A SIMULATOR FOR
TRANSIENT EVOKED OTOACOUTIC EMISSION

A dissertation submitted in

partial fulfillment of the requirements for the degree of

Master of Technology

by

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M.Tech. Dissertation Approval

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ABSTRACT

Transient evoked otoacoustic emission (TEOAE) audiometry is a fast and objective procedure for screening of hearing loss as it does not require responses from the subject. Aim of this project is to develop a TEOAE simulator which can mimic the response of an ear to a transient stimulus for use in testing and calibration of TEOAE audiometers. For simulating different types of ear, dominant frequency components present in the TEOAE response should be selectable, and the latency and the level of dominant frequency component should be in accordance with the stimulus intensity. A microcontroller based instrument is developed for the simulator along with a PC based graphical user interface to set the parameters of the TEOAE response. To test the simulator, electrical part of a TEOAE audiometer is also developed, using a microcontroller based circuit for stimulus generation and response acquisition and serial port interface to a PC for further signal processing and spectral analysis.

CONTENTS

LIST OF ABBREVIATIONS

Abbreviation	Term
ADC	analog-to-digital converter
BPF	band-pass filter
DAC	digital-to-analog converter
dB	decibel
DPOAE	distortion product otoacoustic emission
DSP	digital signal processing
IC	integrated circuit
LPF	low-pass filter
NB	narrow band
OAE	otoacoustic emission
PC	personal computer
PGA	programmable gain amplifier
SFOAE	stimulus frequency otoacoustic emission
SOAE	spontaneous otoacoustic emission
SPI	serial peripheral interface
SPL	sound pressure level in dB with pressure reference of 20 μPa
SRAM	static random access memory
TEOAE	transient otoacoustic emission

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Chapter 1
INTRODUCTION

1.1 Background

Otoacoustic emissions (OAEs) are acoustic signals generated by the cochlea and can be used to monitor its functioning. The signal may be generated spontaneously or evoked by an acoustic stimulus. Spontaneous emissions have been observed in 60% of the population, while the evoked responses are observed in all healthy ears [1]. OAEs are low level signals compared to the applied stimuli. The measurement system with an ear probe contains a speaker to deliver stimulus to the ear canal, and a sensitive microphone to record the response. Transient evoked otoacoustic emission (TEOAE) is generated in response to a transient stimulus. It is useful in screening because it contains response to a stimulus over the entire frequency range of the cochlea simultaneously. The stimulus is generally in the form of a square pulse of 80 µs pulse width [1]. TEOAE response of normal ear consists of a few dominant frequency components. Each component has latency with respect to the stimulus onset and it is related to the centre frequency of the dominant component and the intensity of the stimulus. The response is severely corrupted by noise and artifacts. The amplitude of stimulus artifact has a nonlinear relationship with the amplitude of input stimulus [2][3]. Averaging and filtering of response is needed to separate OAE from noise. The recorded response is also corrupted with stimulus artifact which is linearly related to the stimulus. It is removed either with the help of nonlinear differential averaging method or linear averaging with windowing [1]. The response spectrum is analyzed to find relevant activity of cochlea in different frequency ranges. A TEOAE simulator is required to test linearity, sensitivity, and dynamic range of a TEOAE audiometer during its development stage.

1.2 Project objective

Objective of the project is to develop a TEOAE simulator which can mimic the response of an ear to a transient stimulus for use in testing and calibration of TEOAE audiometers. It is designed using a microcontroller. It consists of a stimulus acquisition module, an ear response delivery module, and a communication module for setting the response parameters. The ear response at required intensity level is generated using a PC-based GUI. The generated response is transferred to the microcontroller internal memory and is delivered whenever a stimulus is detected. A TEOAE audiometer is also developed to test the simulator. It consists of a stimulus delivery module, a response acquisition module, a communication module, and a GUI based signal processing module.

1.3 Dissertation outline

Chapter 2 gives an introduction to otoacoustic emissions. Realization of TEOAE simulator is presented in the third chapter. Chapter 4 describes the simulator hardware and software. Chapter 5 describes TEOAE audiometer hardware and software. Chapter 6 presents the test results. The last chapter provides a summary of the work and some suggestions for further work. Appendix A describes the low-pass filter design used in TEOAE audiometer. Appendix B gives information regarding some of the commercially available TEOAE audiometer and probes.

Chapter 2
OTOACOUSTIC EMISSIONS

2 .1 Introduction

The outer hair cells of the cochlea have more efferent innervations than the afferent ones. Voltage-dependent vibration of the outer hair cells enhances mechanical vibration of the basilar membrane, thus enhances the hearing sensitivity of the ear. So the cochlea both receives and produces acoustic energy during normal hearing process. Gold in 1948 had proposed a hypothesis that the sharp frequency selectivity showed by the cochlea resulted from a feedback system. This feedback system consists of signals from the cochlea to the auditory center (mechanical vibrations to electrical signals) coupled with signals from the auditory center to the cochlea (electrical signals to mechanical vibrations). Gold's hypothesis was proved by Kemp in 1978 by recording the acoustic vibrations emitted by cochlea in the ear canal. These acoustic emissions by the cochlea are known as otoacoustic emissions (OAEs) [1] - [3]. Its recording can be used for obtaining objective information of mechanical activity specific to sensory elements of the organ of Corti in the cochlea. The majority of the peripheral hearing dysfunctions are related to the sensory mechanism of cochlea, and may occur due to noise induced hearing loss, ototoxicity and hereditary hearing loss [1]. Use of OAE for the diagnosis of hearing disorder is known as OAE audiometry. OAE can be spontaneous or evoked by acoustic stimuli. The evoked OAEs are grouped on the basis of the stimulus used as (i) stimulus frequency, (iii) distortion product, and (iii) transient evoked. An acoustic probe consisting of a sensitive microphone and an option to present acoustic stimuli is used to acquire OAE. The probe is coupled to the external ear. Compared to the stimuli, the emission is a very low level signal. Therefore, the sensitivity and noise of the transducer are the major concerns in the probe design.

2.2 Spontaneous otoacoustic emission (SOAE)

SOAE is generally a narrow-band signal and can be recorded at ear canal by placing a sensitive microphone. This acoustic vibration is emitted over a long period of time. A probe with a microphone is used to record SOAE as shown in Figure 2.1. A good seal of the microphone with the ear canal is essential for efficient recording. The acoustic background noise in the ear canal is dominated by low-frequency noises from subject's body such as blood flow, breathing, muscle contraction, and mandibular joint movements. Spectral analysis is used to detect the presence of SOAE components over the noise floor. A high-pass filtering

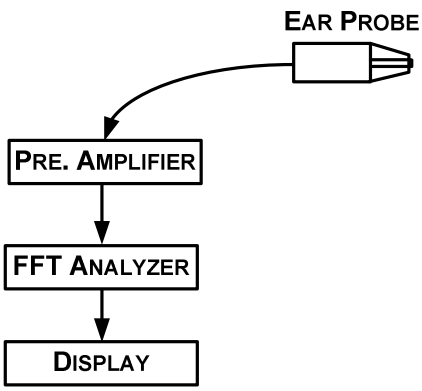

Figure 2.1: Block diagram of SOAE measurement system, adapted from [1].

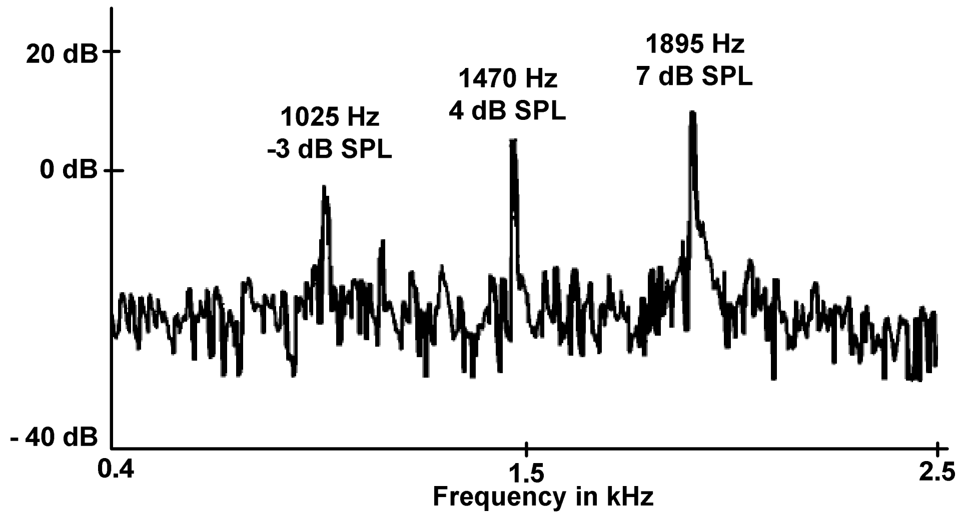

Figure 2.2: Power spectrum of SOAE with three components, adapted from [1].

with a cutoff of 400 Hz followed by FFT analysis and spectral averaging is used to enhance the spectral peaks. Figure 2.2 shows power spectrum of the signal recorded in the ear canal with three SOAE components.

2.3 Evoked OAEs

Evoked responses are grouped, on the basis of stimulus used to evoke them, into three types: stimulus-frequency otoacoustic emission (SFOAE), distortion-product otoacoustic emission (DPOAE), and transient-evoked otoacoustic emission (TEOAE). These responses occur with a definite latency with reference to the stimulus and represent the mechanical response of the cochlea [3].

2.3.1 Stimulus-frequency otoacoustic emission (SFOAE)

A pure tone acoustic stimulus of low level (below 25 dB SPL) can elicit an additional acoustic energy from the cochlea at the same frequency and this response is called stimulus-frequency otoacoustic emission (SFOAE). The evoking stimulus consists of a pure tone that is swept slowly over a settable frequency range. A SFOAE system based on lock-in amplifier

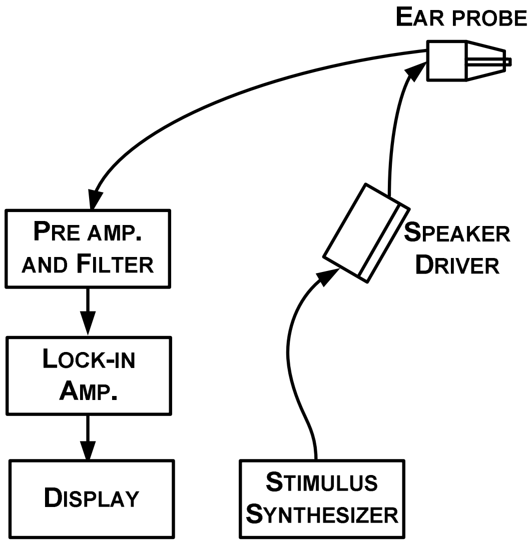

Figure 2.3: Block diagram of SFOAE measurement system, based on [1].

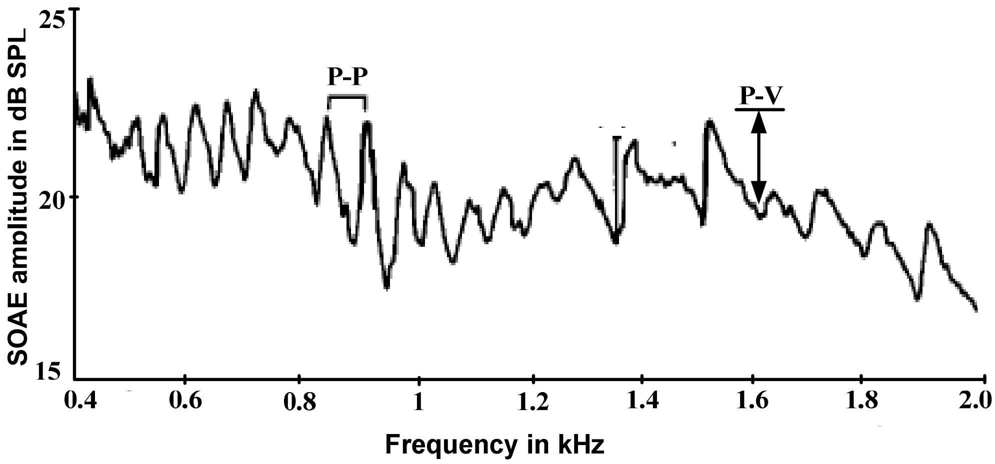

Figure 2.4: Recorded response at ear canal using SFOAE measurement system, adapted from [1].

technique is shown Figure 2.3. This system delivers a pure tone of 20 dB SPL. The frequency is swept slowly over about 150 s, from 0.4 kHz to 2.0 kHz. The recorded response is plotted as shown in Figure 2.4. The frequency-dependent phase changes due to latency between evoking stimulus and the ear response result in frequency regions with peaks and valleys [1]. Peak-to-peak (P-P) frequency span and peak-to-valley (P-V) amplitude measured from this plot are of diagnostic importance.

2.3.2 Distortion product OAE (DPOAE)

Nonlinear response of the cochlea can be assessed by applying two pure tone stimuli simultaneously to the ear canal using an ear probe. The recorded response consists of new frequencies that are not present in the evoking stimuli. The frequency of distortion products is related to the frequency of the two primary stimuli f_1 and f_2, with the largest distortion

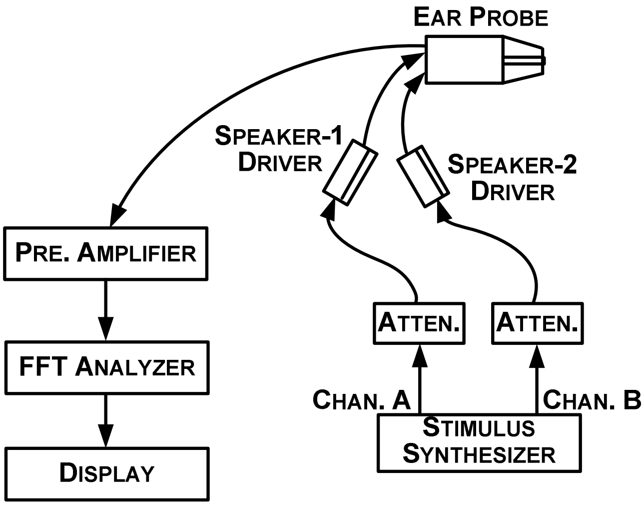

Figure 2.5: Block diagram of DPOAEs measurement system, based on [1].

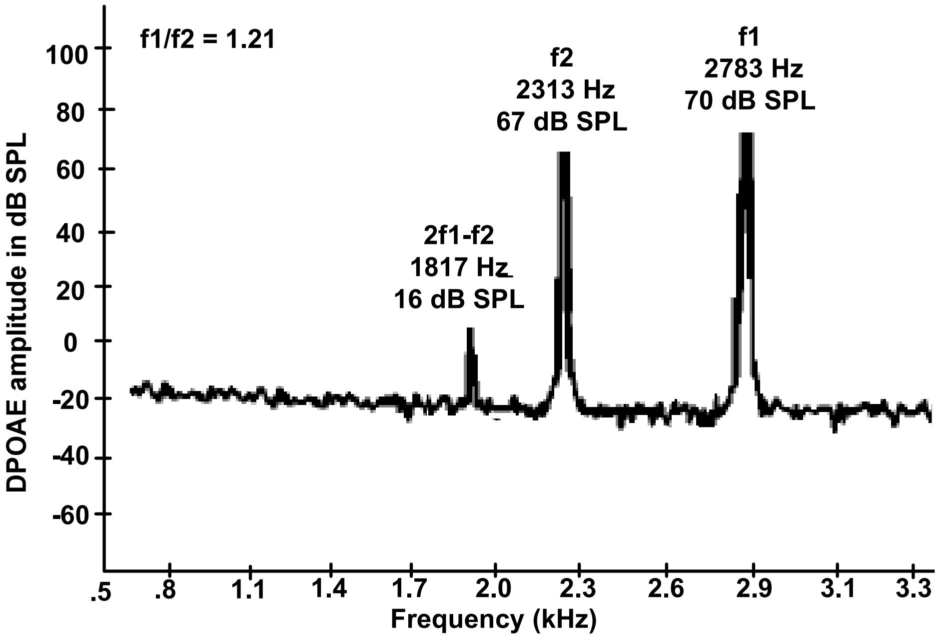

Figure 2.6: Power spectrum of recorded ear response using DPOAE measurement system [2].

product occurring at the frequency $2f_1 - f_2$ [1]. A measurement system with highly linear characteristic in the measuring range is needed to avoid generation of distortion product by itself. The dynamic range of such a system should be 80 dB over the audio frequency range. Two pure tone stimuli can be mixed either electronically or acoustically. In case of electronic mixing, one transducer is used to deliver the sum of two stimulus waves as the stimulus and two transducers are used for acoustic mixing. Generally the second method as shown in Figure 2.5, is less prone to the nonlinearity interactions [1]. Frequency f_1 is delivered through speaker-1 driver and frequency f_2 is delivered through speaker-2 driver and the ratio f_1/f_2 is kept at 1.21 [2]. The stimuli are applied at 70 dB SPL for 100 ms, and the microphone output is sampled for 96 ms. The spectrum of synchronously averaged 32

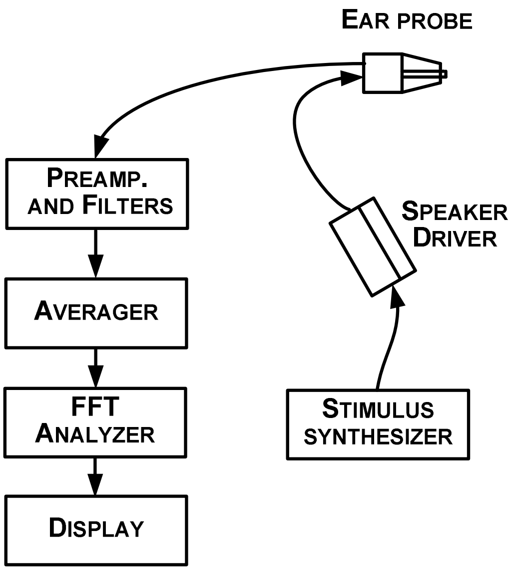

Figure 2.7: Block diagram of TEOAE measurement system, based on [1].

responses is shown in Figure 2.6. An inter-modulation distortion product of $2f_1 - f_2$, not present in the delivered stimuli, is noted in the spectrum of ear response with an amplitude level greater than noise floor.

2.3.3 Transient evoked otoacoustic emission (TEOAE)

TEOAE is elicited by a brief acoustic stimulus in order to get the response over the entire frequency range of the cochlea simultaneously. It occurs with a certain delay after the transient stimulus. An acoustic probe consisting of a miniature speaker to present stimulus and a sensitive microphone to pick up response is used for acquisition. A flat frequency response over 0.3 to 8 kHz range is required within the ear canal for both the recording microphone and the stimulus eliciting speaker. The probe microphone can be used for monitoring waveform of the stimulus thereby assuring the adequate and consistent fit of the probe assembly within the ear canal.

A block diagram of the TEOAE instrument is shown in Figure 2.7. Stimulus locked time averaging of responses improves signal-to-noise ratio. Commonly used transient stimuli are rectangular, Gaussian shaped clicks, and tone bursts. For the recording of TEOAE, the microphone signal is averaged by time locking it to the stimulus. A high-pass filter with 300 Hz cut-off is used to eliminate low-frequency subject-generated noises. Ringing of ear canal happens due to the improper fit of OAE probe and it may be misinterpreted as a TEOAE response. Artifact rejection methods, such as nonlinear differential averaging and windowing are implemented in TEOAE measurement systems. Cancellation of linear artifacts is achieved by a special stimulus presentation protocol based on the nonlinear growth of TEOAE amplitude with stimulus intensity [1]. All components of TEOAE saturate at high intensity

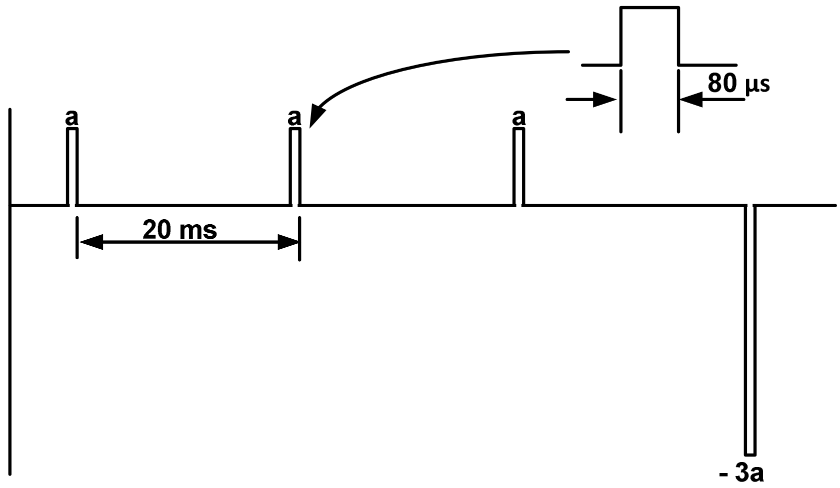

Figure 2.8: Nonlinear click stimulus protocol, adapted from [2].

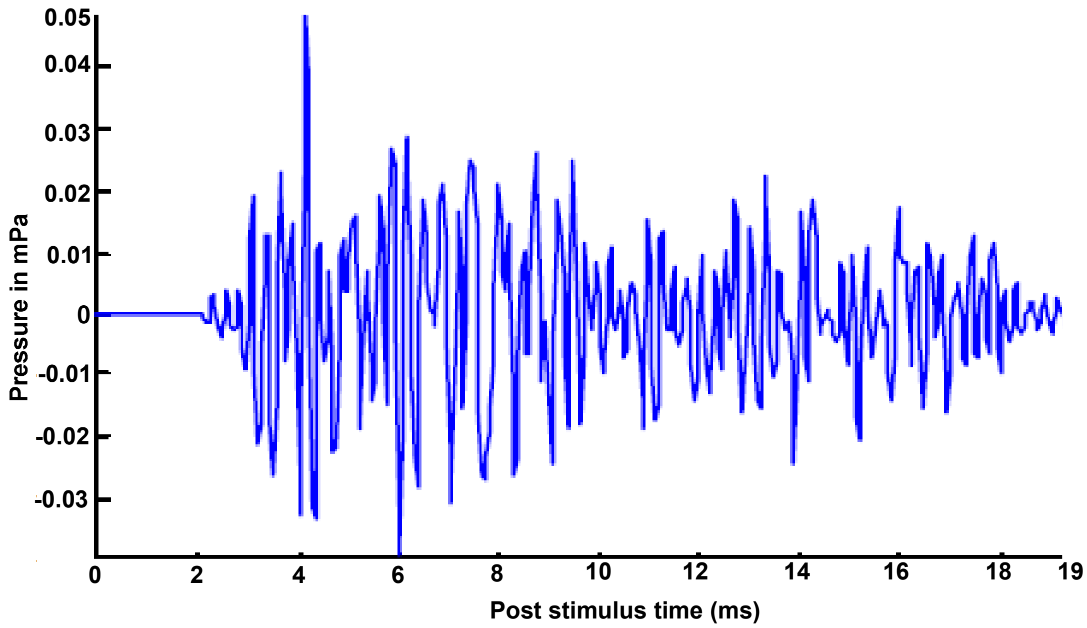

Figure 2.9: Windowed TEOAE response obtained from one block of four clicks [2].

level of 80 dB SPL and above. Kemp *et al* [2] introduced a set of four click stimuli as one block. It consists of three identical clicks of the same polarity and amplitude and a fourth click of opposite polarity and three times the amplitude of the former stimuli, as shown in Fig 2.8. Stimulus amplitudes are selected to have saturated response amplitudes, e.g. 80 dB for the first three clicks and 90 dB for the fourth one. The TEAOE response for each block is obtained by time-synchronous averaging of the responses for the four stimuli in each block. This averaging results in cancellation of linear stimulus artifact and ringing. It also results in cancellation of two TEOAE responses, but this disadvantage is less significant compared to the advantage due to cancellation of artifacts and it helps in accurately identifying the emission components [1]. The response is windowed from 2.5 ms to 20 ms as shown in

Figure 2.9 and such 256 responses averaged to improve signal-to-noise ratio. A band-pass filter followed by FFT analysis helps in interpretation of TEOAE components. Reproducibility of two subsequent averaged waveforms can be tested using cross-correlation techniques. A cross-correlation peak greater than 0.5 indicates the presence of TEOAE.

2.4 Characteristic of TEOAE response

Characteristics of TEOAE include nonlinear growth with the stimulus intensity and saturation at high levels of stimulation, presence of dominant frequency components and their frequency dependent latencies. It has been established that all normal hearing ears produce TEOAE during proper testing procedures [1]. Subject generated noises and equipment related problems pose difficulties in measurement of TEOAE parameters in clinical examination.

2.4.1 Amplitude growth of TEOAE

The amplitude of TEOAE depends on stimulus intensity level, the number and frequency of dominant emissions, and frequency response of the middle ear. The measurement of the peak-to-peak amplitude of TEOAE is useful only when it contains a single dominant frequency component. Generally TEOAE is composed of multiple dominant frequency components having distinct latencies, amplitudes, and durations.

