A(z) and B(z) are not unique due to the complexities of W(z). One possible solution suggested [4] is:

$$A^*(z) = \frac{-P(z)}{H(z)}$$

and

$$B^*(z) = \frac{P(z)\; F(z)}{H(z)}$$

After off-line modeling (discussed in the next section) of H(z) and F(z) , the ANC system is operated in noise cancellation mode. The Filter-U RLMS algorithm is summarized as follows [2]:

1. Input the reference signal x(n) and the error signal e(n) from the input ports.

2. Compute the antinoise y(n):

$$y(n) = \sum_{i=0}^{N-1} a_i(n)x(n-i) + \sum_{j=1}^{J} b_j(n)y(n-j)$$

where N is the order of the filter A(z) and J is the order of the filter B(z).

3. Output the antinoise y(n) to the output port to drive the canceling speaker.

4. Perform the filtered-U operation:

$$x'(n) = \sum_{i=0}^{M-1} c_i x(n-i)$$

and

$$y'(n) = \sum_{i=0}^{M-1} c_i y(n-i-1)$$

where M is the order of the filter C(z).

5. Update the coefficients of the adaptive filters A(z) and B(z) using the filtered-U RLMS algorithm:

$$a_i(n+1) = a_i(n) + \mu_a e(n)x'(n-i), \quad i = 0, 1, \ldots, N-1$$

and

$$b_j(n+1) = b_j(n) - \mu_b e(n)y'(n-j), \quad j = 1, 2, \ldots, J$$

6. Repeat the algorithm for the next iteration.

Off-line Modeling of Secondary and Feedback Path

The transfer functions H(z) and F(z) are unknown and time-variant in reality. This means online modeling techniques must be used to estimate them. If we assume that they are unknown but time invariant, an off-line modeling technique can be used during a training stage. A detailed algorithm for off-line modeling can be found on page 22-23 of "*Design of Active Noise Control Systems With the TMS320 Family*" [2].

6.4 Sampling time

The audio signal received by the microphone is initially an analog input that must be first converted in order to be processed. When Analog-to-Digital conversion occurs, it is important we consider the sampling rate of the whole system. In order to have sampled enough signals for doing the ANC algorithm, it is important we satisfy the Nyquist sampling theorem. The theorem tells us that the sampling frequency must be at least twice the highest-frequency component f in a signal, so that:

$$f_S > 2f$$

To satisfy this condition, we assumed snoring was roughly under 400Hz, and we needed a signal that was 100 times more clear in order for ANC to be efficient. With a sampling rate of 48kHz, we are able to reconstruct the shift version of this signal, which explains our choice of the AIC23 stereo codec.

6.5 Reason for using TMS320C6713

The high speed, high performance DSK which contains the TMS320C6713 DSP is ideal for the development of our product.

This DSP "brought a much more capable, stable, and robust DSP development environment to both university and industry engineers"[11] according to an article by Morrow M.G.

The DSP has 16Mbytes of synchronous DRAM, 512 Kbytes of flash memory which is more than enough for our product. This DSP has proven functional for both novice and beginners which will prove useful in the future for further advancement in our product. With this DSP we can create a stand-alone window application directly linked to the DSK which was not available in the older models.

The DSP contains many features which makes it very versatile to our needs. The multiple I/O ports for microphone and line in and out speakers prove to be vastly practical for our system. Additionally, the DSP includes 4 LEDs and DIP switches which will be later used for the user interface.

This DSP includes many helpful tools to facilitate the manipulation of the input signals. For example, this kit contains a customer support guide, technical reference and quick start guide. Also included in this kit is the software which is compatible with windows to implement the DSP. The DSP is compatible with programming in C/C++ which minimizes the time spent learning unknown languages.

The universally powered DSP is proves sensible in its power consumption. Another requirement this DSP fulfills is its price. Therefore, with this DSP we can satisfy our functional and affordable requirements.

The Texas Instrument TMS320 series are widely used for many applications for speech digitization/compression, pro-audio, medical, military and diagnostic. This results in wide variety of online resources and in the library. The success found with these previous applications has proven its reliability. Specifically, the applications with speech digitization/compression and speech synthesis were the most appealing compared to other models.

The TMS320C6713 DSK is flexible and efficient at handling complex signals which will achieve peak performance for our product.