Kemp [1] reported that TEOAE amplitude is related to square root of the stimulus intensity, with linear growth up to a stimulus level of 25 dB SPL/Hz and strong saturation above this level [1]. A numerical approximation of OAE growth rate before the saturation with respect to the click stimulus intensity is given in [4]. The growth rate of OAEs in normal-hearing ears is about 0.4 dB/dB, but can be smaller (0.1 dB/dB). As mentioned earlier, most of the TEOAE components saturate at 80 dB SPL click stimulus. An inversion of stimulus results in the inversion of TEOAE response [2]. Kemp *et al* [3] reported about 10 dB difference in TEOAE amplitude between infants and adults for same stimulus intensity level at the ear canal. The prevalence of the response declines by 35% for subjects of age above 60 due to gradual loss in hearing [1].

2.4.2 Duration of TEOAE

Duration of TEOAE ranges from few milliseconds to several hundreds of milliseconds. On the basis of duration, the responses can be grouped as short and long, with 20 ms after click onset used to differentiate between them [1]. About 33% of normal-hearing ears show short TEOAE and absence of SOAE is noted in these ears. SOAE is present in 50% of the ears with long TEOAE [1]. Frequency dispersion or amplitude modulation is evident in the waveforms of long TEOAE because of the presence of more than one dominant emission frequency. The dominant frequencies of SOAE synchronized by the transient stimuli become long TEOAE. Long TEOAE may also get generated as weakly damped oscillations at dominant frequencies after the stimulus presentation [1]. Duration of TEOAE is related to the number of emission frequencies in the response.

2.4.3 Spectrum of TEOAE

Spectrum of TEOAE depends on the spectral energy of the stimulus, and structurally dependent resonances of an individual ear [1]. For a click stimulus, most of the normal ears emit TEOAE with spectra containing several dominant frequencies [1]. Identical dominant emission frequencies are emitted by different stimuli. It is assumed that dominant frequencies are produced by emission generators located at fixed places in the organ of Corti. The number and tuning of dominant frequencies are different for different ears. The spectrum of TEOAE in ears with SOAEs contains dominant frequencies which are not detected as SOAE, generally seven distinct frequencies are noted in this case. The dominant frequencies show amplitude and tuning properties similar to those in SOAE. Dominant frequencies are also found in ears without SOAE and the properties of these dominant frequencies are similar to those present in ears with SOAE but the number of dominant frequencies generally reduces to four [1].

Dominant frequencies generally occur in the frequency range of 0.5 to 4 kHz. Responses to a transient stimulus can be synthesized by adding responses to tone burst stimulus placed at dominant frequencies. Kemp *et al* [1] reported absence of dominant frequencies in the spectrum of TEOAE of some normal hearing ears. They noted a spectral component in the form of a broadly tuned frequency band ($0.5 - 2.5$ kHz) in the TEOAE spectrum of these ears and the maximum amplitudes of broad-TEOAE components were located between 1 kHz and 2 kHz. Tone burst stimulus can be used to facilitate the detection of TEOAE in these ears [1].

2.4.4 Latency of TEOAE

Transiently evoked emission is present at the ear canal with a specific latency and the latency depends on the frequency components of the emission. The overlapping of stimulus tail with TEOAE response makes the latency measurement more difficult. The time of appearance of a component in TEOAE is related to its frequency. Tognola *et al* [4] used time frequency analysis to find the time frequency property of TEOAE. They defined the latency of each TEOAE component as the time interval between the stimulus onset and the occurrence of maximum amplitude of the component and noted that the latency measurement of each frequency component within the dominant frequency band was not possible. Experiment was conducted for 8 adult subjects having no previous history of ear disease. TEOAE measurement was carried out with the help of Otodynamics ILO88. Click stimulus was applied in linear mode and its level was varied from 47 to 68 dB SPL with step size of 3 dB

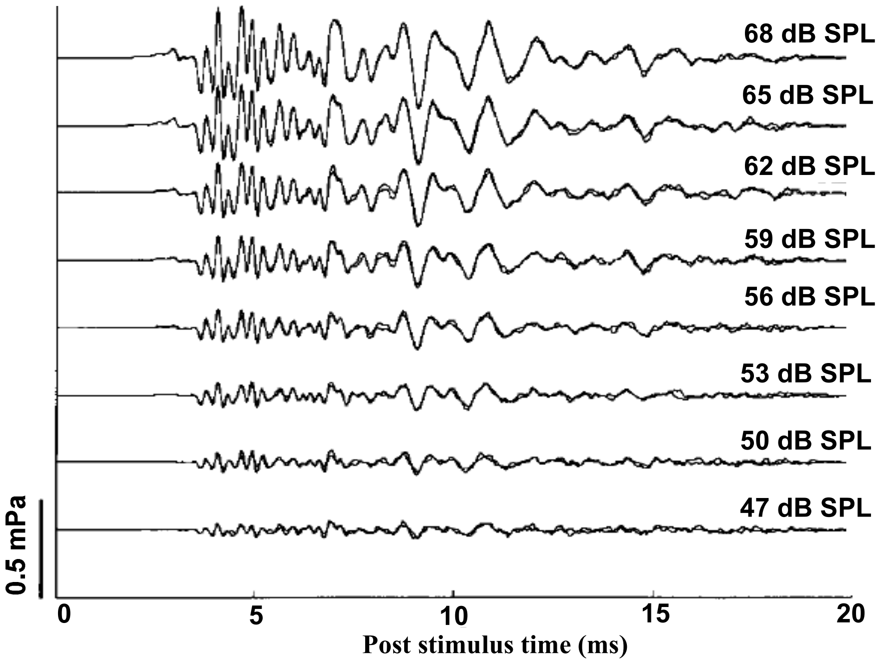

Figure 2.10: TEOAEs of a 25 year old subject for different stimulus levels, based on [4].

SPL. For each stimulus level, 1024 responses (each of 20 ms) were recorded. Synchronous averaging was carried out after band-pass filtering ($0.6-10$ kHz) and windowing (2.5 ms $-$ 20 ms). The responses for different stimuli after windowing and signal processing are shown in Figure 2.10. For measuring the latency, the response was decomposed into elementary components by wavelet transform as shown in Figure 2.11, using modulated cosine function

$$\psi(t) = (1/1 + t^{\beta}) cos\,(\alpha t) \tag{2.1}$$

as the mother wavelet, with $\alpha = 20$ and $\beta = 4$ chosen based on the best approximation of the stimulated response using gammatone functions [4]. The signal below the first trace in Figure 2.11 is the reconstructed signal from the wavelet derived components. The central frequency of each band is shown at the right side of panel in kHz. The dominance of low frequency components are around 1.0 kHz to 1.5 kHz and high frequency components are around 3.0 kHz to 3.5 kHz. By analyzing latency data of 8 subjects at different stimulus levels, the authors obtained an equation for relating the latency Ʈ of the elementary component with its central frequency *f*, the stimulus intensity level *i* (in dB divided by 100), as the following,

$$Ʈ = cd^{i} f^{e} \tag{2.2}$$

where *c*,*d*, and *e* are constants. The estimated value of coefficients are $c = 17.98$, $d = 0.36$, and $e = -0.44$. With data from all the subjects at all the intensities pooled together, the fit was good (Fisher, $P \leq 0.01$) with a correlation coefficient of 0.72 [4]. Figure 2.12 gives a plot of latency (ms) as a function of frequency, as a set of parellel lines with a slope of *e* and moving downward with increasing stimulus level.

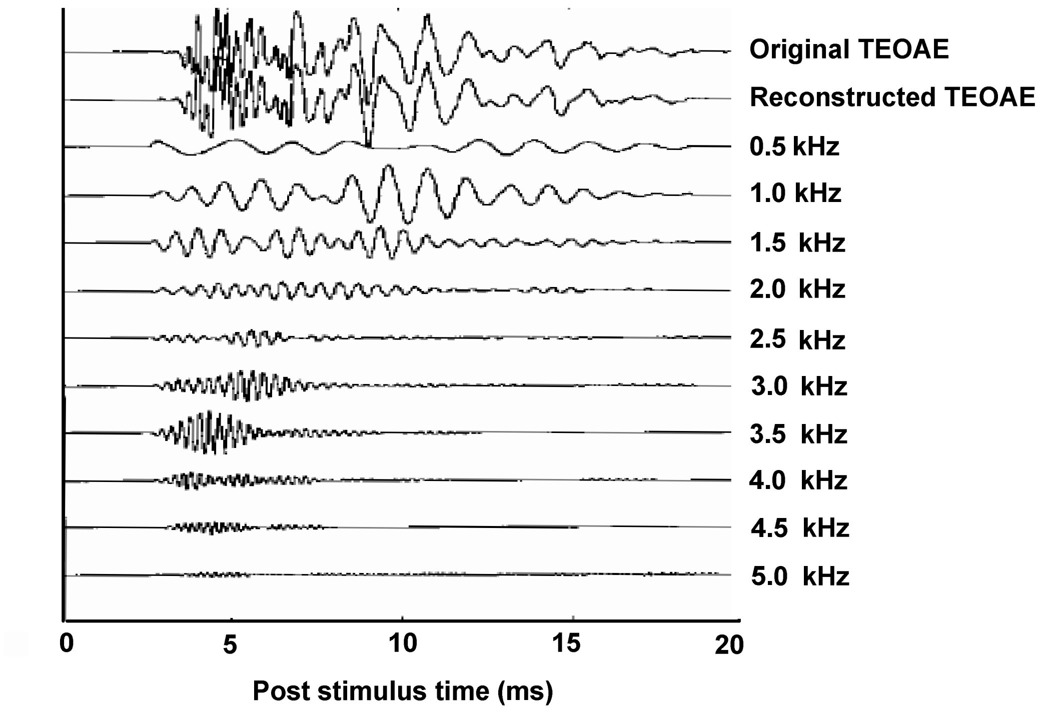

Figure 2.11: Decomposed frequency components of TEOAEs of 68 dB SPL stimulus, based on [4].

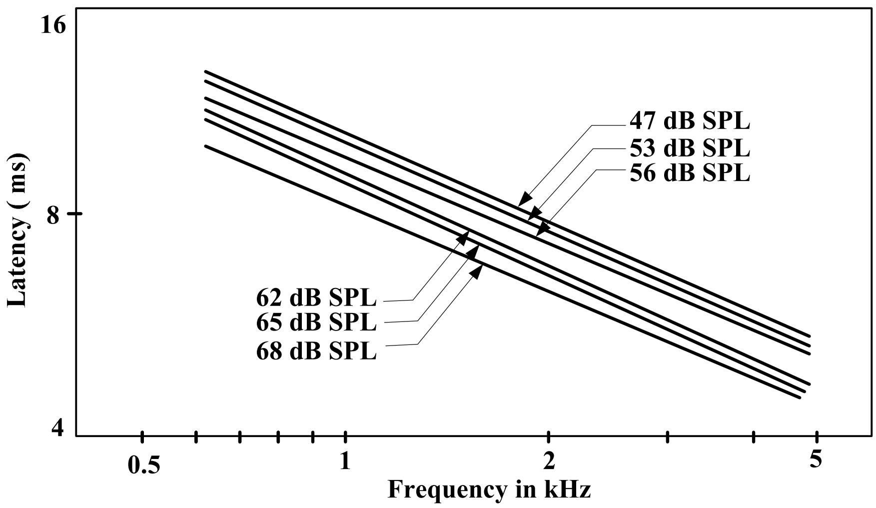

Figure 2.12: Latency versus frequency component of OAE at different stimulus intensities [4].

Chapter 3
SIMULATOR DESIGN

3.1 Introduction

The objective is to develop a TEOAE simulator which can mimic the response recorded from an ear to a transient stimulus (80 µs square pulse) for use in testing and calibration of TEOAE audiometers. Transient stimulus is used to get response from the entire frequency range of the cochlea. In addition to the emission from the cochlea, the response recorded from the ear contains artifact such as stimulus echo, ear canal response, and middle ear response and ringing due to improper fit of the probe. The growth of TEOAE with intensity of square pulse stimulus varies from ear to ear. The lowest rate noted with strong TEOAE is 0.1 dB/dB and the average value is around 0.4 dB/dB [2],[3]. A rate of 0.5 dB/dB is selected here for our TEOAE simulator design. All components of TEOAE response growth saturate at stimulus intensity of 80 dB SPL and intensities above it [1]. TEOAE response and stimulus artifact get inverted with inversion of the stimulus. In contrast to the TEOAE response, the stimulus artifact is linearly related to the stimulus level. The simulator should be able to generate a realistic response with settable parameters such as dominant frequency and ear type. The settable parameters can be controlled with the help of a graphical user interface (GUI). Simulator is designed to deliver TEOAE response for transient stimulus ranging from 70 dB to 95 dB in both positive and negative polarity. An option to add noise with the TEOAE response is provided to stimulate the environmental and instrumental noise during TEOAE recordings.

3.2 Ear model

By analyzing ear responses to transient stimuli, an ear can be modeled as a combination of a low-pass system and a nonlinear system as shown in Figure 3.1. The ear canal and middle ear act as a low-pass filter to acoustic signals. The cochlear response to the acoustic stimuli can be represented as a nonlinear system. The influence of environmental noise and subject generated noise is represented by a noise generator. As mentioned in the earlier chapter (subsection 2.4.3), the spectrum of ear response to transient stimulus consists of dominant frequency components [2]. The latency and amplitude of each dominant frequency component is related to the stimulus intensity. Each of these dominant frequency components can be generated by using a gammatone function [3] as the following,

$$g_i(t) = at^3 e^{-2\pi\beta f_i t} \cos 2\pi f_i t \tag{3.1}$$

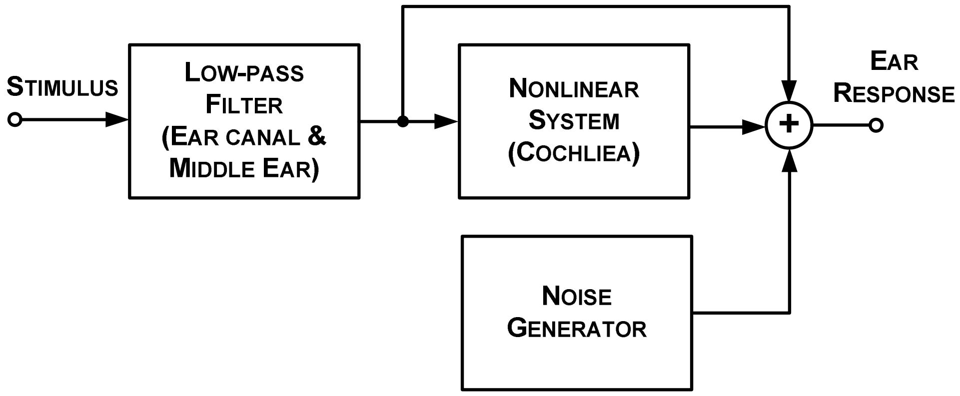

Figure 3.1: Model of an ear.

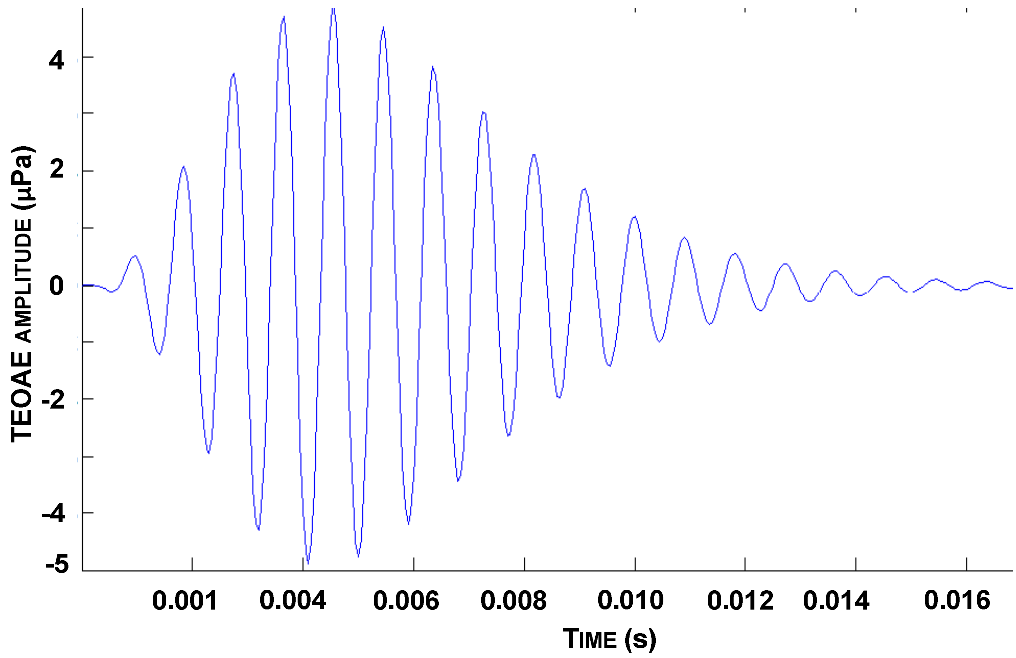

Figure 3.2: Synthesized response with dominant component at 1.5 kHz.

where, $a = (2\pi f_i)^{3.5}$, β = 0.2, and f_i is the centre frequency of the dominant component. A dominant frequency centered at 1.5 kHz is synthesized and shown in Figure 3.2. Its power spectrum is shown in Figure 3.3. As mentioned in the previous chapter (subsection 2.4.4), the latency of a dominent component is defined as the time interval between the stimulus onset and the peak of the component. It is related to the frequency of the component and the intensity of the stimulus. The latency of dominant frequencies at different intensity of click stimulus is calculated using Equation 2.2 and shown in Table 3.1. TEOAE response *s(t)* with more than one dominent frequencies can be synthesized by adding gammatones at the dominant frequencies as the following,

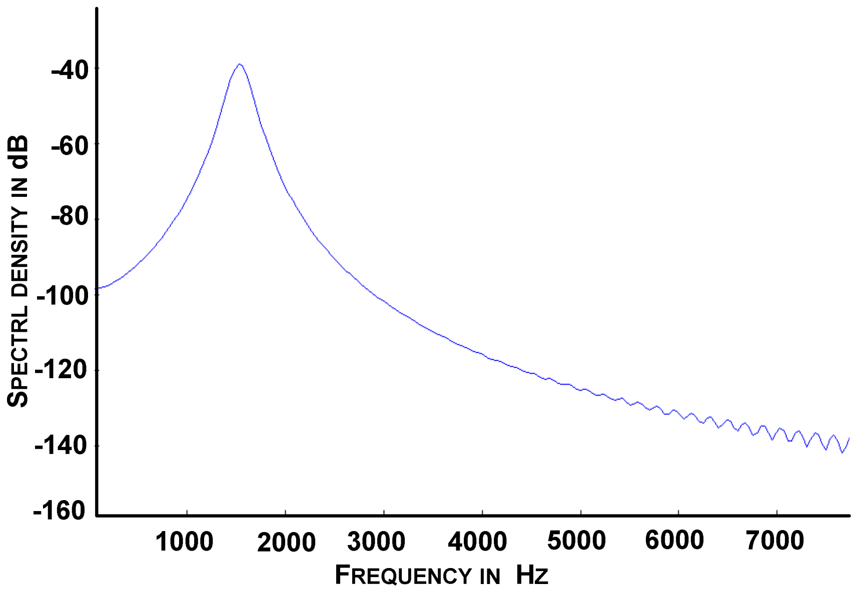

Figure 3.3: Power spectrum of synthesized response with dominant component at 1.5 kHz.

Table 3.1: Latency of dominant component at different click intensity.

Stimulus intensity (dB SPL)	1.1 kHz Delay (ms)	1.5 kHz Delay (ms)	2.2 kHz Delay (ms)	2.9 kHz Delay (ms)	3.9 kHz Delay (ms)
70	8.4331	7.3573	6.2163	5.5049	4.8321
71	8.3474	7.2826	6.1532	5.4489	4.7830
72	8.2626	7.2085	6.0906	5.3935	4.7343
73	8.1786	7.1353	6.0287	5.3387	4.6862
74	8.0954	7.0627	5.9674	5.2844	4.6386
75	8.0132	6.9910	5.9068	5.2307	4.5914
76	7.9317	6.9199	5.8467	5.1776	4.5448
77	7.8511	6.8496	5.7873	5.1249	4.4986
78	7.7713	6.7799	5.7285	5.0728	4.4528
79	7.6923	6.7110	5.6703	5.0213	4.4076
80	7.6141	6.6428	5.6126	4.9702	4.3628
81	7.5367	6.5753	5.5556	4.9197	4.3184
82	7.4601	6.5085	5.4991	4.8697	4.2745
83	7.3843	6.4423	5.4432	4.8202	4.2311

$$s(t) = \sum_{i=1}^{N} g_i(t) \tag{3.2}$$

TEOAE response synthesized with dominant frequencies at f_i = 1.10, 1.53, 2.10, 2.90, and 4.00 kHz is shown in Figure 3.4. Its spectrum is shown in Figure 3.5. Stimulus artifact representing echo and ringing is synthesized by applying a transient stimulus pulse (square pulse of 80 µs pulsewidth) to a unity gain digital lowpass filter with a cutoff of 6.50 kHz. The output signal from the low-pass filter is shown in Figure 3.6. The waveform samples are stored in memory and used to generate stimulus artifact. Generated TEOAE and stimulus artifact with respect to the stimulus are added by a summer module and delievered as the simulator output.

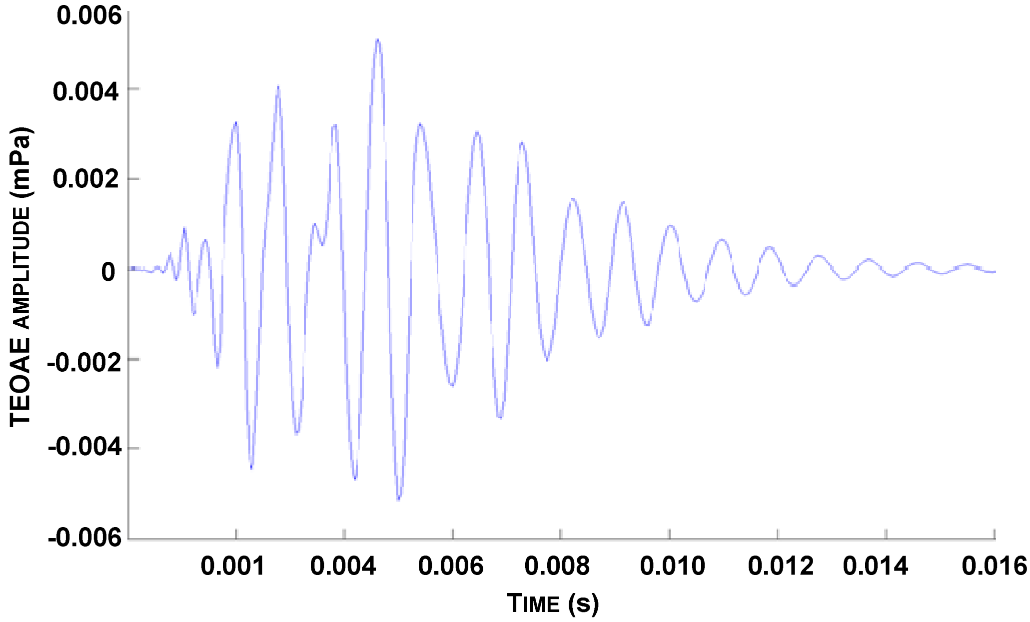

Figure 3.4: Synthesized TEOAE response waveform with five dominant components.

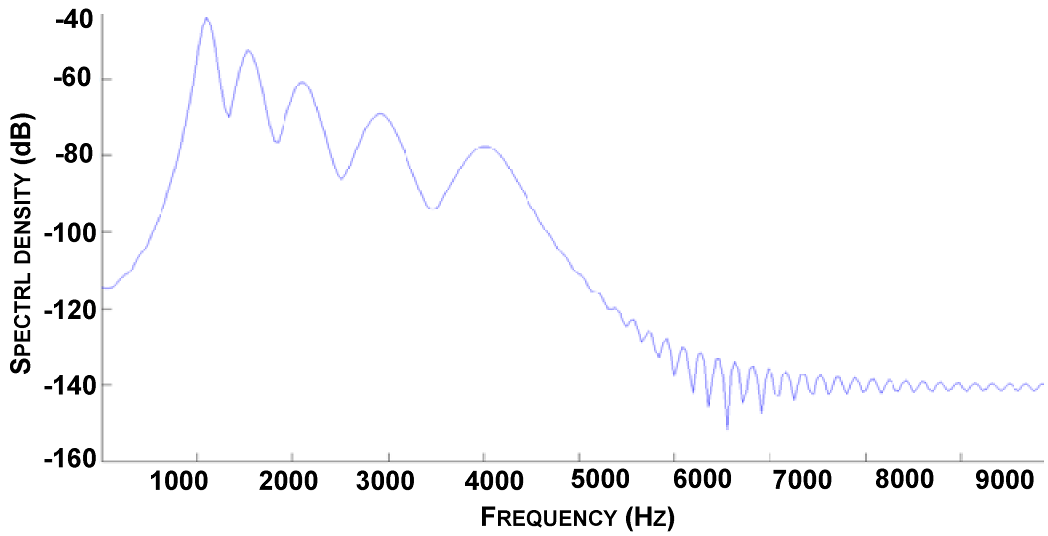

Figure 3.5: Power spectrum of TEOAE response waveform of Figure 3.4.

3.3 Realization of TEOAE simulator

A block diagram representation of the microcontroller-based TEOAE simulator is shown in Figure 3.7. The system consists of a stimulus acquiring module to detect transient stimulus (square pulse of 80 µs), stimulus analyzing module and ear response delivery module. A level shifter and ADC module of microcontroller are used to detect stimulus pulse. Stimulus analyzing module, as part of microcontroller program finds the polarity of pulse.

The ear response delivery module consists of the microcontroller program and its two-channel DAC, programmable gain amplifier (PGA), and a summing amplifier as adder. The left channel DAC of the microcontroller is used to deliver either stimulus artifact or

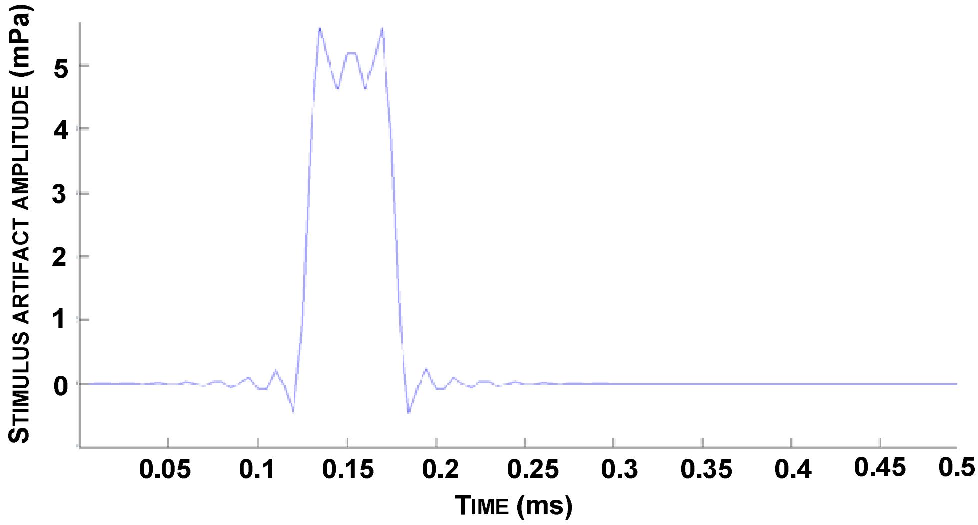

Figure 3.6: Synthesized Stimulus artifact waveform.

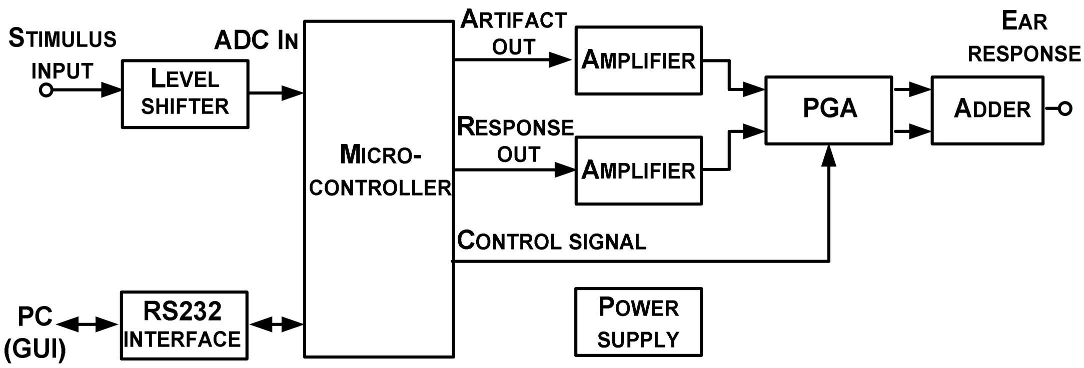

Figure 3.7 Block diagram of the TEOAE simulator.

generated noise and right channel is used to deliver TEOAE response. The maximum amplitude level of stimulus artifact and TEOAE response as output by DACs are same. The level difference between the stimulus artifact and TEOAE response is more than 50 dB as shown in Table 3.2. Such a large variation in the level of stimulus cannot be obtained using digital scaling of the values output through DACs. Hence external PGA is needed. For this purpose, we have used 2-channel programmable gain amplifier chip "Texas Instruments PGA2311". In this chip, gain of each channel can be varied independently from -95.5 dB to 31.5 dB with step size of 0.5 dB. Either artifact signal or noise is given to the left channel of PGA and TEOAE response is given to its right channel. The microcontroller controls the gain of each channel with respect to the stimulus level as shown in Table 3.3. The microcontroller communicates with programmable gain amplifier using SPI. A PC-based GUI is used to set the parameters such as stimulus type, stimulus intensity, type of ear, dominant frequency, noise type and its level, and ear canal coupling. Communication between PC and hardware

Table 3.2: TEOAE response and artifact levels for different stimulus levels, for neonatal and adults ears.

Sound pressure level equivalent (dB SPL)	Neonatal ear		Adult ear	
	Stimulus artifact level (dB SPL)	TEOAE response level (dB SPL)	Stimulus artifact level (dB SPL)	TEOAE response level (dB SPL)
70	70	45.0	70	35.0
71	71	45.5	71	35.5
72	72	46.0	72	36.0
73	73	46.5	73	36.5
74	74	47.0	74	37.0
75	75	47.5	75	37.5
76	76	48.0	76	38.0
77	77	48.5	77	38.5
78	78	49.0	78	39.0
79	79	49.5	79	39.5
80	80	50.5	80	40.5
81	81	50.5	81	40.5
82	82	50.5	82	40.5
83	83	50.5	83	40.5
84	84	50.5	84	40.5
85	85	50.5	85	40.5
86	86	50.5	86	40.5
87	87	50.5	87	40.5
88	88	50.5	88	40.5
89	89	50.5	89	40.5
90	90	50.5	90	40.5

module is realized by a UART communication module. Either linear stimulus protocol or nonlinear stimulus protocol as discussed in previous chapter (subsection 2.3.3) can be used. In TEOAE audiometry linear stimulus protocol can be used to detect TEOAE, if there is a proper fit of the probe with the ear canal. The stimulus artifact dies out within 1 ms after the stimulus onset for proper fit. In this protocol, the response waveform of selected intensity level is called as waveform-1 and is stored in the microcontroller memory. The waveform is delivered whenever a positive pulse is detected and inverted waveform is delivered whenever a negative pulse is detected. The nonlinear stimulus protocol should be used whenever the oscillation of ear canal artifacts extends more than 1 ms. Two waveforms are stored in the microcontroller memory. The first corresponds to smaller intensity of the first three positive stimulus pulses and second waveform corresponds to the larger intensity of the fourth negative stimulus pulse. The first waveform is delivered whenever a positive pressure pulse is detected and second waveform is delivered whenever a negative pulse is detected. The level of TEOAE in the neonatal ears is 10 dB greater than in the adult ears for same stimulus intensity. The number and frequency of dominant components vary in different ears and hence a provision is given to set it using GUI. Ear having high-frequency hearing loss at frequencies greater than 2 kHz may not have any dominant component at frequencies greater

Table 3.3: Gain of right and left channels of PGA for different stimulus levels.

Stimulus level (dB SPL)	Gain of right channel (dB)	Gain of left channel (dB)
70	-21	-66.0
71	-20	-65.5
72	-19	-65.0
73	-18	-64.5
74	-17	-64.0
75	-16	-63.5
76	-15	-63.0
77	-14	-62.5
78	-13	-62.0
79	-12	-61.5
80	-11	-61.0
81	-10	-60.5
82	-9	-60.0
83	-8	-59.5
84	-7	-59.0
85	-6	-58.5
86	-5	-58.0
87	-4	-57.5
88	-3	-57.0
89	-2	-56.5
90	-1	-56.0

than 2 kHz. Influence of noise, is unavoidable such as low-frequency subject generated noise and high-frequency instrumental noise, during clinical testing. The type of noise and its level can be set using GUI. Variation of delivered stimulus intensity at ear canal is also simulated, within ± 4 dB SPL. A large variation in stimulus intensity corresponds to improper fit of OAE probe with the ear.

Chapter 4
SIMULATOR HARDWARE AND SOFTWARE

4.1 Introduction

This chapter presents hardware and software design of the TEOAE simulator. Its block diagram has been shown earlier in Figure 3.7. The system consists of a microcontroller with on-chip ADC and DAC, a level shifter for connecting stimulus to ADC of the microcontroller, two differential amplifiers connected to the two differential DAC channels of the microcontroller, a two-channel programmable gain amplifier, a summing amplifier for outputting the response, and a serial interface for PC-based GUI for setting the parameters. ADC is used to monitor the input and detect the pulsed stimulus. One of the DAC channels is used to deliver TEOAE response and the other is used to deliver either stimulus artifact or noise. The amplitude level of TEOAE, stimulus artifact, and noise is achieved by varying the PGA gain. The outputs from two PGA channels are added using a summer amplifier and delivered as simulator output or ear response. A description of the circuit blocks and power supply is given in the following section. This is followed by a description of the software.

4.2 Microcontroller

The core of the simulator circuit is the microcontroller "Microchip dsPIC33FJ128GP802" [6]. The important features are as the following: on-chip ADC with 12-bit resolution and maximum conversion rate of 1.1 M samples/second, audio DAC with two channels (left and right), 16-bit resolution and maximum sampling rate of 100 K samples/s., SPI module for serial communication and maximum clock frequency of 10 MHz., UART communication module with settable baud rate. The pin connections to the microcontroller U2 are shown in Figure 4.1. Power supply and inline programming connections are also shown. Decoupling capacitors C3 and C4 are provided with AVDD, AVSS, VDD, and VSS. A low-ESR capacitor of 10 µF (C5) is connected between VCAP and ground to stabilize the output voltage of internal voltage regulator. The connector CON2 is provided for inline programming and debugging, where PGD1, PGC1, and MCLR pins are used for in-circuit serial programming and debugging. The shifted stimulus is given to input pin (AN0) of ADC set for 12-bit sampling. The stimulus artifact is generated using DAC1 (left channel). Its differential outputs (DAC1LN and DAC1LP) are applied as inputs of a differential amplifier. The TEOAE response is generated using DAC1 right channel and the differential outputs (DAC1RN and DAC1RP) are applied as inputs of a differential amplifier. Two pins (RB4 and

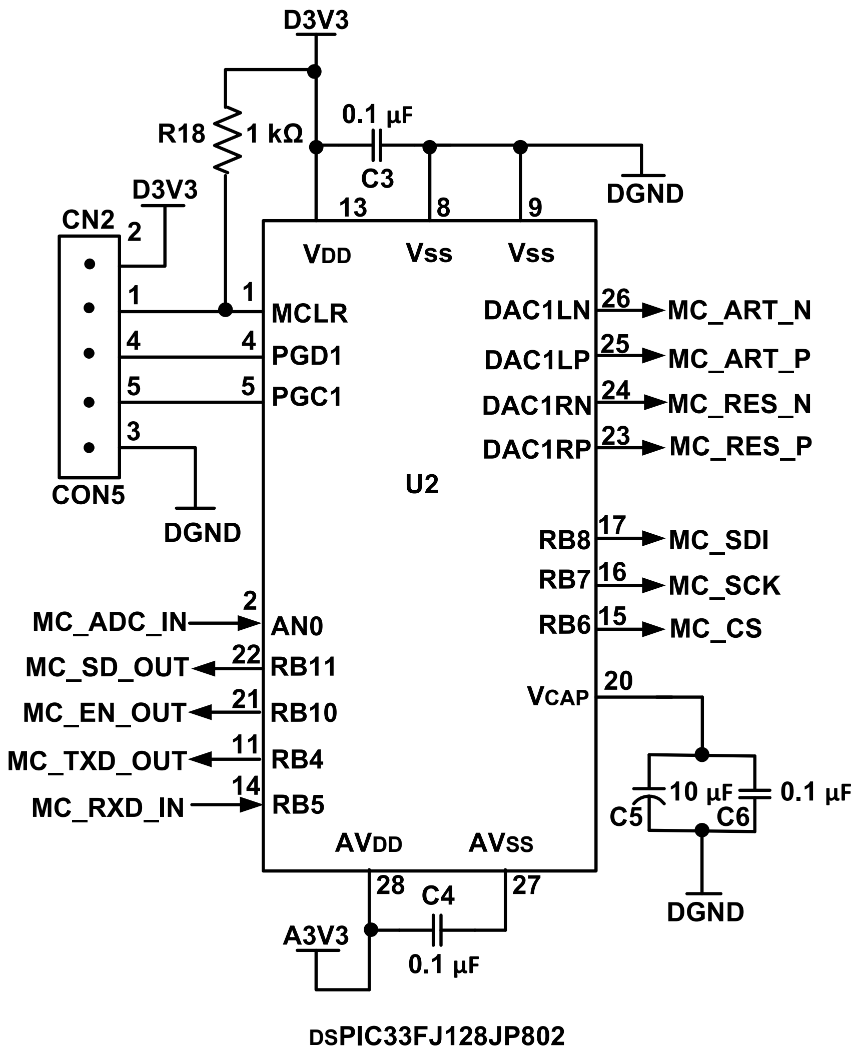

Figure 4.1 Microcontroller pin connections for the TEOAE simulator.

RB5) are used for UART serial communication and two pins (RB10 and RB11) are used to enable and shutdown the RS232 transceiver chip ADM3222 [8]. The gains of two-channel PGA are controlled by SPI, realized using 3 pins: serial data out SDO (RB8), serial clock out SCLK (RB7), and chip select CS (RB6).

4.3 Stimulus acquisition module

The stimulus input is a square pulse with a pulse width of 80 μs and zero offset voltage. The ADC input of the microcontroller U2 has a unipolar input range of $0 - 3.3$ V. The stimulus amplitude can have a maximum voltage of +1.65 and minimum voltage of -1.65 V. A pulse with amplitude greater than 120 mV and of either polarity is taken as stimulus. The stimulus input at connector-1 (CN1) is given to input pin of level shifter circuit. The level shifter shifts the stimulus input (CN1_SHIFT) by 1.65 V and the resulting output is given to the input pin (AN0) of ADC as shown in Figure 4.2. The level shifter circuit is realized using “Texas Instruments LME49710” audio operational amplifier as U1 and powered by ±5 V [5] as

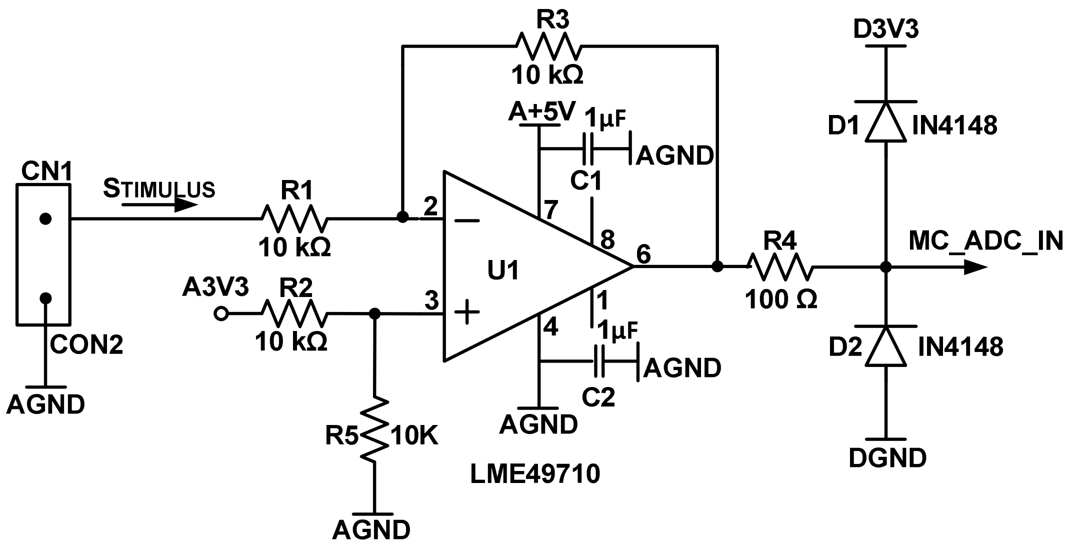

Figure 4.2: Level shifter circuit.

shown in Figure 4.2. A diode protection circuit (two IN4148 diodes) with a series resistor of 100 Ω (R4) is provided at the output of the level shifter. The ADC module works in signed mode, with 1.65 V taken as the baseline voltage with digital value of 0, 0 V input taken as −2048, and 3.3 V input taken as 2047. ADC continuously monitors the input signal and an input value greater than 150 and less than − 150 is recognized as occurrence of a stimulus pulse.

4.4 Response delivery module

This module consists of differential amplifiers, a two-channel programmable gain amplifier, and a summing amplifier or adder. The differential outputs of DACs of microcontroller U2 are given to differential amplifiers to remove bias voltage. Differential amplifiers are realized using “Texas Instruments LME49710” audio operational amplifiers as U3 and U4, as powered with ±5 V, and shown in Figure 4.3. The output U3_VINR delivers stimulus artifact and U4_VINL delivers TEOAE response with zero baseline voltage. Usually a level difference of 50 dB or more is present between stimulus artifact and TEOAE response. This level difference is achieved by varying the programmable gain. The TEOAE response from the output pin U4_VINL is given to left channel and stimulus artifact from the output pin U3_VINR is given to the right channel of a two-channel programmable gain amplifier. “Texas Instruments PGA2311" [7] as U5, the gain of each channel is independently controlled from −95.5 to +31.5 dB with steps 0.5 dB, by an external device with SPI communication module as shown in Figure 4.4. A voltage supply of ± 5 V is used to power the analog part and +5 V supply to power the digital part of U5. The gains are controlled by the microcontroller U2 using its on-chip SPI serial communication module. SPI module of the PGA chip U5 consists of four pins: active low chip select CS (pin 2), serial data input pin SDI

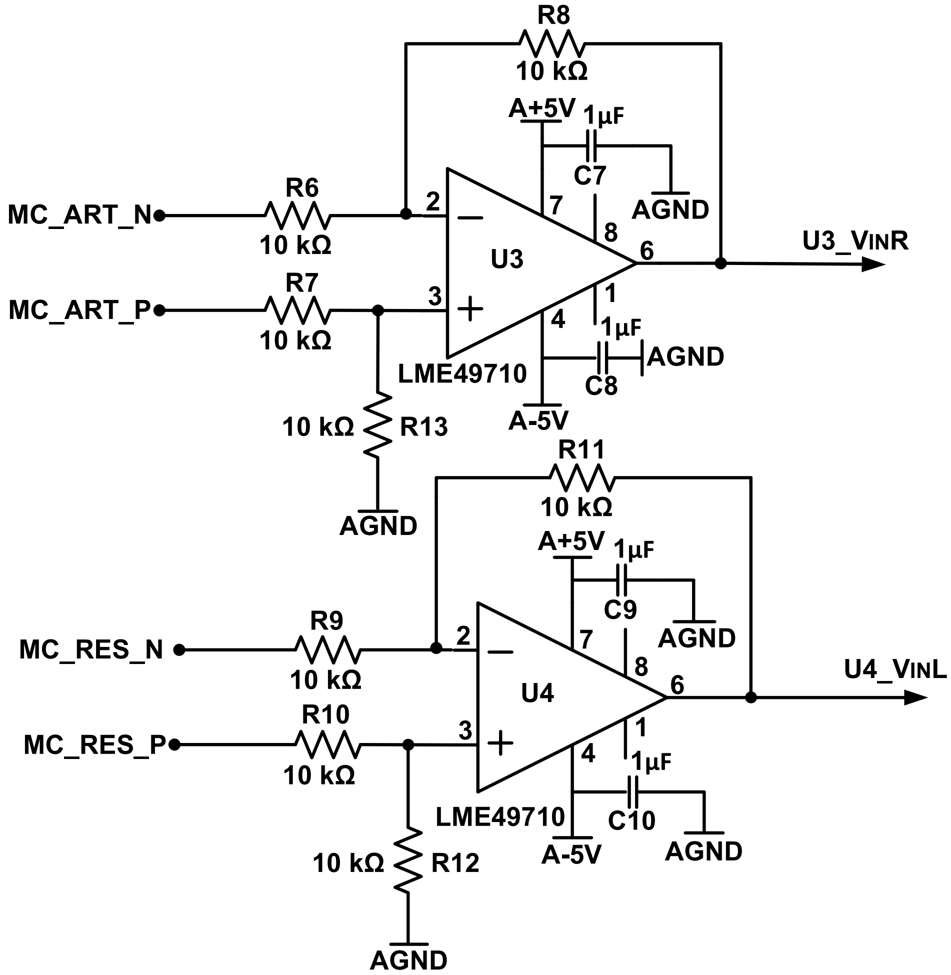

Figure 4.3: Differential amplifiers circuit for response generation.

(pin 3), and serial clock input SCLK (pin 6). These pins are connected to respective pins of SPI module of the microcontroller as shown in Figure 4.1. The control data is a 16-bit word, the first 8 bits are used to configure the gain of the right channel and the remaining 8 bits that of the left channel as shown in Figure 4.5. R7 is the MSB of the right channel gain byte and L7 is the MSB of the left channel gain byte. Decimal equivalent of gain byte is represented by N, and it varies from 0 to 255. The gain in dB is given by, $31.5 - (0.5(255 - N))$. This results in a gain of +31.5 dB at N = 255 and −91.5 dB at N = 1. At N = 0, PGA goes to the mute condition where the internal buffer amplifier is disconnected from output pin. Both the channels can be simultaneously muted by pulling down MUTE pin. In our application, this function is disabled by connecting the pin to +5 V. The output of both channels is summed by a summing amplifier realized by using "Texas Instruments LME49710" audio operational amplifier as U6. Its output can drive 600 Ω load. The circuit is shown in Figure 4.6. The simulator ear response is available at connector CN3 as shown in Figure 4.6.

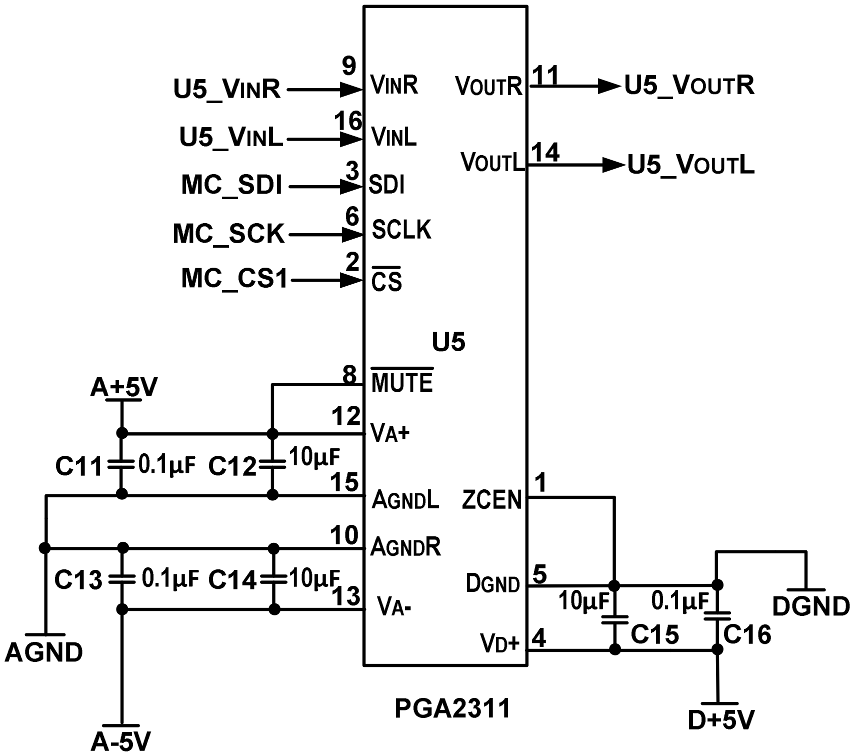

Figure 4.4 Programmable gain amplifier using PGA2311 for response generation.