7. Pillow

The overall structure of a pillow is important for a good nights rest. The structure of our pillow will maximize comfort for a good nights rest. To receive an optimal night's rest the pillow should provide a firm support under the neck. The cotton fill used to fill the pillow should mold to the users shape so it cradles their neck thus creating a personalized mold.

The casing for the cotton fill is flexible and durable to protect and increase the longevity of the pillow.

Our pillow design will incorporate the support the neck needs and our anti-snoring system to produce a product which results in peaceful sleep.

The structure of the pillow will provide both support and comfort. Speakers that will transmit the manipulated signals will be located inside the pillow. The speakers will be carefully positioned so no discomfort is brought upon the sleeper. The speakers will be located at the end of each pillow and be directed towards the sleeper's ears. The location of the speakers is shown in Figure 7.2. In the future, more time and money can be spent towards research to create the optimal pillow for each individual user.

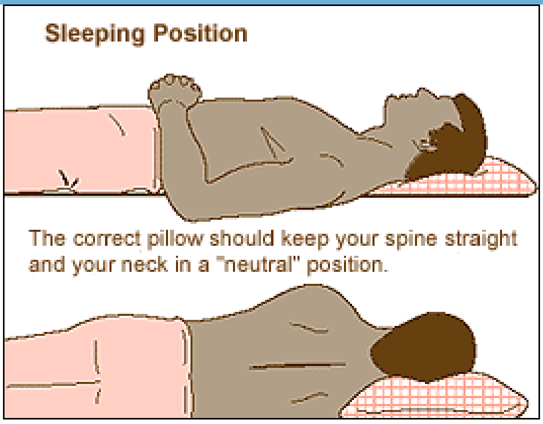

Figure 12: Proper sleeping position [1]

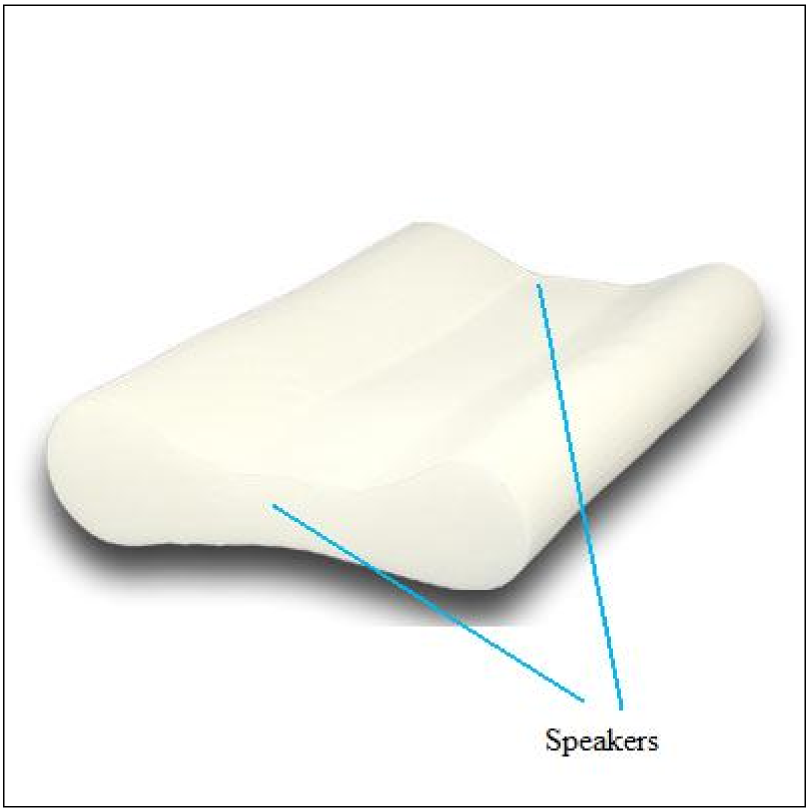

Figure 13: Speaker Positions[10]

8. System Setup

The ANC algorithm we are programming into the DSP will depend on the way the environment is setup. Some important factors to consider are:

- Control source arrangement

- Error sensor arrangement
- Quality of reference signal
- Control system performance

These factors are ordered from the most to the least important factor in determining the optimality of the system. These factors should be optimized in the defined order.

8.1 Control Source Arrangement

In our case, finding the most optimal position for the control source (the speakers) is a complex process. Although the same noise control can be achieved even if the source is not placed at the optimal position, the speakers have to be driven much harder in

practice. In general, finite element analysis and genetic algorithm optimization are required to find these optimal positions.