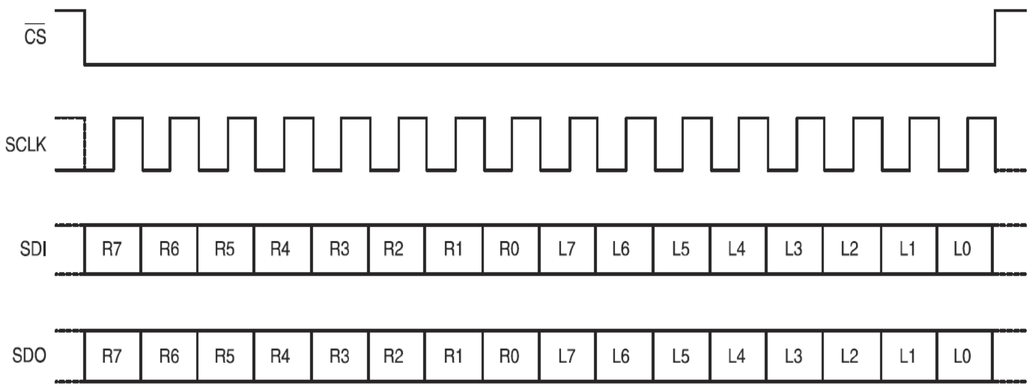

Figure 4.5: Serial interface protocol of PGA2311 [7].

4.5 Serial interface

Serial communication between PC and the simulator is realized using RS232 transceiver "Analog Devices ADM3222" as the buffer U7 as shown in Figure 4.7. It works with a single 3.3 V and provides level shifting between ±12 V level of RS232 and 0/3.3 V levels of microcontroller. Four external capacitors of 0.1 µF are used for voltage doubler/inverter [8]. The chip has two additional pins: EN is low-logic for tri-stating the receiver output, SD is a low-logic pin to power down the charger pump and transmitter output thus reducing quiescent current to 0.5 µA. The enable pin (EN) is connected to RB10 of U2 and shutdown pin (SD) is connected to RB11 of U2 as shown in Figure 4.1 and Figure 4.7. Pin RX (RB5) and pin TX (RB4) of U2 UART module are connected to R1OUT and T1IN of U7, respectively. The RXD,

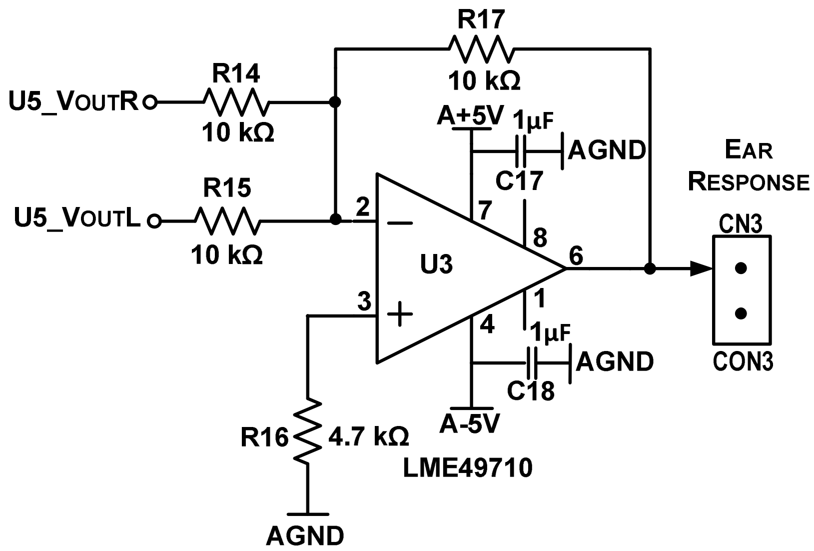

Figure 4.6: Summer amplifier.

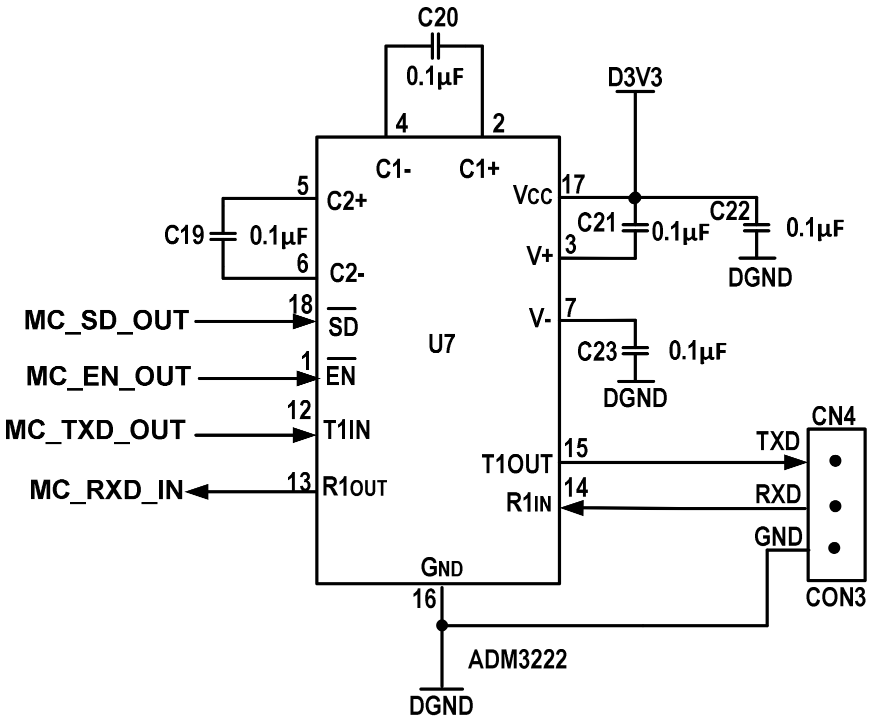

Figure 4.7: ADM322 pin connection for serial communication.

TXD and GND are available on CN2 as shown in Figure 4.7 for RS232 communication to PC.

4.6 Power supply

The simulator is designed to work with ± 5 V and 3.3 V supplies. The microcontroller U2 (Figure 4.1) works with analog supply of 3.3 V labeled as A3V3 and digital supply of 3.3 V labeled as D3V3. The serial transceivers U7 (Figure 4.7) is powered with D3V3. Analog part of the PGA chip U5 (Figure 4.4) and operational amplifiers U1 (Figure 4.2), U3, U4 (Figure 4.3), and U6 (Figure 4.6) work with analog +5 V labeled as A+5V and -5 V labeled as A-5V.

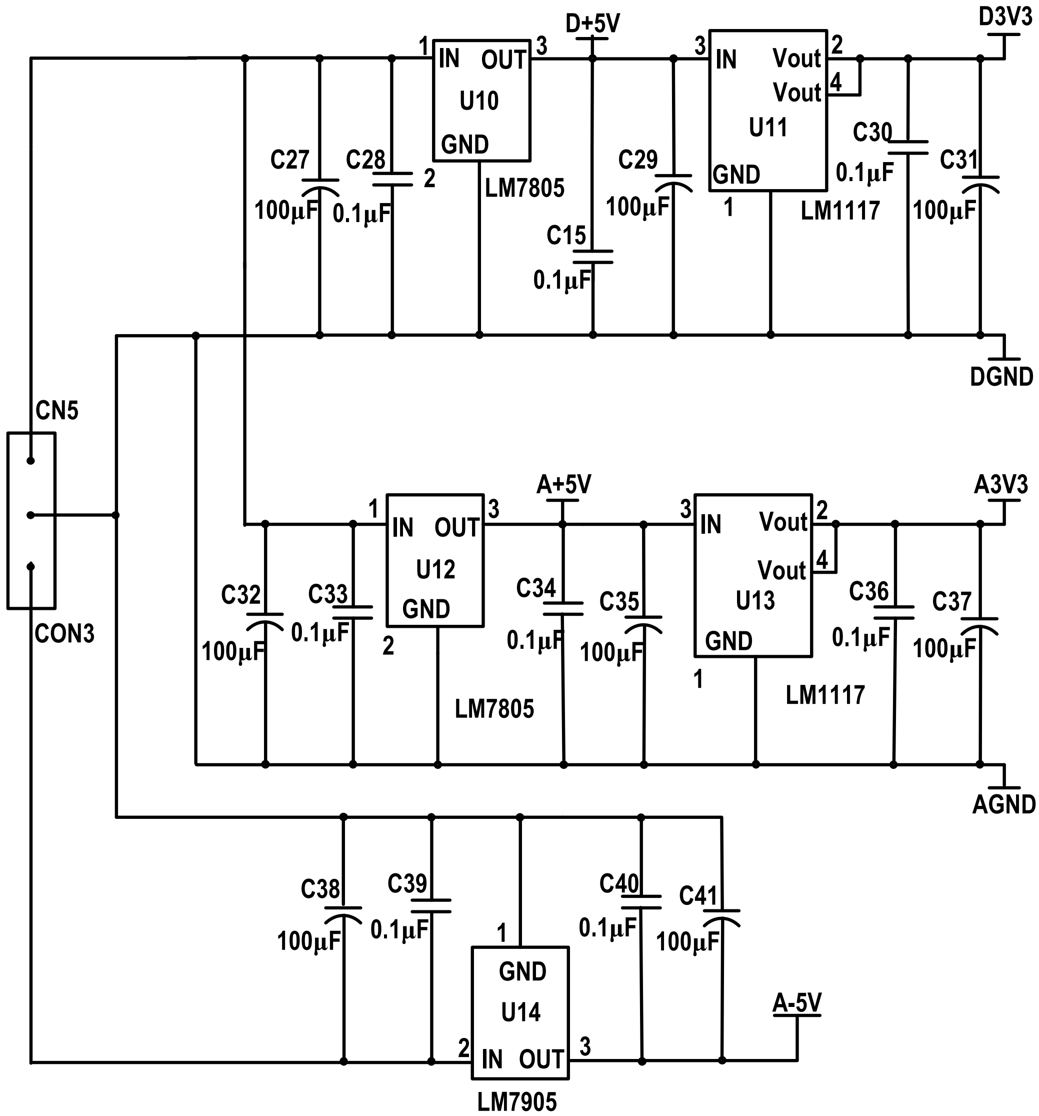

Figure 4.8: Power supply circuit.

The digital part of the chip U5 (Figure 4.4) is powered with +5 V labeled as D+5V. The estimated current loads on A+5V, A-5V, D+5V, A3V3, and D3V3 are 20, 22, 1, 26, and 5 mA, respectively, as listed in Table 4.1. The whole circuit has been tested using +5 V, -5 V, and +3.3 V supplies, with current loads of 21 mA, 22 mA, and 31 mA, respectively, obtained from three adjustable laboratory power supplies. A power supply circuit for obtaining power from +9 V and – 9V supplies is shown in Figure 4.8. Linear regulator "TI LM1117" is used as U11 and U13 is used for obtaining regulated 3.3 V. "TI LM7905" is used as U14 for A-5 V and "TI LM7805" is used as U12 for A+5 V. Another "LM1117" is used as U11 for D3V3.

4.7 Software module

Software is developed to control the simulated ear response to a transient square pulse stimulus. It consists of two programs: (i) a PC-based GUI program to set parameters of TEOAE and communicate with TEOAE simulator hardware and (ii) a microcontroller

Table 4.1: Estimation of current consumption by simulator hardware.

Supply voltage	Block/Component	Current (mA)
D3V3	U2 (DSPIC33FJ128GP802)	24.0
	U7 (ADM3222)	2.0
	Total	26.0
A3V3	U1 (DSPIC33FJ128GP802)	5.0
	Total	5.0
A+5V	U1 (LME49710)	2.5
	U3 (LME49710)	2.5
	U4 (LME49710)	2.5
	U5 (PGA2311)	10.0
	U6 (LME49710)	2.5
	Total	20.0
A−5V	U1 (LME49710)	2.5
	U3 (LME49710)	2.5
	U4 (LME49710)	2.5
	U5 (PGA2311)	12.0
	U6 (LME49710)	2.5
	Total	22.0
D+5V	U5 (PGA2311)	1.0
	Total	1.0

program to deliver the ear response. The parameters settable through GUI include stimulus type, stimulus intensity, type of ear, dominant frequency, type of noise, noise level, and variation of coupling at ear canal. The microcontroller program function includes communication with PC, sampling of the input stimulus pulse, detection of stimulus pulse, and its polarity, setting the gains of two-channel PGA, delivering either stimulus artifact or noise by DAC1 left channel and TEOAE response through DAC1 right channel.

4.8 Graphical user interface

A programme with graphical user interface (GUI) is developed using MATLAB R2009b as the programming tool. It interacts with the hardware by serial communication protocol. Its main function is to set different parameters of the simulator: stimulus type, stimulus intensity, ear type, coupling of ear canal with probe, type of noise, noise level, and response waveform. Pressing "Synthesize" button synthesizes the TEOAE response with dominant frequencies selected using the radio button. In linear stimulus protocol mode, one waveform is generated as “waveform-1” with respect to selected stimulus intensity and dominant frequency components. In case of nonlinear stimulus protocol, two waveforms are generated as “waveform-1” and “waveform-2”. The second waveform is generated for the negative stimulus with the selected stimulus intensity and the first waveform is generated based on the stimulus intensity which is 10 dB lesser than the selected intensity. These two waveforms are used to deliver response for nonlinear stimulus application. The synthesized TEOAE response and its spectrum are displayed in GUI display panel. An example of GUI panel is shown in Figure 4.9. Dominant frequency components are present at 1.10 kHz, 1.50 kHz, and 2.90 kHz.

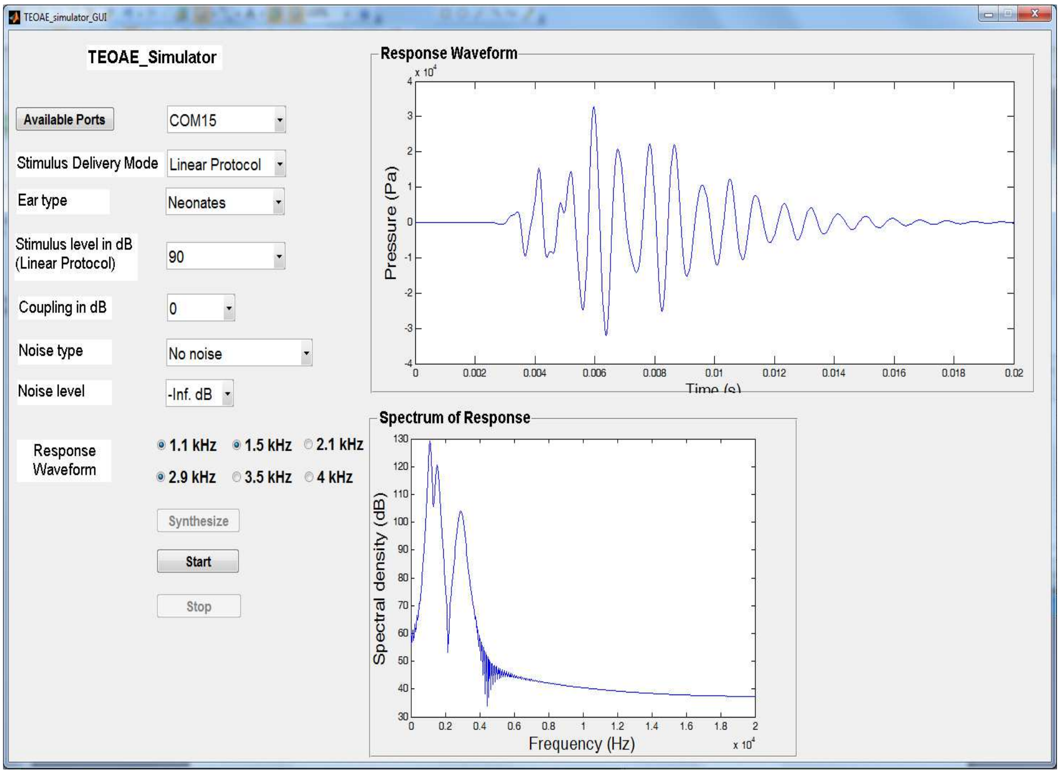

Figure 4.9: GUI of TEOAE simulator.

The response waveform starts approximately 3.5 ms after the stimulus onset and ends at 20 ms. A total of 400 samples for the duration of 20 ms with sampling frequency of 20000 Hz is generated for waveform and sent to the microcontroller of the simulator for linear stimulus protocol. In nonlinear stimulus method, 400 samples of waveform-1 and waveform-2 are stored in the microcontroller with the same sampling frequency. A flowchart representation of the GUI program is shown in Figure 4.10. "Start" button is enabled after synthesis and display of the waveform. Pressing "Start" button leads to the transfer of parameter values along with waveform samples to the microcontroller. "Stop" button is enabled after of the transfer of these values. Pressing it stops the simulator from responding to the stimulus input.

4.9 Microcontroller program

The microcontroller program is written in C. It consists of UART interrupt service routine program which stores the samples, and set the gains of two-channel PGA, stimulus detection program which monitors the input and detects the square pulse stimulus, ear response delivery program delivers the artifact and TEOAE response. The main program starts with setting of system clock, initialization of input output ports, ADC module, UART module, DAC module and SPI module as shown in Figure 4.11. Start and stop function of the device is controlled using GUI. The microcontroller works with instruction cycle clock F_{CY} of 37 MHz, achieved by configuring FRC oscillator with PLL. Built-in ADC module is configured in 12-bit resolution, signed mode of operation and timer-triggered start of conversion. The total ADC

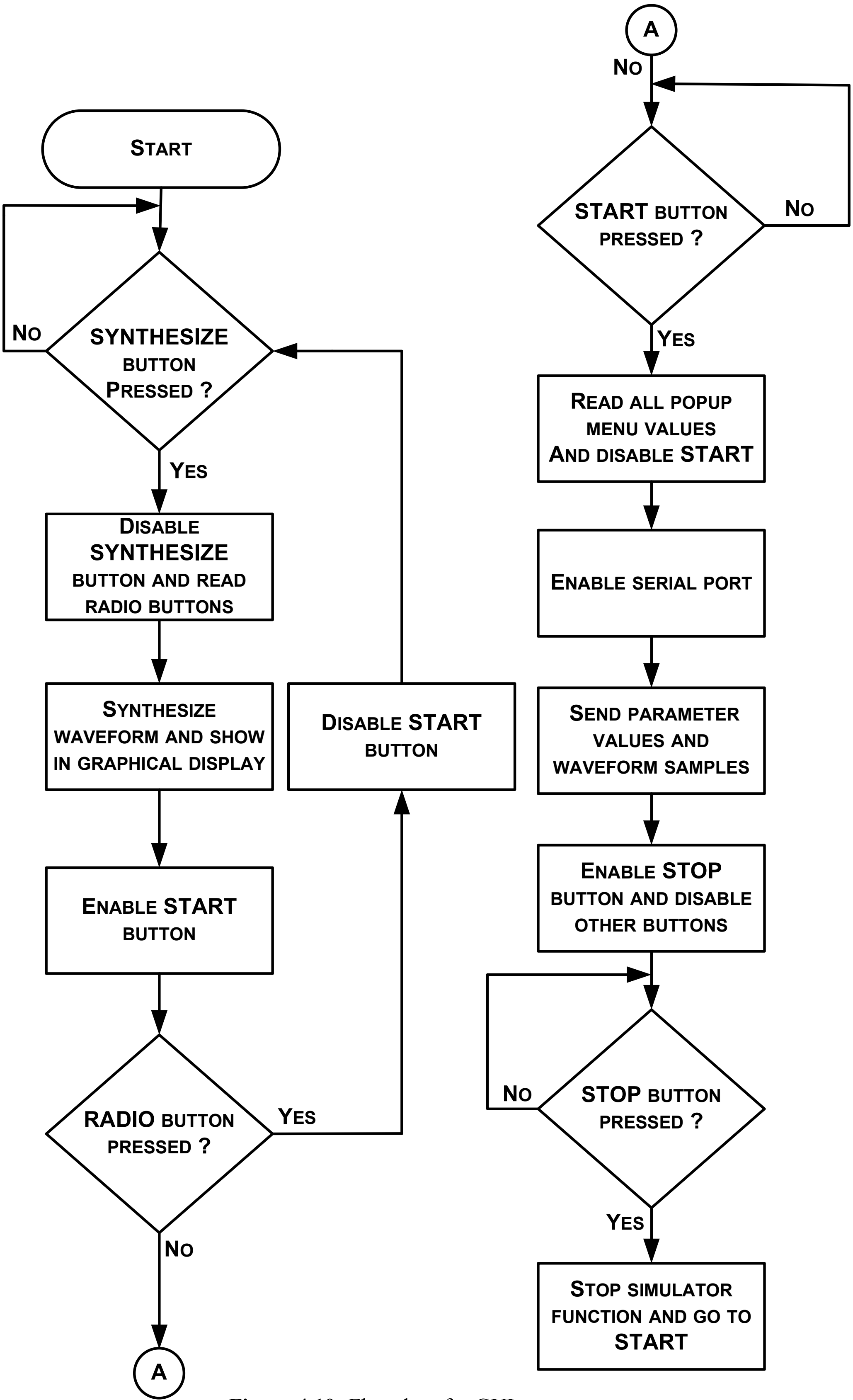

Figure 4.10: Flowchart for GUI program.

conversion time is 1.89 µs for F_{CY} of 37 MHz. Built-in audio DAC1 consists of two channels, right and left. The DAC1 is configured in signed mode of operation. A default value of

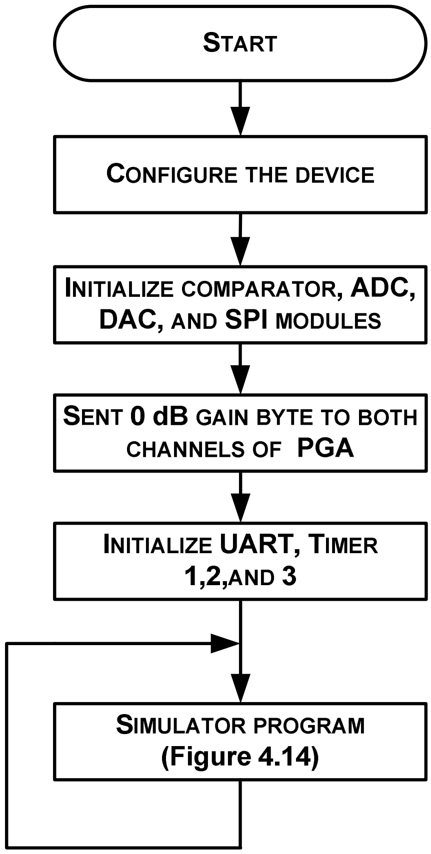

Figure 4.11: Flow chart for initialization of microcontroller.

0x0000 H is assigned to DAC output registers, which sets the negative and positive terminal outputs of both channels at 1.65 V. The SPI module is configured to send the gain word of 16 bits to U5 with a clock frequency of 6 MHz. The microcontroller sends mute word to U5 after initialization of SPI module. The UART module is configured in flow control mode with a baud rate of 9600 bits/s.

4.9.1 UART module

PC-based GUI sends either start command with parameters and waveform samples or stop command. Stop command stops the simulator module from responding to the stimulus. After detecting start command at the interrupt subroutine of UART receiver module, the service routine stores the waveform samples in the data RAM of the microcontroller, receives ear type, stimulus level, noise and its level as shown in Figure 4.12 and Figure 4.13. As mentioned earlier (Section 3.3), one waveform (waveform-1) is transferred from GUI to the microcontroller for the linear stimulus protocol. For nonlinear stimulus protocol, two waveforms (waveform-1 and waveform-2) are transferred to the microcontroller. The ear type can be either neonatal or adult. The level difference between neonatal response and adult response is 10 dB for the same stimulus intensity as shown in Table 3.2. The stimulus level value is used to set the gain of both channels of PGA. The gain values of both channels as a

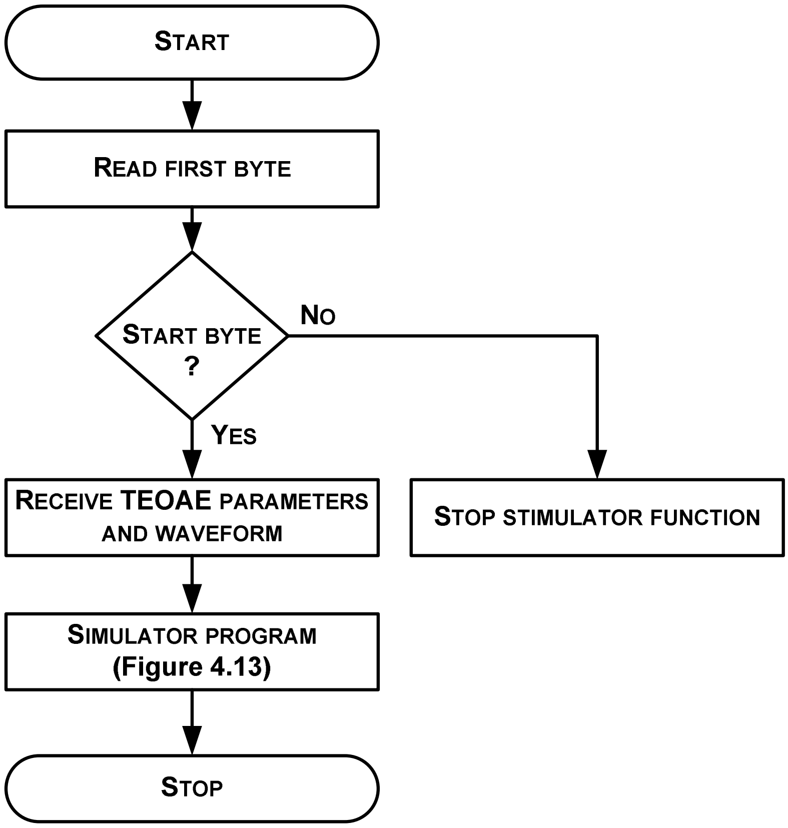

Figure 4.12: Flow chart for interrupt service routine of UART: first of two parts.

function of the stimulus level for an adult ear are given in Table 3.3. The coupling level in dB is added to the given gain of U5. The gain byte for neonatal ear is also stored in a look-up table. Noise can be either periodic sine wave or random noise, and its delivery is described later. Noise is delivered with the help of timer 1. The samples of sine wave as the periodic noise are stored in the data memory. The waveform is delivered with a time period determined from the set frequency. The count of timer 1 determines the time interval between the successive samples. If the noise is selected as random noise, the noise is generated sample-by-sample and output in timer 1 interrupt subroutine, as described later in Subsection 4.9.4.

4.9.2 Stimulus detection

The flow chart shown in Figure 4.13 represents delivery of ear response after the detection of stimulus at the input. A square pulse with magnitude greater than 120 mV is detected by monitoring the sampled input values. Presence of stimulus is confirmed if 8 consecutive input samples have absolute digital value greater than 150 (i.e. ±120 mV). The ADC module is switched off after the detection of stimulus, until the next stimulus detection cycle.

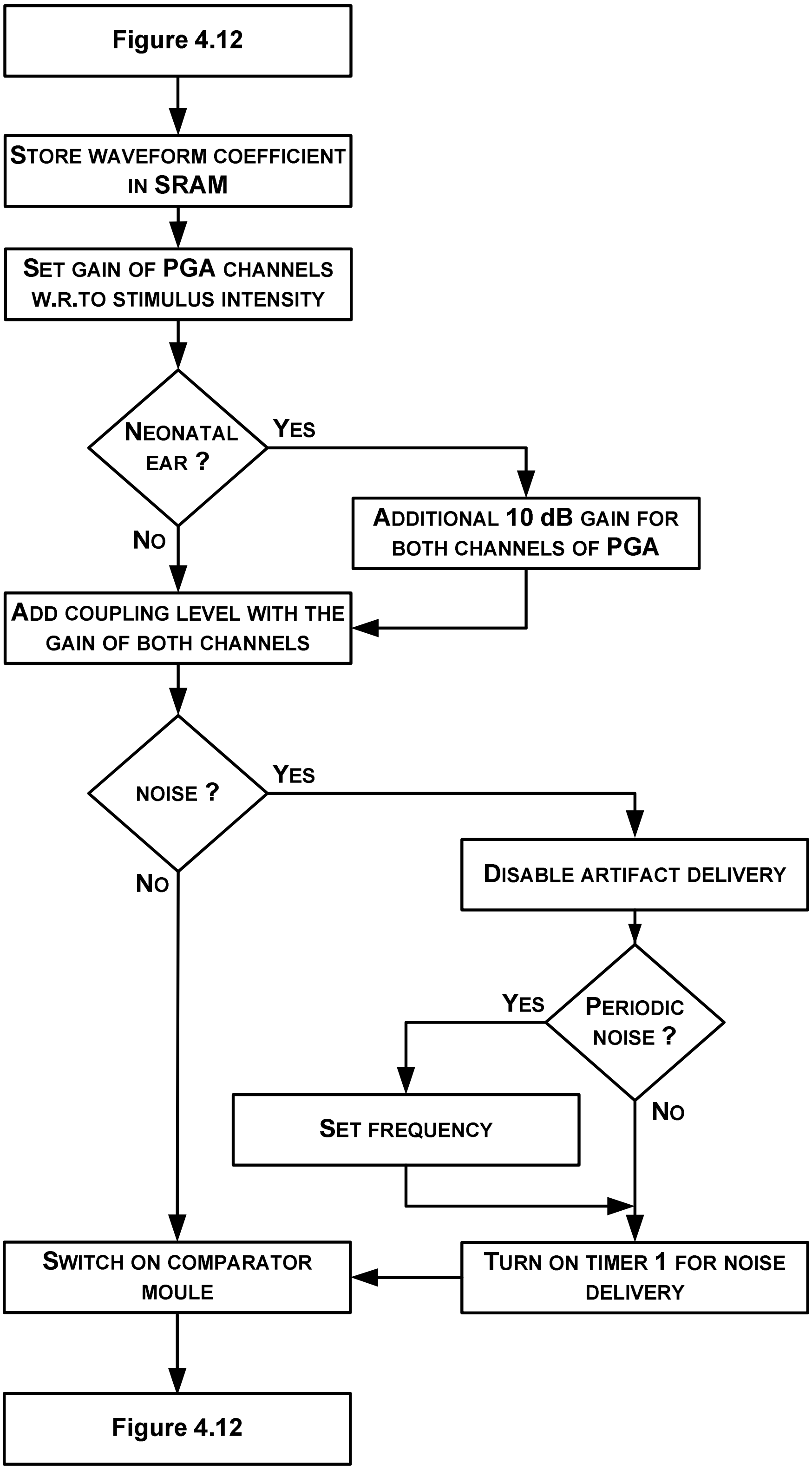

Figure 4.13: Flow chart for interrupt service routine of UART:second of two parts.

4.9.3 Ear response generation

An averaging of 8 sampled values is used to find the polarity of stimulus. Waveform-1 is delivered whenever a positive pulse is detected and inverted waveform-1 is delivered

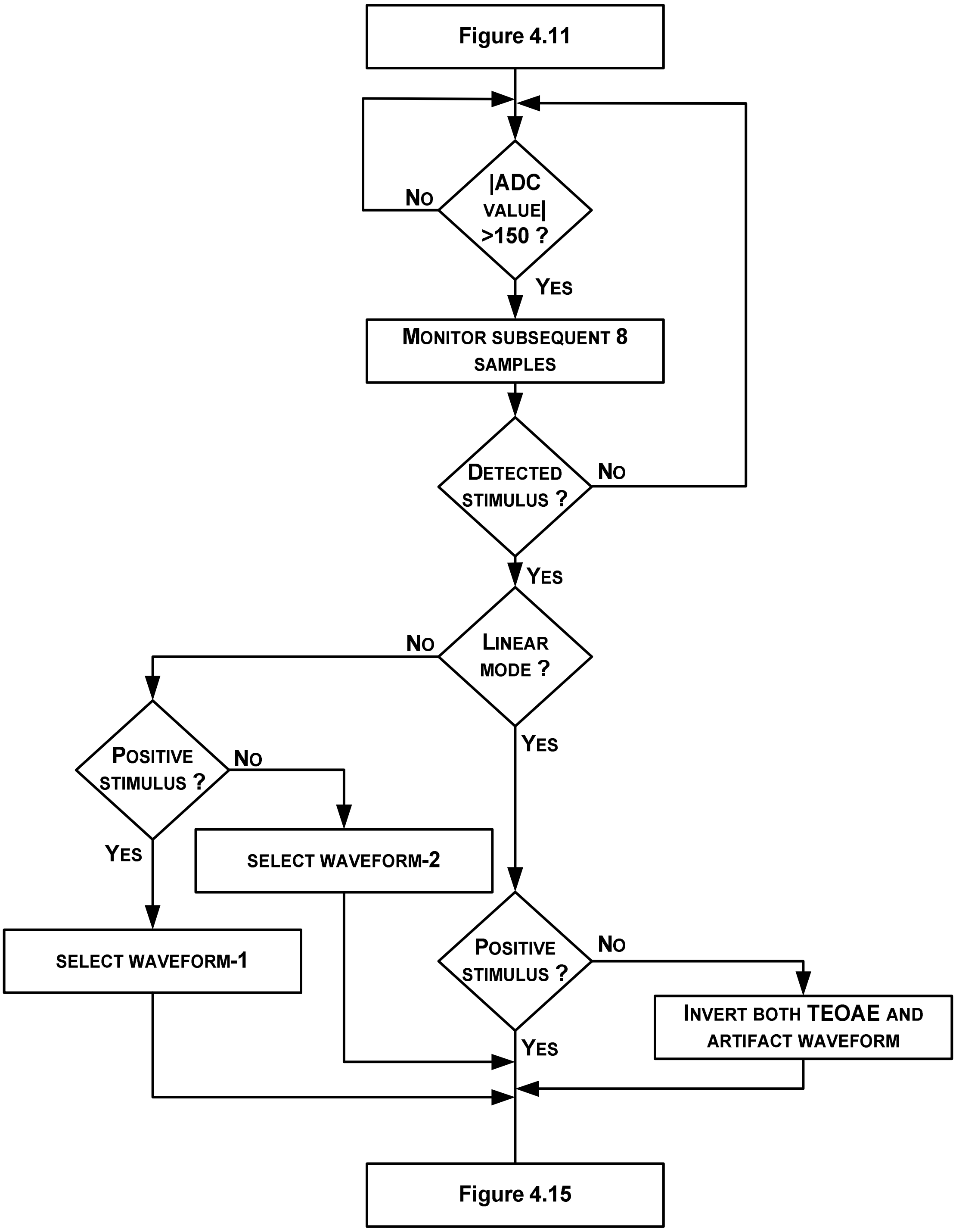

Figure 4.14: Flowchart for simulator program: first of two parts.

whenever a negative pulse is detected in linear stimulus protocol. In case of nonlinear stimulus protocol, the waveform-1 is delivered for positive pulse detection and waveform-2 is delivered for negative pulse detection. Respective gain byte of each channel of U5 for the selected stimulus intensity is calculated and set at UART interrupt subroutine. The coupling is also considered on calculation. The resultant gain bytes of both channels are loaded to U5 by

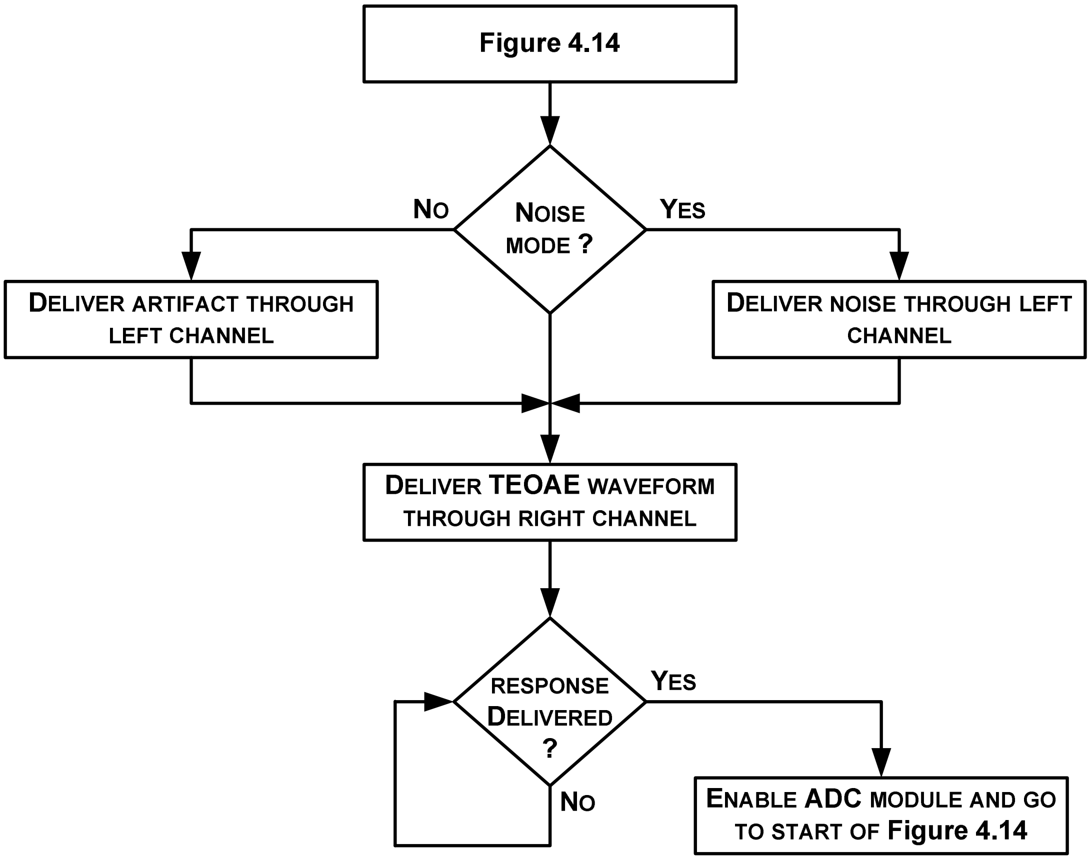

Figure 4.15: Flowchart for simulator program: second of two parts.

SPI communication protocol. If noise mode is off, the stimulus artifact coefficients and TEOAE response samples are output through respective DAC channels with sampling frequency of 20 kHz. It takes 20 ms to deliver the ear response. After delivery of the response for monitoring the next stimulus pulse, the ADC module is switched on as shown in Figure 4.15.

4.9.4 Noise generation

Noise mode is used to add noise to the TEOAE response, type of noise is selected using GUI. Types of noise include sine wave of 50 Hz, 250 Hz and 500 Hz, and random noise. The level of noise can be set in dB. If the noise option is selected, stimulus artifact delivery is disabled. The delivery of noise is implemented in interrupt service routine of timer 1. The sample values of a sine wave cycle are stored in data memory. The values are delivered with a sampling rate which is calculated from the set frequency and is used to set the count of timer 1. The random noise is generated by a pseudorandom number generator using a 16-bit shift register with feedback and implemented as part of interrupt subroutine of timer 1.

Chapter 5
A TEOAE AUDIOMETER FOR TESTING THE SIMULATOR

5.1 Introduction

Electrical part of a TEOAE based audiometer is designed to test and verify the simulator described in the previous chapter. The main aim of the device is to deliver transient click stimulus with variable sound intensity levels, acquire ear response, and send to PC for displaying and processing.

5.2 TEOAE audiometer hardware

The instrument is designed using a microcontroller based system with a PC-based GUI. The block diagram of audiometer is shown in Figure 5.1. It consists of stimulus delivering module, response acquisition module, and communication module. Stimulus is generated using DAC. The stimulus level is controlled by a programmable gain amplifier and a power amplifier is used to drive the speaker of the probe. Since testing the simulator can be carried out using the electrical signal itself, the power amplifier circuit after the programmable gain amplifier (PGA) is not included to test the simulator and PGA output is directly given as input to the simulator. The response output from the simulator is acquired by the acquisition module. It includes a low-pass filter, two PGA amplifiers, a level shifter, and an ADC. The acquired response is sent to PC for display and processing to detect the presence of OAE components.

5.2.1 Stimulus generation module

This module consists of a microcontroller DAC for generating stimulus, a differential amplifier, and a programmable gain amplifier. The microcontroller used here is "Microchip dsPIC33fJ128GP802" as U4 and shown in Figure 5.2. It consists of two audio DAC channels for stereo application [6]. Left audio DAC channel is assigned for the generation of stimulus. Each DAC channel has a positive and a negative output pin and both pins are biased at 1.65 V for signed mode of operation [6]. The maximum output voltage of DAC is 2.25 V and minimum is 1.05 V. The positive and negative outputs of DAC1 left channel are given to a differential amplifier inputs for removing 1.65 V DC offset and centers the waveform at 0 V, as shown in Figure 5.3. The differential amplifier is realized using audio operational amplifier “Texas Instruments LME49710” [5] as U1. The output of the differential amplifier is given to the input pin LOW_O of a unity gain low-pass filter. Audio DAC channel with differential amplifier generates square pulse of constant amplitude and pulse width of 80 µs. The stimulus

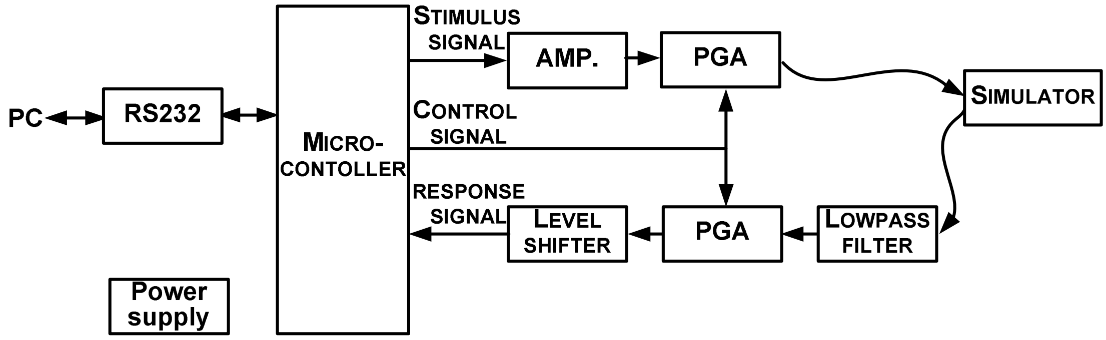

Figure 5.1: Block diagram of TEOAE audiometer.

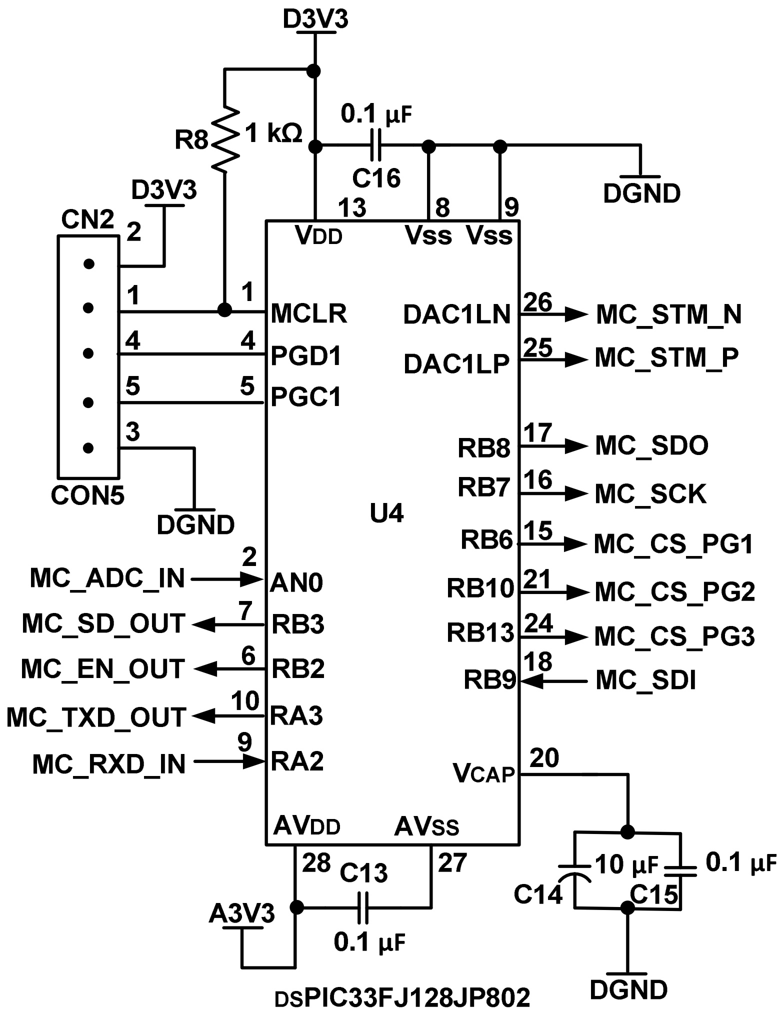

Figure 5.2: Microcontroller pin connections for the audiometer.

signal is low-pass filtered with cutoff frequency of 8 kHz. The active low-pass filter realized using “Texas Instruments LME49710” as U5 is shown in Figure 5.4. Its output pin LOW_O is connected to the right channel of programmable gain amplifier. The level of stimulus is determined by the gain of “Texas Instruments PGA2311” as U3. It is a two-channel programmable gain amplifier for stereo application [7]. Its gain can be varied from $-$ 95.5 to 31.5 dB in steps of 0.5 dB. The gain can be controlled by an external device using SPI

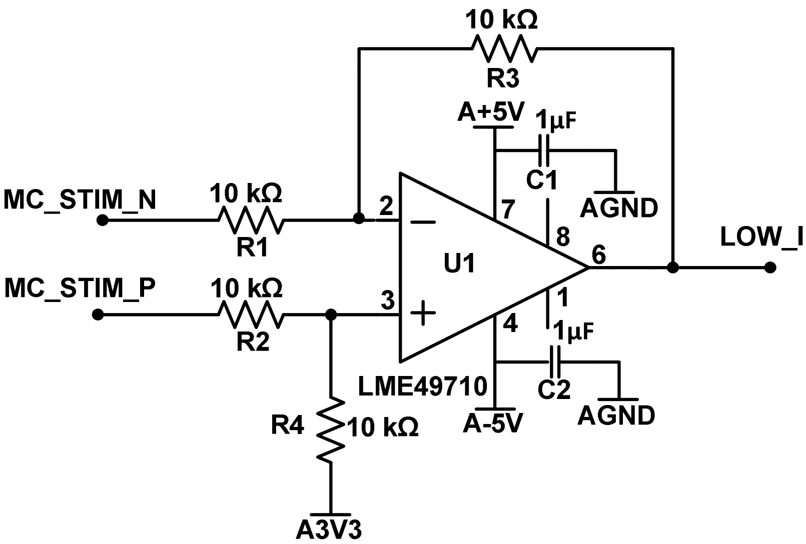

Figure 5.3: Differential amplifier for stimulus generation.

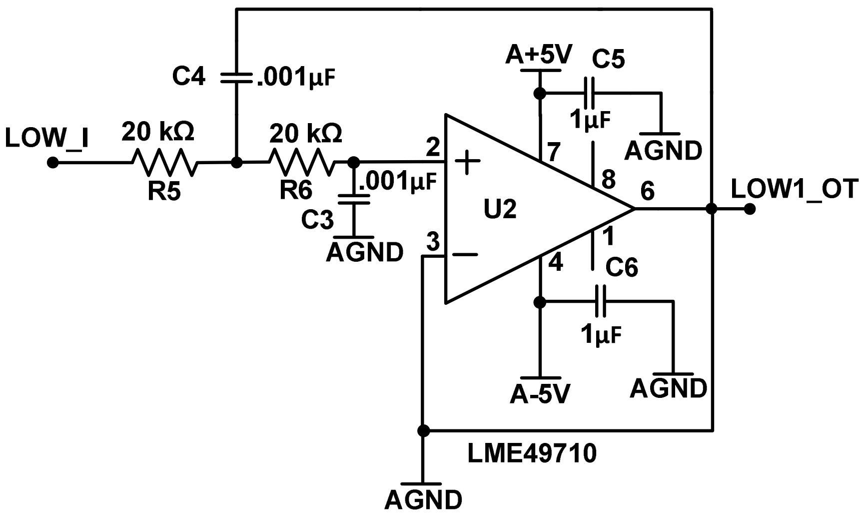

Figure 5.4: Low-pass filter circuit for stimulus generation.

communication module. This chip also used in simulator module. The pin diagram with the power supply connection of U3 is shown in Figure 5.5. The voltage supply of ± 5 V is used to power its analog part and +5 V supply to power its digital part. Its gain is controlled by microcontroller U4 using SPI. SPI module of U3 consists of four pins: active low chip select CS (pin 2), serial data input pin SDI (pin 3), serial clock input SCLK (pin 6), and serial data out SDO (pin 7). These pins are connected to respective pins of the microcontroller's SPI module as shown in Figure 5.2. The control data is a 16 bit-word, the first 8 bits of control word is used to configure the gain of the right channel and the remaining 8 bits for the left channel as shown earlier in Figure 4.5. R7 is the MSB of the right channel gain byte and L7 is the MSB of the left channel gain byte. Decimal equivalent of gain byte is represented by N, and it varies from 0 to 255. The gain in dB is $31.5 - (0.5(255 - N))$ [7]. This results in a gain of +31.5 dB at $N = 255$ and −91.5 dB at $N = 1$. At $N = 0$, PGA goes to the mute condition

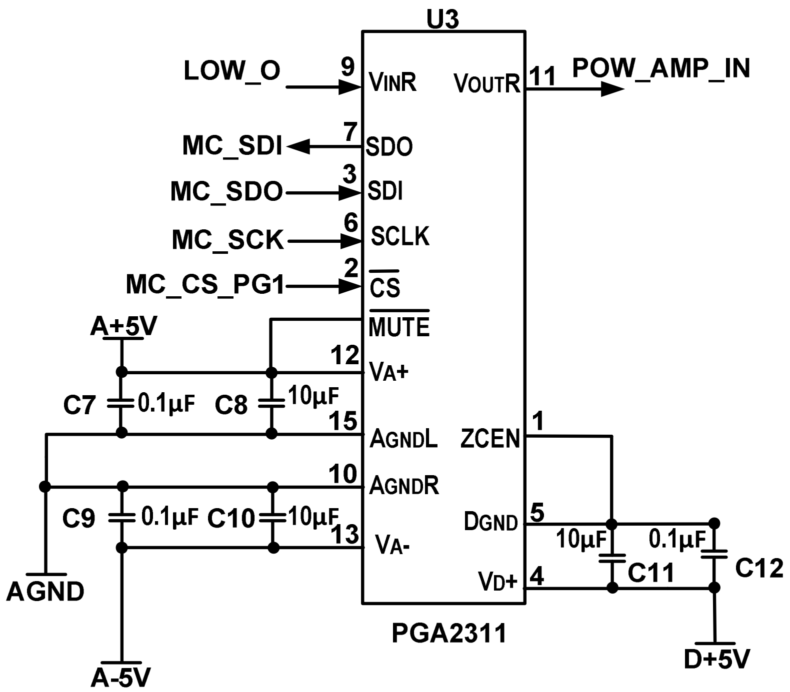

Figure 5.5: Programmable gain amplifier using PGA2311 for stimulus generation.