Finite element analysis is used to calculate the system modal characteristics, in order to find the sound field generated by the primary and control sources. This optimization process should be half analytical, and half trail and error. In our case, due to the lack of acoustic measuring equipments, we are going to make good approximations for the analytical portion of this optimization process.

Also, this optimization process only applies to one frequency. We have decided to pick the center frequency of snoring and find the optimal position based on that. More trail and error experimentation will be needed to optimize our device.

Lastly, in order to obtain substantial amount of noise reduction, the control source signal must be coherent with the primary source output, the separation distance between the two must not be too far, and the control source must be capable of producing the same volume velocity at the frequencies to be cancelled.

8.2 Error Sensor Placement

The optimal position for the error microphone can be found only after a location for the control source has been chosen. The error microphone location is important for the system to be functional at large distances from the source. The optimally controlled sound field may be one that minimizes the sum of the squared sound pressures at specific locations or the total potential energy in a closed space. The optimum error sensor locations are the locations of greatest difference in acoustic pressure levels between the primary and controlled sound fields. Again, this process will be done with trial and error.

8.3 Reference Signal Influence

The delay for the unwanted noise signal to travel between the reference sensor location and control source location must be greater than the electronic group delay of the whole system in order to notice any performance within the system. The total group delay through the controller includes delays associated with A/D interface, the signal processing, and the electroacoustic conversions associated with the reference sensors and the control sources.

Secondly, the quality of the reference signal greatly influences the amount of noise controlled within the system. The quality of the reference signal is governed by the following:

- coherence between the reference and error signals
- feedback of the control signal to the reference signal
- relative magnitude and character of the main frequency components that make up the reference signal

We have experimentally tested and adjusted our setup by adding delays and changing the placements of the error microphone, in order to achieve an optimal point of these 3 factors.

9. User Interface

To conserve power during the day the Anti-Snoring pillow system will have an on/off button. This power button will be clearly labeled on our product. The system should always be turned off whenever not in use to retain maximum life span. The power button is directly connected to the microcontroller which results in a direct shutdown.

Another item on the user interface would be a LED which will act as a status indication. The LED will light up to indicate an error in the system thus informing the user to turn off the system and wait a couple seconds before turning it back on again.

10. System Test Plan

Test cases to be executed will be separated in 4 main categories: Functional Testing, Interface Testing, Performance Testing and Acceptance Testing.

10.1 Functional Testing

10.1.1 Normal Case 1

Procedures:

1. Anti-Snoring device is switched on
2. User will put his head on the pillow, in between the two speakers
3. Demo snoring noise will start playing
4. User rotates his head in all directions

Expected Result:

- User does not notice any negative effects

10.1.2 Normal Case 2

Procedures:

1. Anti-Snoring device is switched on
2. Device is left on for 20 hours

Expected Result:

- Device still remain functional without over-heating

10.1.3 Extreme Case 1

Procedures:

1. Anti-Snoring device is switched on
2. User will put his head on the pillow, in between the two speakers
3. Demo snoring noise will start playing
4. User moves his head out of the pillow

Expected Result:

- Snoring noise should not be louder than usual

10.2 Performance Testing

10.2.1 Normal Case 1

1. Anti-Snoring device is switched on
2. User places his head on the pillow

3. Demo snoring noise starts playing

Expected Result
- User should notice at least 5-10 dB of improvement

10.2.2 Extreme Case 1

1. Anti-Snoring device is switched on
2. User will put his head on the pillow, in between the two speakers
3. Demo snoring noise will start playing
4. User lifts his head out of the pillow

Actual Result
- Device should not worsen the snoring noise

10.3 Acceptance Testing

10.3.1 Normal Case 1

Procedures:

1. Anti-Snoring device is switched on
2. User puts his head on the pillow
3. Demo snoring noise will start playing
4. User attempts to relax and possibly fall asleep

Actual Result:
- User should find the level of noise still acceptable and feel a noticeable improvement

10.3.2 Normal Case 2

1. User puts his head on the pillow

Actual Result
- User finds the comfort of the pillow acceptable to be employed in everyday use

11 Conclusion

This document has clearly outlined the proposed design for the Anti-Snoring Pillow. In the actual development phase, the design will be followed as closely as possible to meet the requirements of the functional specification. The test plans described will ensure that all the required functionalities are present in the product.

12 References

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