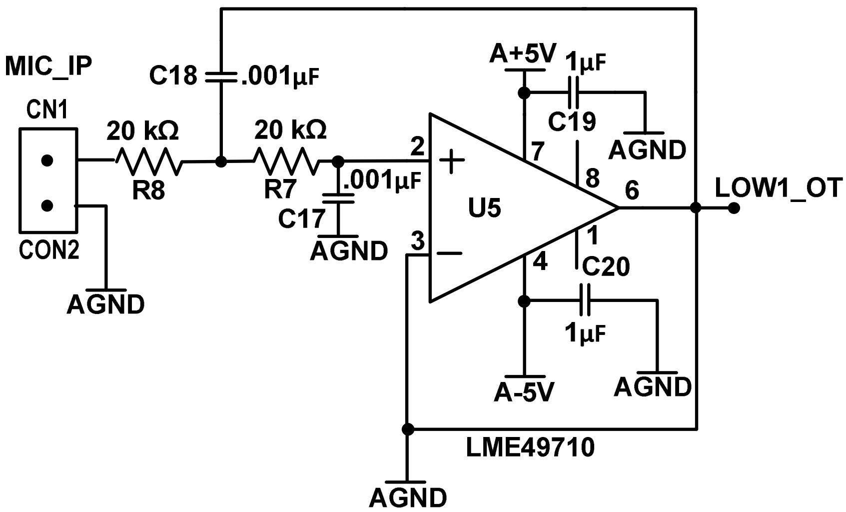

Figure 5.6: Low-pass filter circuit for response acquisition.

where the internal buffer amplifier is disconnected from the output pin. It is used to vary the click stimulus level from 70 to 95 dB in steps of 1 dB. The output from PGA is given as the input to simulator module.

5.2.2 Response acquisition module

This module consists of a low-pass filter, two preamplifiers, a differential amplifier and a microcontroller ADC. The acoustic ear response generated by the simulator is given to an active low-pass filter of with cutoff of 8 kHz, and realized using “Texas Instruments LME49710” operational amplifier as U5 and shown in Figure 5.6. The transfer function and design of low pass filter is described in Appendix A. The filtered signal is given to the input

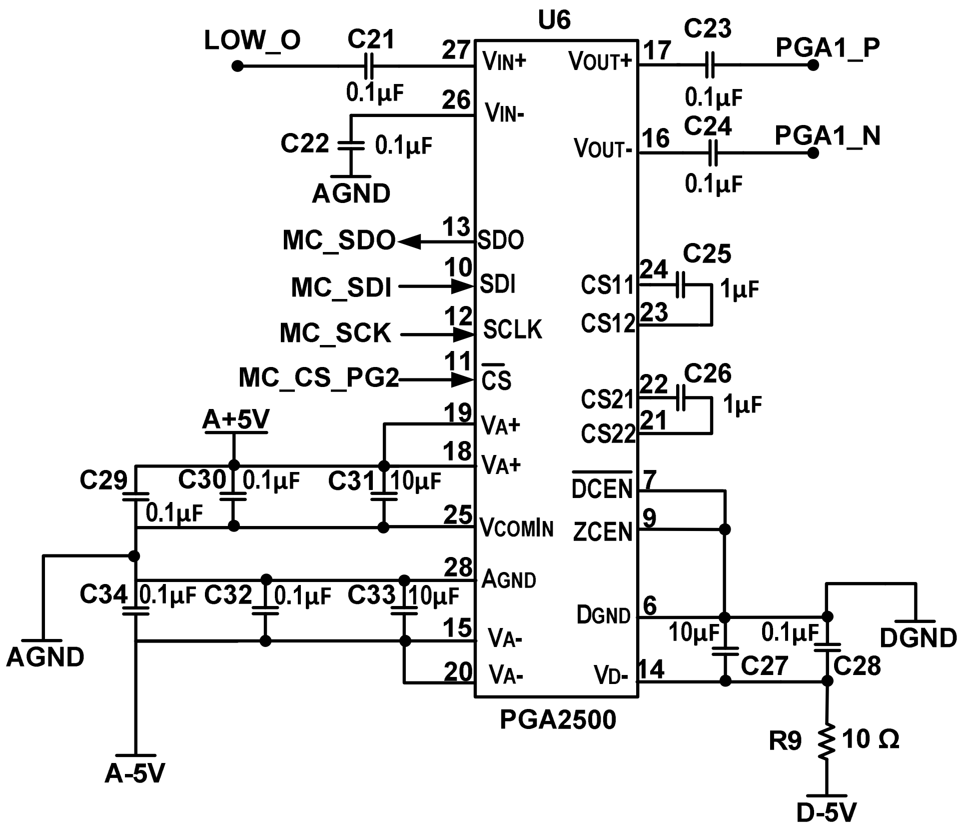

Figure 5.7: Programmable gain amplifier-1 using PGA2500 for response acquisition.

pin of a programmable gain amplifier. We have used "TI PGA2500", a digitally controlled microphone preamplifier with differential input and differential output architecture, as U7. Its gain can be controlled from 10 to 65 dB in steps of 1 dB by using SPI communication module. Its features include low noise, wide dynamic range, on-chip DC servo loop is employed to minimize DC offset, and a common mode servo function to enhance common mode rejection. The chip provides four general purpose digital output pins (CMOS logic output) for controlling external switches and devices. The pin diagram with power supply connection is shown in Figure 5.7. The voltage supply of ± 5 V is used to power the analog part and − 5 V supply to power the digital part of the chip. The gain is controlled by microcontroller U4 using SPI. SPI module of U6 consists of four pins: active low chip select CS (pin 11), serial data input pin SDI (pin 10), serial data output pin (13) and serial clock input SCLK (pin 12). These pins are connected to respective pins of microcontroller's SPI module as shown in Figure 5.2.

External capacitors C25 and C26 for U6 and C38 and C39 for U7 are required for the DC servo function as shown in Figure 5.7 and Figure 5.9. The DC servo function can be enabled either by software method or hardware method. In our application, this function is enabled by pulling down DCEN pin. Zero-crossing detection function and 0 dB or unity gain function are disabled by pulling down the ZCEN pin and 0 dB pin. A 16 bit-word is used as

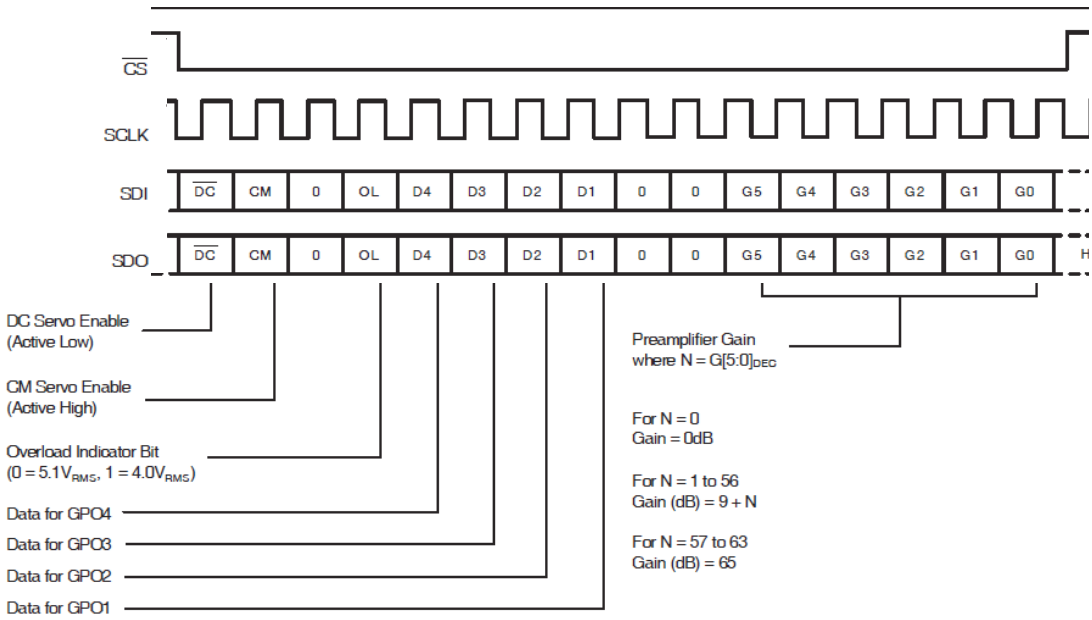

Figure 5.8: Serial interface protocol of PGA2500.

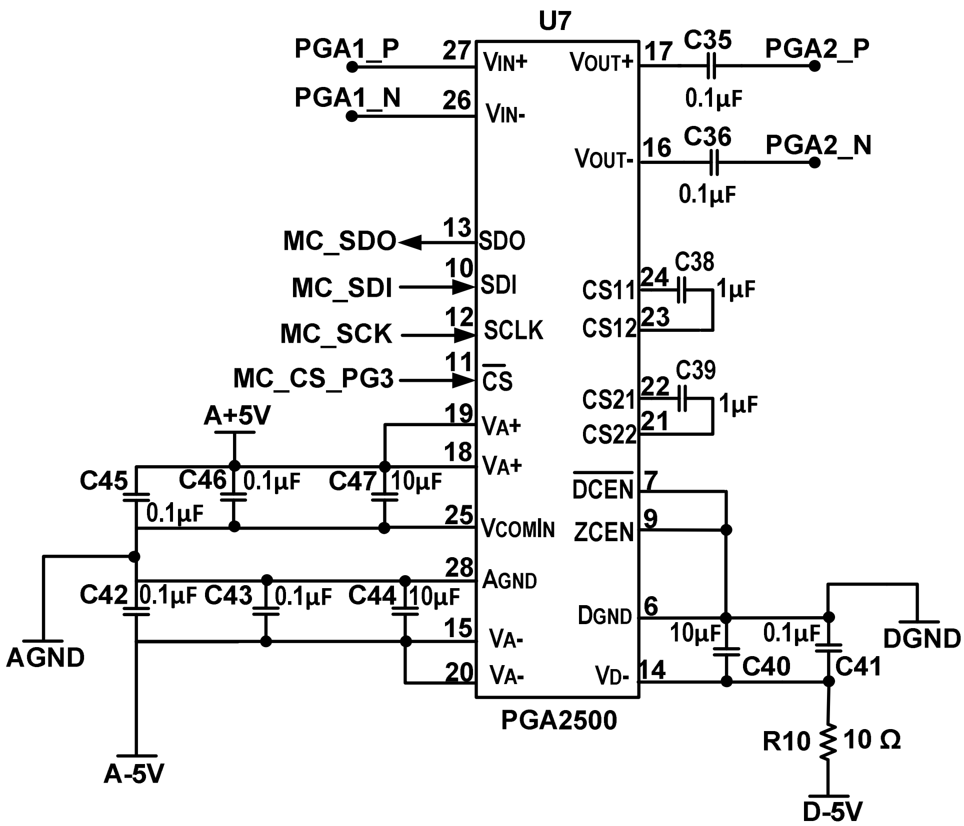

Figure 5.9: Programmable gain amplifier-2 using PGA2500 for response acquisition.

the control data, where the 16th bit (active low) is used to enable DC servo loop in software method and the 15th bit (active high) is used to enable common mode servo enable. The 14th bit is used to set the threshold of overflow indication function. It is not used in our application. The 13th to 10th bits are used to set the level of digital output pins. The 5th to 0th

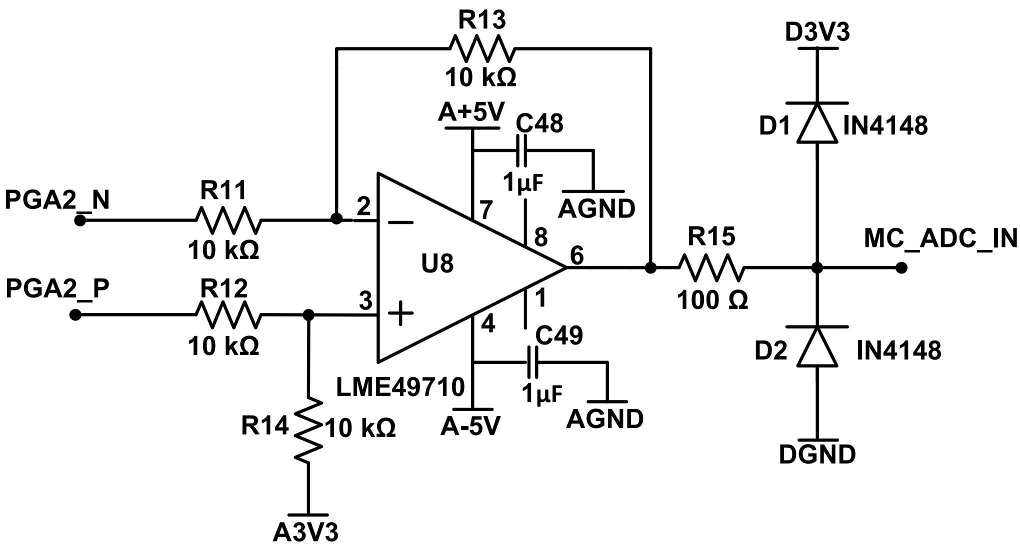

Figure 5.10: Differential amplifier.

bits are used to control the gain of preamplifier. The decimal equivalent of gain bits is represented by N and its value ranges from 0 to 64. For $N = 0$, the gain of the amplifier is 0 dB and for the values from 1 to 56, the gain of amplifier is given as $(9+N)$ dB. For the values of N from 57 to 64, the gain is equal to 65 dB as shown in Figure 5.8. High gain requirement is achieved by cascading two preamplifiers (U7 and U8). The low-pass filter output LOW_O is given to the positive input of U6 through a coupling capacitor C12. The negative input terminal of U6 is grounded through a capacitor C22 as shown in Figure 5.7. The input pins of U6 and U7 are internally biased at 0.65 V. The differential output pins PGA1_P and PGA1_N of U6 are connected to the differential input pins of U7 through capacitors C23 and C24 as shown in Figure 5.8. Thus a gain of 60 dB is achieved by setting the gain of U6 at 30 dB and U7 at 30 dB. The differential output pins PGA2_P and PGA2_N of U7 are given to the differential amplifier input pins. The output of differential amplifier is shifted to 1.65 V for interfacing with the microcontroller ADC as shown in Figure 5.10. The differential amplifier is realized using “Texas Instruments LME49710” operational amplifier as U8 [5].

5.2.3 Communication module

Serial communication between PC and simulator is realized using RS232 transceiver "ADM3222" as U9 and shown in Figure 5.11. It works with a single 3.3 V and provides level shifting between ±12 V level of RS232 and 0/3.3 V levels of the microcontroller. Four external capacitors of 0.1 µF are used for voltage doubler/inverter [8]. The chip has two additional pins: EN is logic low for tri-stating the receiver output, SD is a logic low pin to power down the charger pump and transmitter output thus reducing quiescent current to 0.5 µA. The enable pin EN is connected to RB2 of U4 and shutdown pin SD is connected to R3 of U4 as shown in Figure 5.2. The RX pin RA2 and TX pin RA3 of the UART module of U4

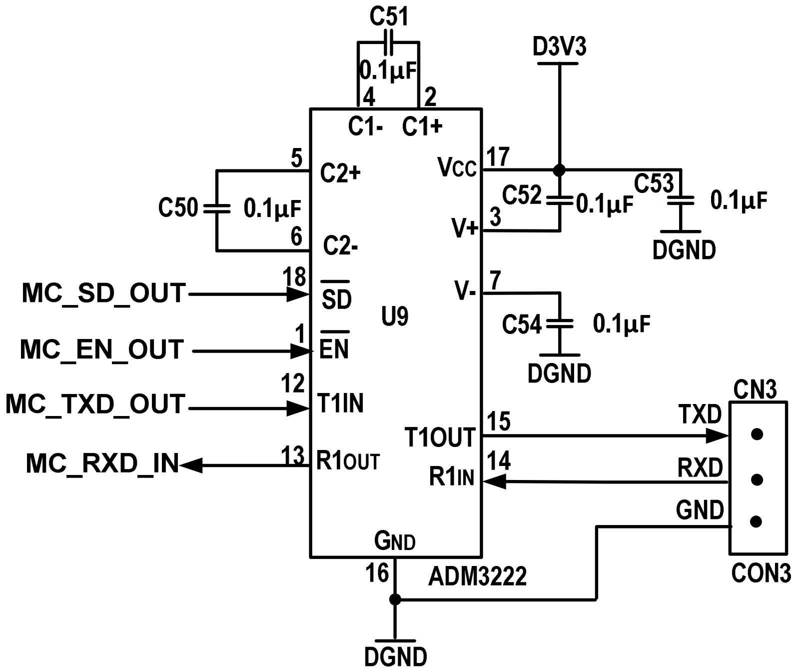

Figure 5.11: ADM3222 pin connection for serial communication.

are connected to R1OUT and T1IN of U9 respectively. The RXD, TXD and GND are available on CN3 as shown in Figure 5.11 for RS232 communication to PC.

5.2.4 Power supply

The analog part of the circuit includes preamplifiers, filters and level shifter is driven by ± 5V. The digital part of PGA2311 is driven by +5 V and digital part of PGA2500 is driven by −5V. The analog and digital part of microcontroller is supplied by +3.3 V supply. The circuit is powered by +5 V, -5 V, and 3.3 V regulated dc outputs obtained from laboratory DC power supplies.

5.3 TEOAE audiometer software

The software module includes two parts: a graphical user interface (GUI) program and microcontroller program. GUI is developed in “MATLAB R2009b” programming tool and microcontroller program is written in C with the help of “MPLAB IDE V 8.63” tool set.

5.3.1 Graphical user interface

The designed GUI is shown in Figure 5.12. Settable parameters are included such as stimulus type, number of responses to be acquired, and stimulus level. A pressing of "Start Acquisition" button sends parameter values along with start of acquisition command to the microcontroller based hardware. The acquisition procedure can be stopped by pressing "Stop Acquisition" button. Four panels are present in GUI. TEOAE Waveform panel displays acquired TEOAE waveforms such as waveform A and waveform B. "TEOAE spectrum"

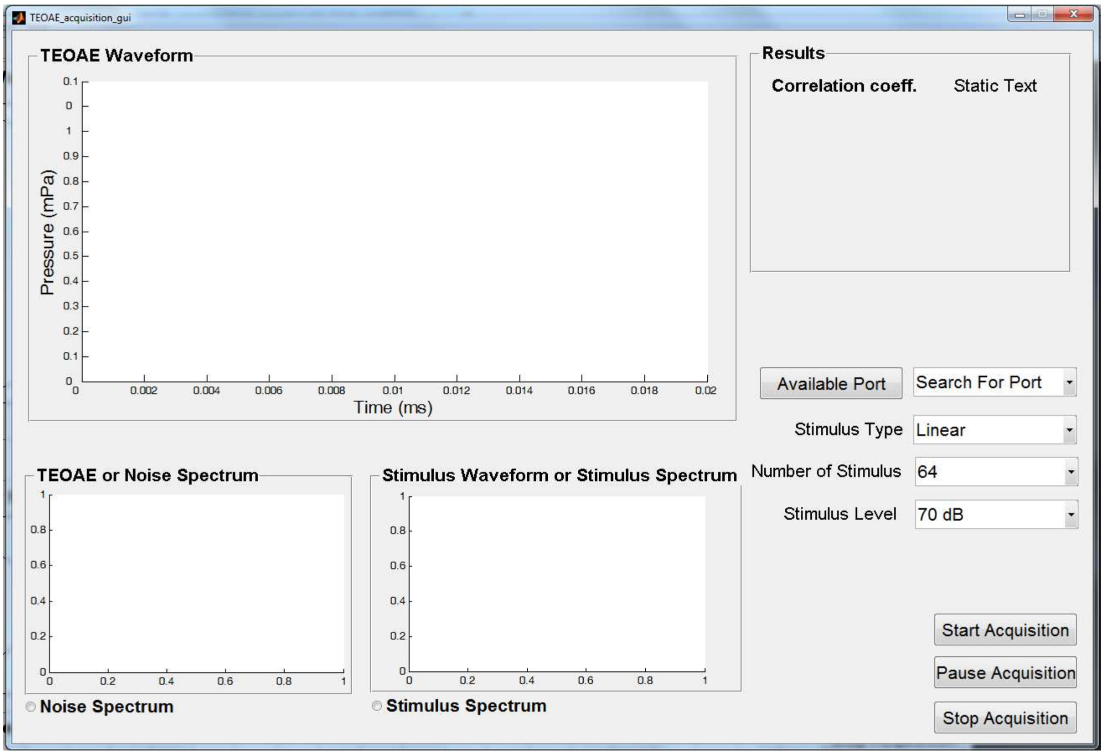

Figure 5.12: GUI of TEOAE audiometer.

panel displays spectrum of TEOAE and spectrum of resultant noise present in the averaged TEOAE response. Stimulus waveform and its spectrum are monitored in TEOAE acquisition procedure to ensure the proper fit of probe with the ear canal. The "Results" panel displays the cross-correlation between waveform A and waveform B and SNR value of frequency bands present in TEOAE waveform A. Communication between the PC and microcontroller module is realized by serial port communication. The flow chart representation of GUI program is shown in Figure 5.13.

5.3.2 Microcontroller program

The microcontroller software and consists of UART receiver program, stimulus delivery program, response acquisition, UART transmission program. The instruction from the GUI is received by the UART module and stored in respective variables assigned for each parameter. The flowchart shown in Figure 5.14 represents the interrupt service routine of the UART receiver module. The program checks whether it is a start acquisition instruction or a stop acquisition instruction. If it is a start acquisition instruction, the respective bytes are stored in the variables such as stimulus level and number of stimulus. If it is a stop acquisition function, the acquisition procedures is stopped. TEOAE acquisition is started after receiving start command from PC. The steps include stimulus delivery using DAC and response sampling using ADC. A constant amplitude square pulse of 80 µs duration is generated by left channel of audio DAC in dsPIC33FJ128GP802. Timer 1 is used to get the pulse duration of

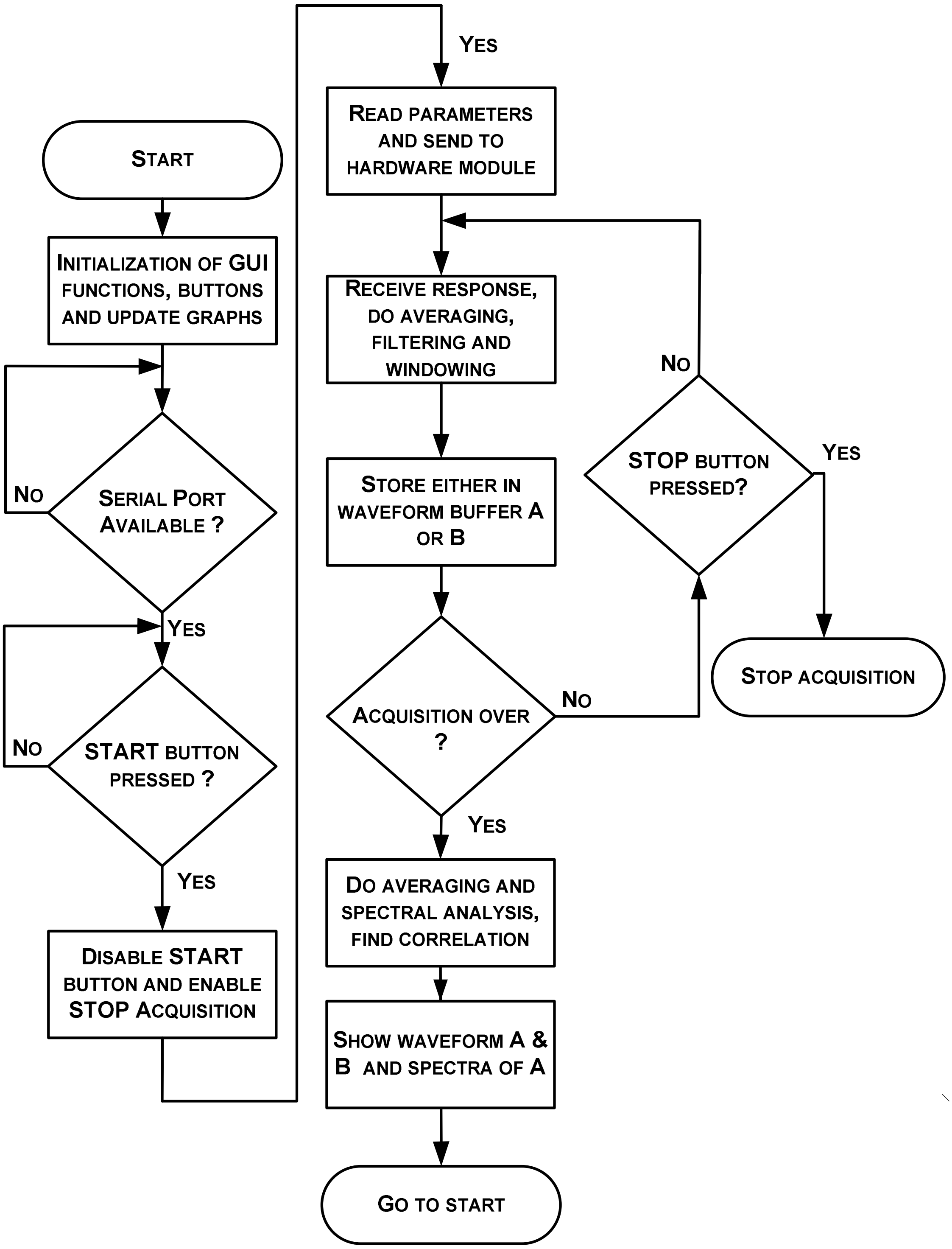

Figure 5.13: Flow chart for GUI program.

square pulse. The level of stimulus is determined by the gain of PGA2311 as mentioned earlier. The ADC module starts sampling the input with sampling period of 20 kHz for 20 ms after the stimulus onset with 12-bit resolution. Timer 3 is used to trigger the start of conversion of ADC module. The gain of PGA2500 is switched to higher level after 2 ms from the stimulus onset. The flow chart representation of TEOAE acquisition is shown in Figure

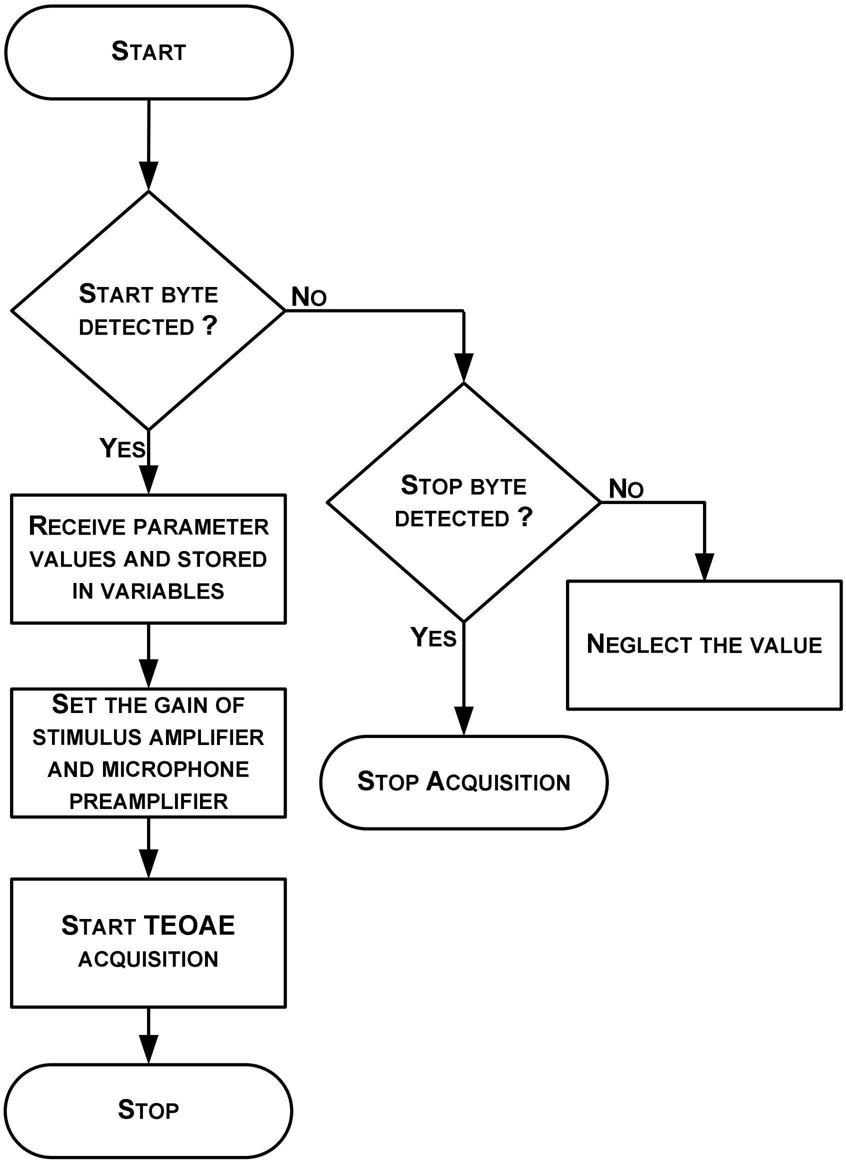

Figure 5.14: Flow chart for subroutine of microcontroller UART receiver module.

5.15. The total number of responses to be acquired is set through GUI. The acquisition program acquires four responses continuously and sends to PC. Four responses are averaged, windowed, and band-pass filtered in PC and stored alternatively either in waveform A or B. The acquisition procedure stops after the required responses are acquired and hardware module sends "acquisition completed" message to PC. The GUI program calculates correlation between waveform A and B and the value is displayed in the result panel. The noise is calculated by taking difference between waveform A and B and multiplied by $1/\sqrt{2}$. The noise spectrum is displayed in TEOAE spectrum panel. Presence of the TEOAE component is detected by calculating SNR value of frequency bands centered at 1, 1.5, 2, 3, 4 kHz. SNR value greater than 3 dB for a frequency band corresponds to the presence of an OAE component.

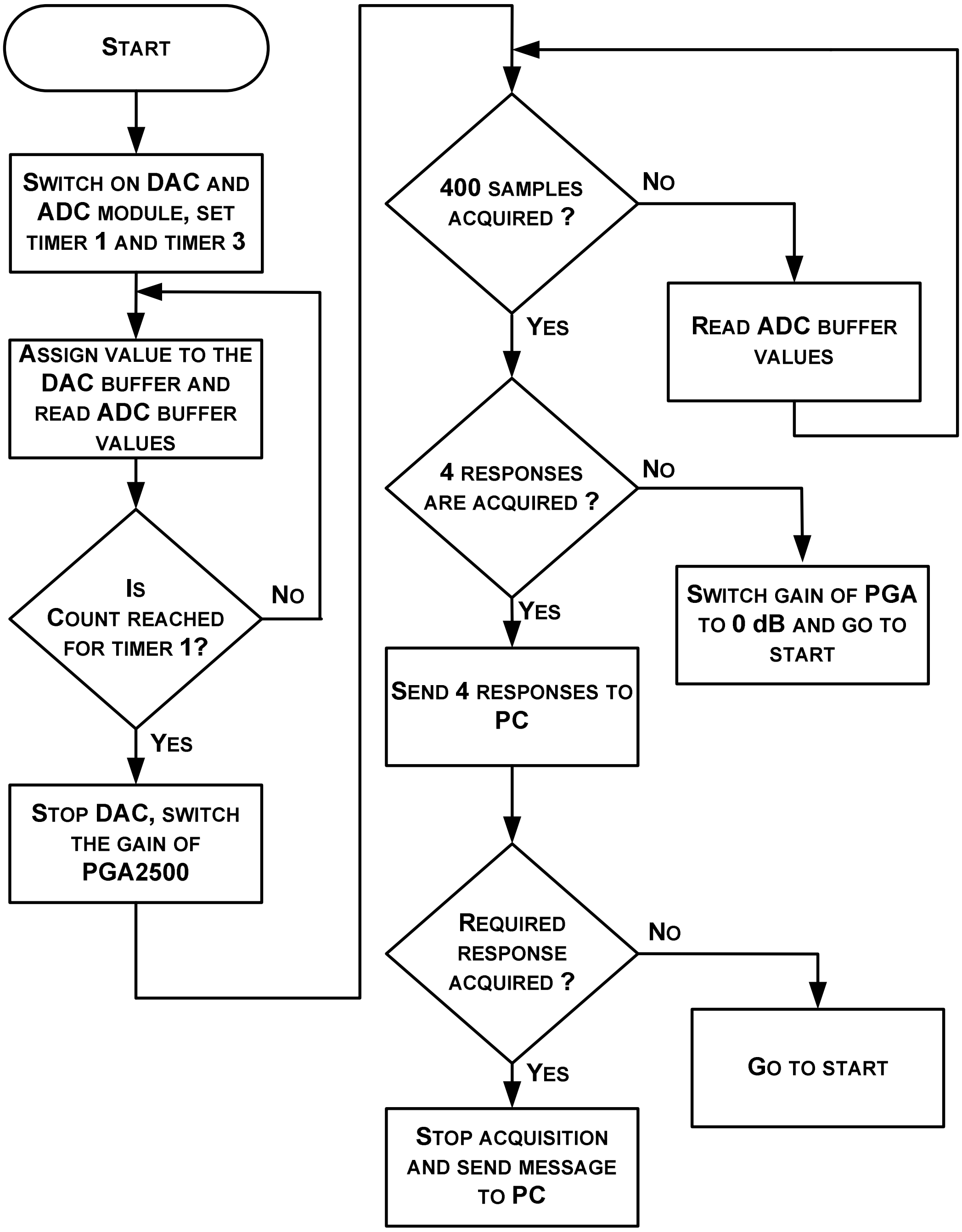

Figure 5.15: Flow chart for response acquisition program .

Chapter 6
TEST RESULTS

6.1 Audiometer output

The audiometer produces stimulus in linear stimulus protocol as well as in nonlinear stimulus protocol for TEOAE acquisition. In both protocols, stimulus is a square pulse with width of 80 µs and repeated every 20 ms. The stimulus level can be varied from 70 to 90 dB in steps of 1 dB. The selection of stimulus protocol and level of stimulus can be carried out using the audiometer GUI. To test and verify the stimulus delivery by the audiometer, the stimulus waveforms of both linear and nonlinear stimulus protocols at different stimulus levels were captured using DSO "Tektronix DPO 4034". Figure 6.1 shows a single pulse of stimulus waveform corresponding to 90 dB stimulus. The voltage level is noted to be 1.65 V. Figure 6.2 shows the waveform of stimulus corresponding to 80 dB and the voltage level is noted to be 508 mV. Figure 6.3 represents the waveform of stimulus corresponding to 70 dB and its voltage level is noted to be 160 mV. Figure 6.4 shows the stimulus pulses of linear stimulus protocol where 90 dB stimulus is delivered with a time period of 20 ms. Figure 6.5 shows the stimulus pulses of the nonlinear stimulus protocol with stimulus level selected through GUI as 80 dB. The level of first three positive pulses is 70 dB and the level of fourth negative pulses is 80 dB. Figure 6.6 shows the stimulus pulses of nonlinear stimulus protocol with stimulus level of 90 dB. The positive pulses are of 80 dB and the fourth negative pulse is of 90 dB.

6.2 Simulator output

The simulator can work either in linear stimulus protocol mode or in nonlinear stimulus protocol mode with stimulus level settable in GUI of simulator. Simulator delivers TEOAE response corresponding to the set stimulus intensity in response to the stimulus pulses detected at its input. For initial testing and verification, the stimulus input to the simulator was delivered using a function generator. The outputs from the differential amplifiers U3_VinR and U3_VinL of simulator circuit and the simulator output were captured using DSO "Tektronix DPO 4034". The dominant frequency components were selected at 1.1 kHz, 1.5 kHz, and 2.9 kHz and the simulator was set working in linear stimulus protocol mode, using GUI as shown in Figure 6.7. Neonatal ear mode was selected for the testing, where the TEOAE response is 10 dB greater than the adult ear TEOAE response.

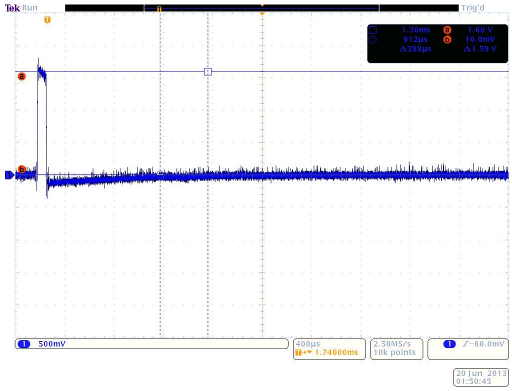

Figure 6.1: Waveform of 90 dB stimulus.

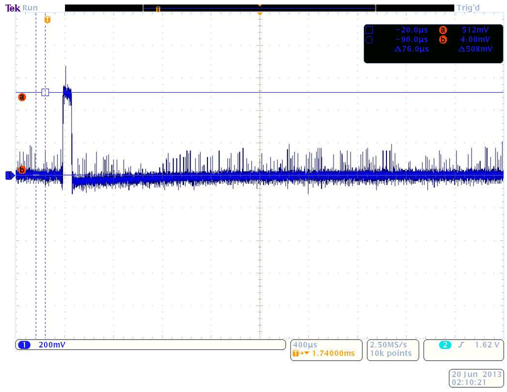

Figure 6.2: Waveform of 80 dB stimulus.

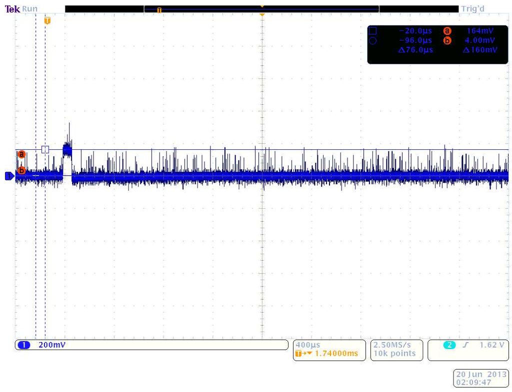

Figure 6.3: Waveform of 70 dB stimulus.

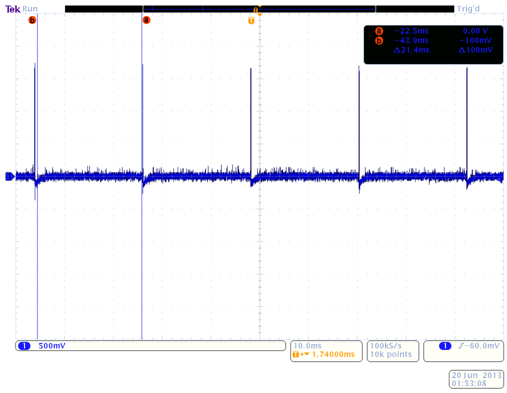

Figure 6.4: Linear stimulus protocol with 90 dB stimulus

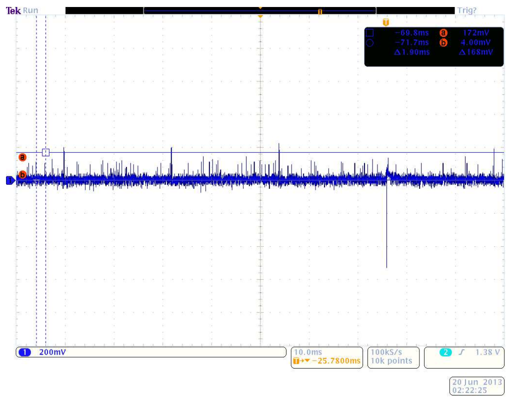

Figure 6.5: Nonlinear stimulus protocol with negative stimulus of 80 dB.

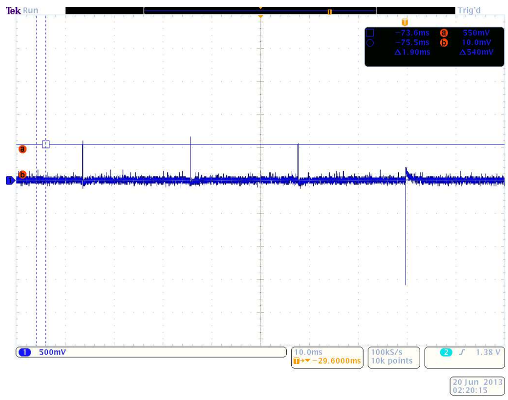

Figure 6.6: Nonlinear stimulus protocol with negative stimulus of 90 dB.

Figure 6.8 shows the waveforms for a positive 90 dB stimulus at different sections of simulator circuit, where the blue colored waveform (1) represents stimulus input, the cyan colored waveform (2) represents the stimulus artifact waveform at the output of differential amplifier U3_VinR, the pink colored waveform (3) represents TEOAE response at the output of differential amplifier U4_VinL, and the green colored waveform (4) represents the simulator output where the TEOAE response is 50 dB lesser than the stimulus artifact. Figure 6.9 represents TEOAE response for a negative 90 dB stimulus where the TEOAE response is inverted. The nonlinear stimulus protocol was tested by selecting the nonlinear stimulus

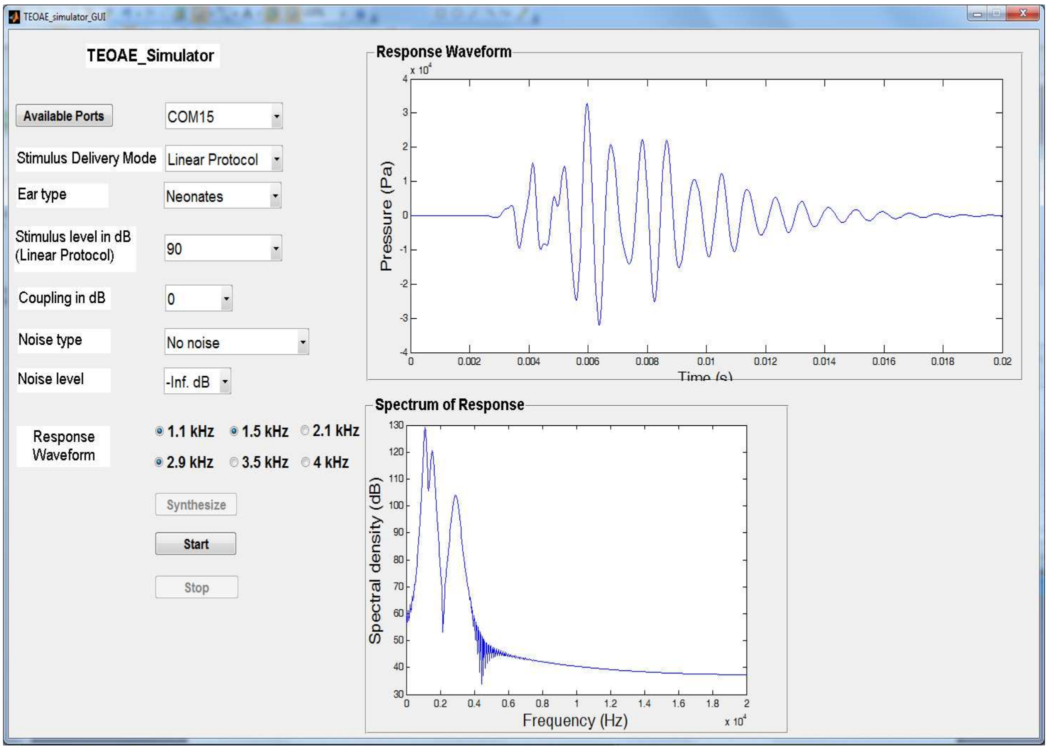

Figure 6.7: TEOAE response with components at 1.10, 1.50, and 2.90 kHz showed in simulator GUI.

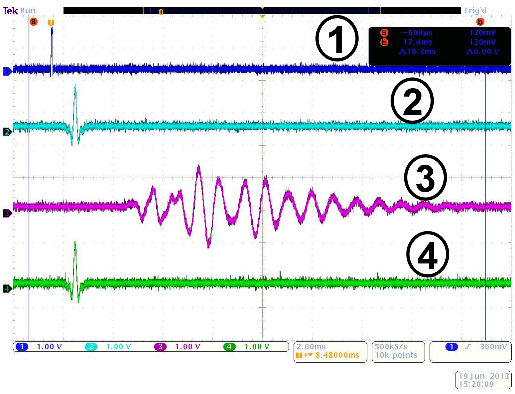

Figure 6.8: Simulator waveforms with respect to positive 90 dB stimulus.

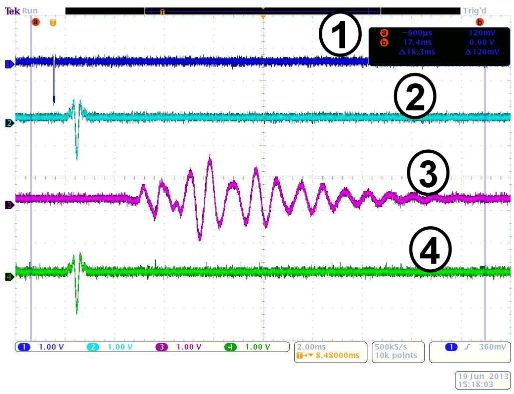

Figure 6.9: Simulator waveforms with respect to negative 90 dB stimulus.

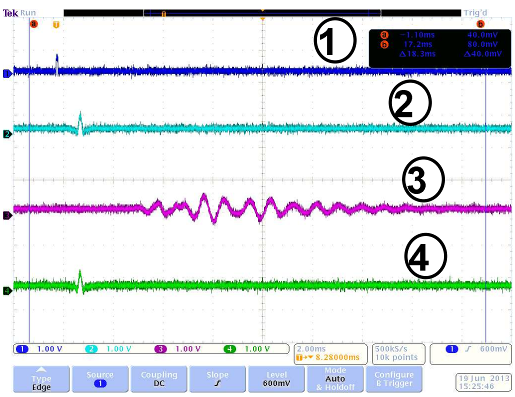

Figure 6.10: Simulator waveforms with respect to positive 70 dB stimulus in nonlinear stimulus protocol.

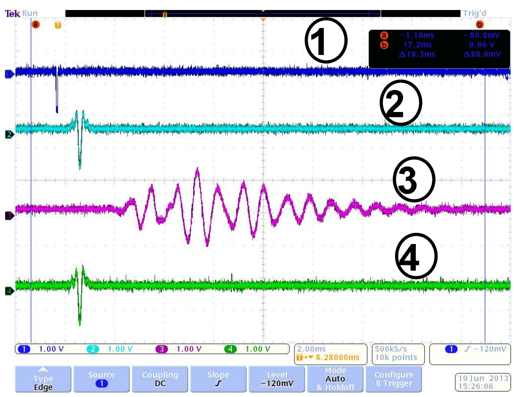

Figure 6.11: Simulator waveforms with respect to negative 80 dB stimulus in nonlinear stimulus protocol.

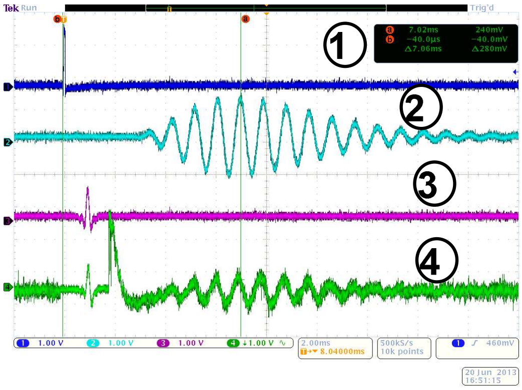

Figure 6.12: TEOAE response contains dominant frequency at 1.1 kHz with respect to 90 dB stimulus.

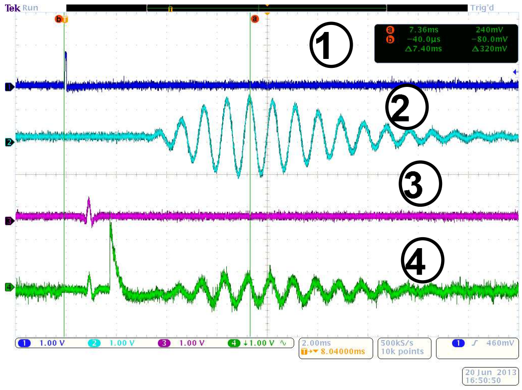

Figure 6.13: TEOAE response contains dominant frequency at 1.1 kHz with respect to 85 dB stimulus.

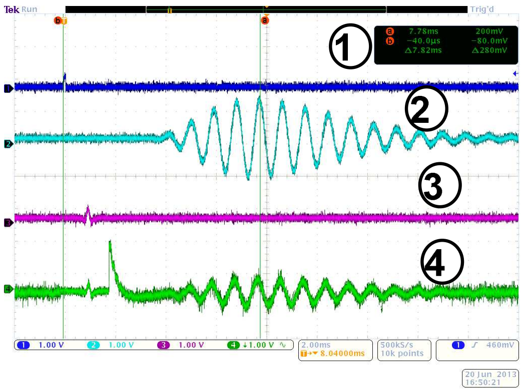

Figure 6.14: TEOAE response contains dominant frequency at 1.1 kHz with respect to 80 dB stimulus.

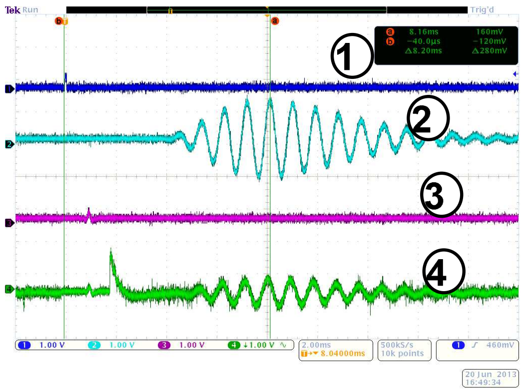

Figure 6.15: TEOAE response contains dominant frequency at 1.1 kHz with respect to 75 dB stimulus.

protocol option and the stimulus level of 80 dB in GUI of simulator. It generates first three positive stimuli of 70 dB and a fourth negative stimulus of 80 dB. Figure 6.10 shows the output waveforms for the positive stimulus of 70 dB where blue colored waveform (1) represents the positive stimulus to the simulator, cyan colored waveform (2) represents artifact the differential amplifier output U3_VinR, pink colored waveform (3) represents TEOAE response at the output U4_VinL, and green colored (4) response represents simulator response. Figure 6.11 represents output waveforms for the negative stimulus of 80 dB.

Latency variation with respect to the stimulus level was verified is shown in from Figure 6.12 to Figure 6.15. To verify the latency variation, we selected the TEOAE response with one dominant frequency at 1.10 kHz. Stimulator output was captured for the input stimulus of 90 dB, 85 dB, 80 dB, and 70 dB. In Figure 6.12, cyan colored waveform (2) represents the output from the TEOAE response amplifier U4_VinL for the stimulus input (1) of 90 dB. The latency as the time interval between stimulus onset and peak of the response was found to be 7.06 ms. Pink colored waveform (3) represents simulator output and green colored waveform (4) represents output from the differential amplifier MC_ADC_IN in audiometer as explained in the next section. Figure 6.13 shows the TEOAE response

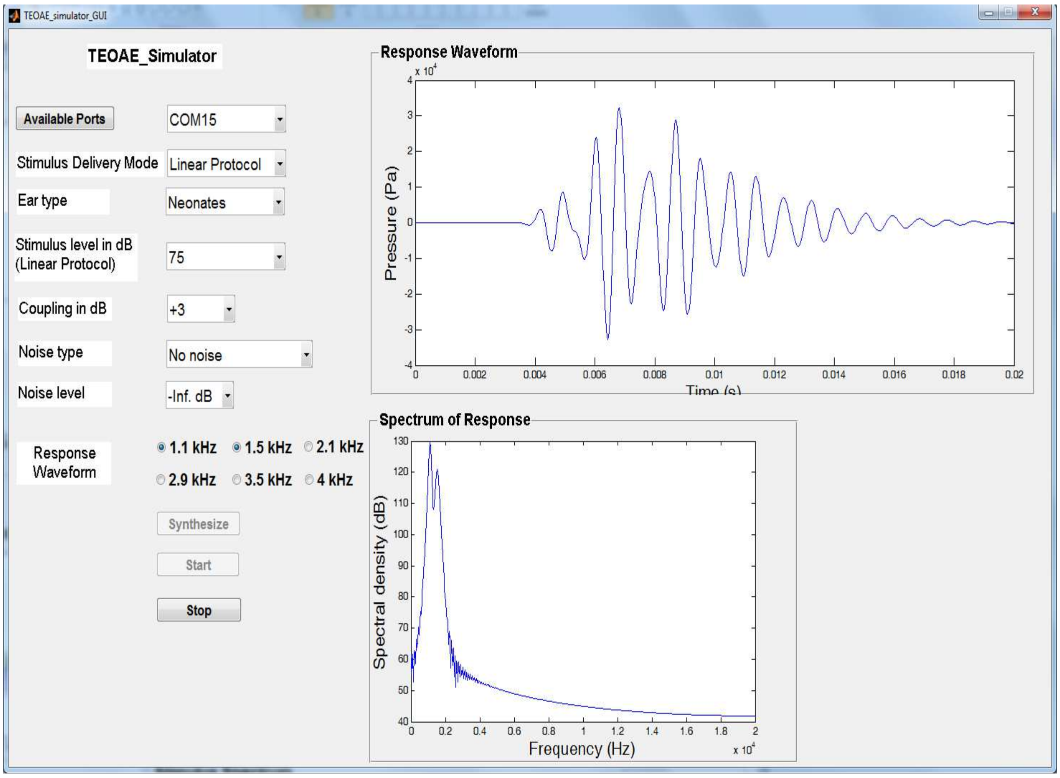

Figure 6.16: TEOAE response having components at 1.10 and 1.50 kHz for 75 dB stimulus.

waveform for 85 dB stimulus where the latency was found to be 7.4 ms, Figure 6.14 shows the TEOAE response waveform for 80 dB stimulus where the latency was found to be 7.8 ms, and Figure 6.15 shows the TEOAE response for 70 dB stimulus where the latency was found to be 8.2 ms.

6.3 Testing of simulator using audiometer

Results of initial testing of the stimulus delivery by the audiometer and response delivery by TEOAE simulator using DSO have been presented in previous two sections. Next the simulator was tested using the audiometer. The audiometer was used to deliver the stimulus at settable intensity level for linear and nonlinear stimulus protocol. The linear stimulus protocol of simulator was tested at stimulus intensity of 70, 80, and 90 dB. The audiometer delivered the stimulus at these intensities. Figure 6.16 shows the GUI screenshots of the simulator where linear stimulus protocol, neonatal ear, 3 dB coupling, and 75 dB stimulus level were selected. Figure 6.17 show the simulator output waveforms for 90 dB stimulus input. The blue colored waveform (1) is the 90 dB stimulus generated by the audiometer. The cyan colored waveform (2) is the differential amplifier output U4_VinL. The pink colored waveform (3) is the simulator output, and the green colored waveform (4) is the output from differential amplifier in audiometer MC_ADC_IN. This output was given to the input of ADC module. In addition to the stimulus artifact, one more artifact could be seen in this waveform and it was due to the gain switching of PGA. Figure 6.18 shows waveforms for stimulus input with 80

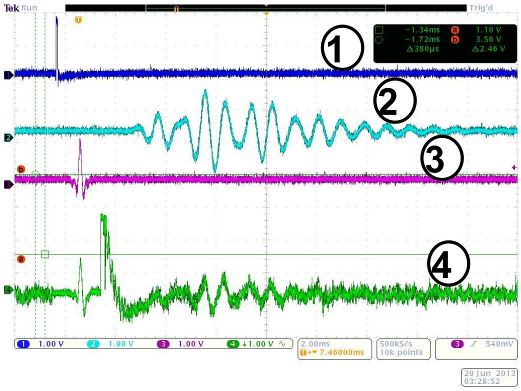

Figure 6.17: TEOAE response acquired in audiometer for 90 dB stimulus.

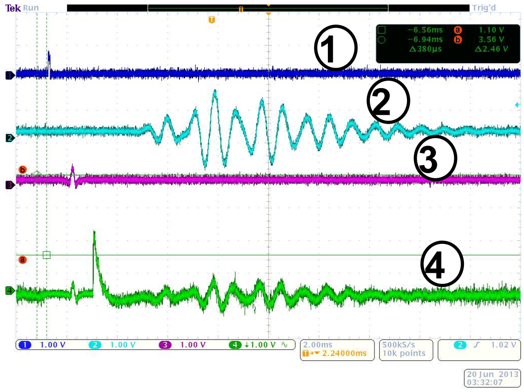

Figure 6.18: TEOAE response acquired in audiometer for 80 dB stimulus.

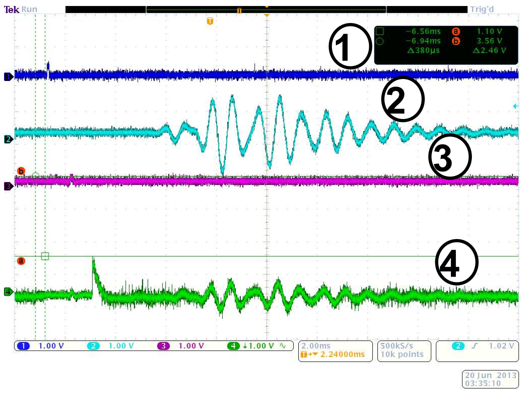

Figure 6.19: TEOAE response acquired in audiometer for 75 dB stimulus.

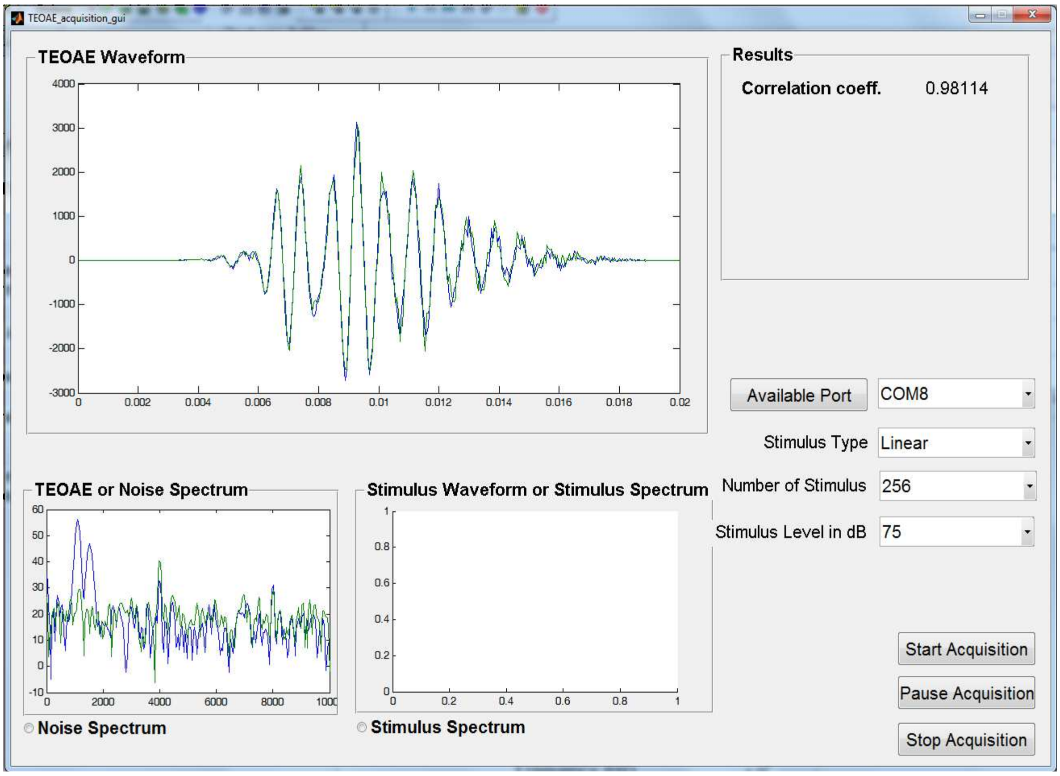

Figure 6.20: Acquired TEOAE response in linear mode is displayed in audiometer GUI.

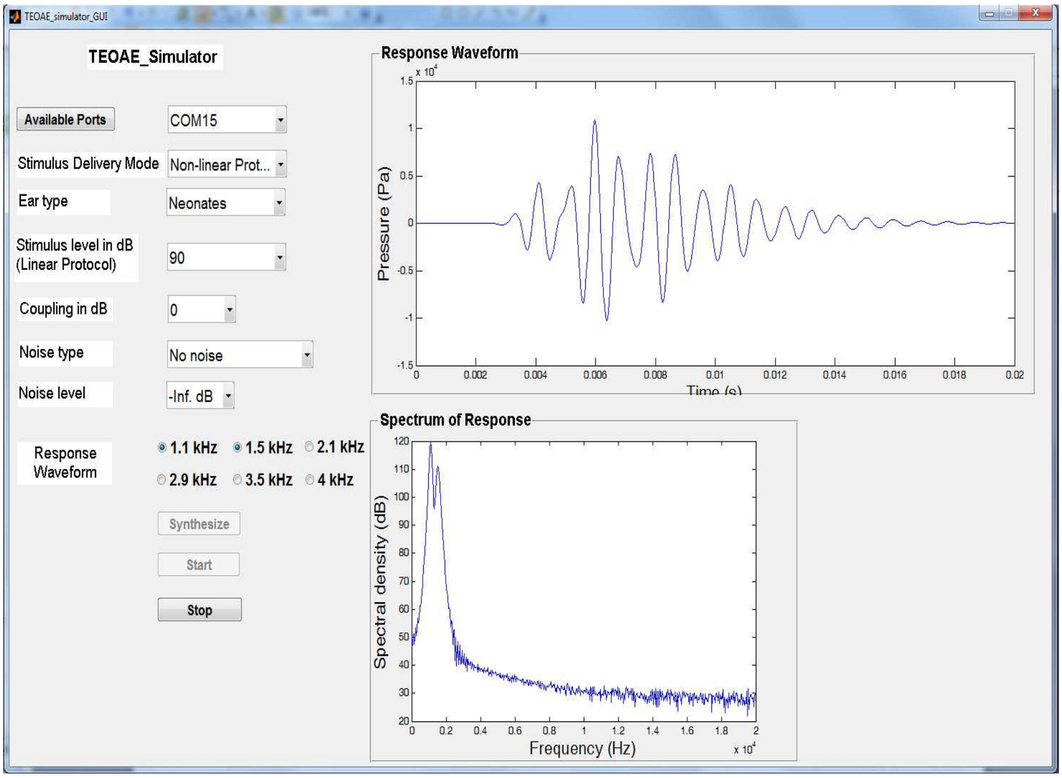

Figure 6.21: TEOAE response having components at 1.10 and 1.50 kHz for 90 dB stimulus.

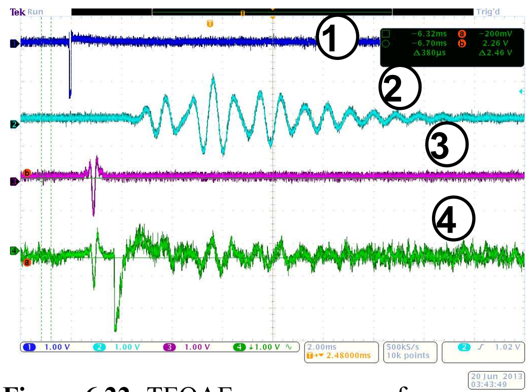

Figure 6.22: TEOAE response waveform captured by audiometer for negative 80 dB stimulus.

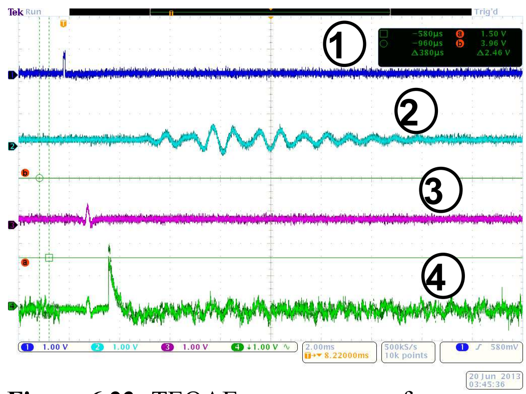

Figure 6.23: TEOAE response waveform captured by audiometer for positive 70 dB stimulus.

dB and Figure 6.19 shows waveforms for stimulus input of 75 dB. The recorded response for 75 dB stimulus by the audiometer was displayed in audiometer GUI as shown in Figure 6.20. An average of 256 responses of both waveform A (green colored waveform) and waveform B (blue colored waveform) are displayed in TEOAE waveform panel. The spectrum of waveform A was displayed in TEOAE spectrum panel where the dominant frequency component at 1.1 kHz and 1.5 kHz is noted. A correlation coefficient of 0.98 was obtained between waveform A and waveform B.

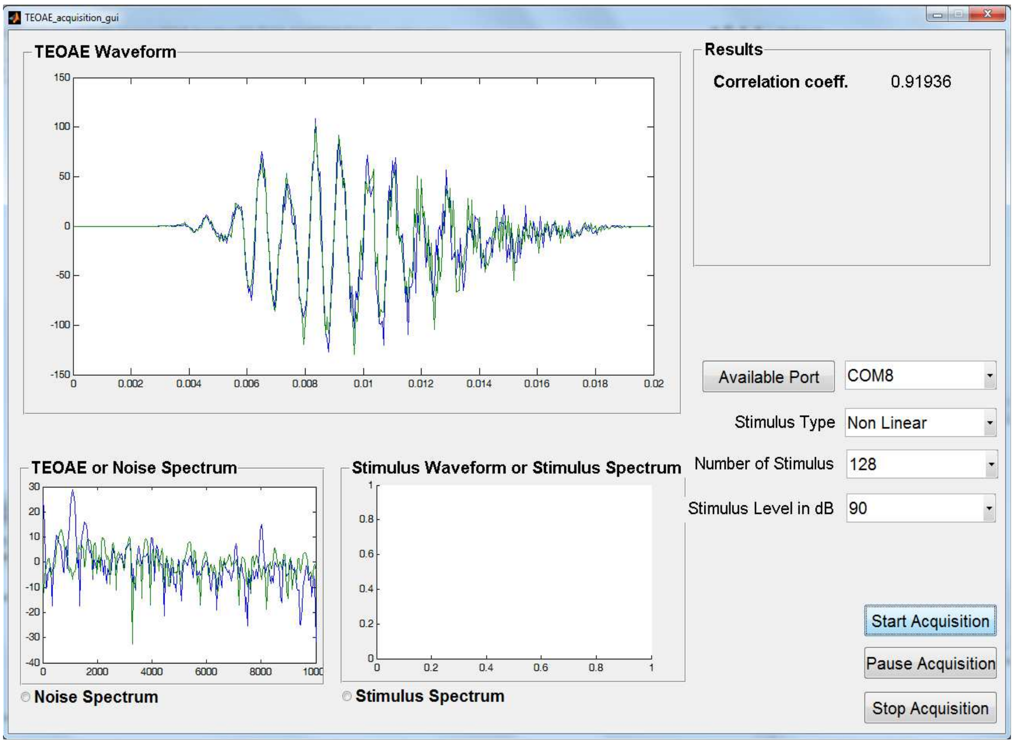

Figure 6.24: Acquired TEOAE response in non-linear mode is displayed in audiometer GUI.

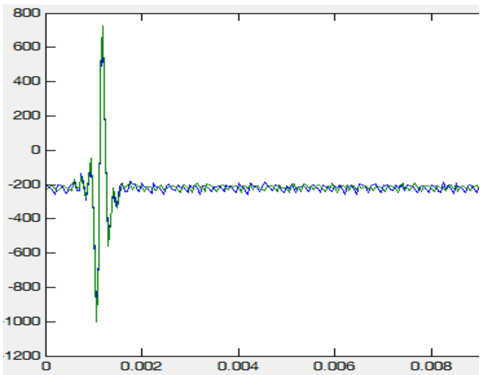

Figure 6.25: Artifact in nonlinear stimulus protocol.

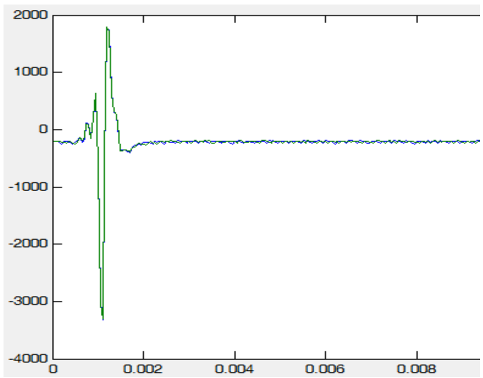

Figure 6.26: Artifact in linear stimulus protocol.

Nonlinear stimulus protocol of simulator was tested with the audiometer. The nonlinear mode and stimulus level of 80 dB were selected in simulator GUI as shown in Figure 6.21. The stimulus was generated by the audiometer and delivered to the simulator. Figure 6.22 shows the waveforms with respect to the negative stimulus of 80 dB where blue colored waveform (1) represents the stimulus waveform from audiometer to the simulator, cyan colored waveform (2) represents the TEOAE response at the output U4_VinL of differential amplifier, the pink colored waveform (3) represents the simulator output, and green colored waveform (4) represents the output MC_ADC_IN from the differential amplifier of audiometer. The recorded response is displayed in GUI screenshot of audiometer as shown in Figure 6.23. An average of 256 responses of both waveform A (green colored waveform) and waveform B (blue colored waveform) are displayed in TEOAE waveform panel. Dominant frequency element at 1.10 kHz and 1.50 kHz were noted in the spectrum of waveform buffer A.

The cancellation of artifact in nonlinear stimulus protocol was checked. The artifact of one block (three positive pulses at 80 dB and one negative pulse at 90 dB) was summed as shown in Figure 6.25 and its peak-to-peak was founded to be 1600 units. The artifact with respect to four stimuli of 90 dB in linear stimulus was summed and its peak-to-peak was founded to be 5000 units as shown in Figure 6.26. Thus a significant reduction in the stimulus artifact is noted in the nonlinear stimulus protocol.

Chapter 7
SUMMARY AND CONCLUSION

The project involved developing a transient acoustic emission simulator. The simulator has been developed to work with both linear stimulus protocol and nonlinear stimulus protocol. The stimulus level can be set from 70 to 90 dB in 1 dB steps. The number of dominant frequency components in the TEOAE can be set using the simulator GUI. The waveform and spectrum of TEOAE response are shown in the GUI screen to help in verifying the recorded response by an audiometer. The latency and amplitude of each dominant component is related to the stimulus level. A TEOAE audiometer has also been developed to test the TEOAE simulator. The audiometer can be used to deliver stimulus both in linear stimulus protocol and nonlinear stimulus protocol and to acquire response from the simulator, which is sent to PC for further processing and analysis. The simulator and audiometer both are designed using a microcontroller based circuit and PC-based GUI for parameter setting.

The audiometer and simulator were initially tested using DSO and function generator. After verification of their satisfactory operation using DSO, the simulator was tested extensively using the audiometer and all modes were found to work satisfactorily.

A PCB needs to be designed for assembly the simulator circuit. As RS232 ports are generally not available on PCs and other computing devices, a USB interface needs to be incorporated in the simulator design.

Appendix A

COMPONENT LISTS & CIRCUIT ASSEMBLY

Table A.1: Component list of the simulator

Component designator	Component description	Part number / Value	Quantity
C3, C4, C6, C11, C13, C16, C19, C20, C21, C22	Capacitor, ceramic	0.1 µF	10
C1, C2, C7, C8, C9, C10, C17, C18	Capacitor, ceramic	1.0 µF	8
C5, C12, C14, C15	Capacitor, electrolytic	10 µF	4
R4	Resistor	100 Ω	1
R18	Resistor	1 kΩ	1
R16	Resistor	4.7 kΩ	1
R1, R2, R3, R5, R6, R7, R8, R13, R9, R10, R11, R12, R14, R15, R17	Resistor	10 kΩ	15
D1,D2	Diode	IN4148	2
CN1, CN3	Connector, 2-pin	CON2	2
CN2	Connector, 5-pin	CON5	1
CN4	Connector, 3-pin	CON3	1
U2	IC, microcontroller, 28-pin, DIP	DSPIC33FJ128-GP802	1
U1, U3, U4, U6	IC, audio operational amplifier, 8-pin, DIP	LME49710	1
U5	IC, programmable gain amplifier, 16-pin, DIP	PGA2311	1
U7	IC, serial transceiver, 18-pin, DIP	ADM3222	1

Table A.2: Component list of the audiometer

Component designator	Component description	Part number / Value	Quantity
C3, C4, C17, C18	Capacitor, ceramic	0.001 µF	4
C7, C9, C12, C16, C13, C21, C22, C23, C24, C29, C34, C30, C32, C28, C35, C36, C45, C46, C42, C43, C40, C49, C50, C51, C52, C53	Capacitor, ceramic	0.1 µF	26
C1, C2, C5, C6, C19, C20, C26, C25, C37, C38, C48, C47	Capacitor, ceramic	1.0 µF	12
C8, C10, C11, C14, C15, C31, C33, C27, C1, C44, C39	Capacitor, ceramic	10 µF	11
R9, R10	Resistor	10 Ω	2
R15	Resistor	100 Ω	1
R6	Resistor	1 kΩ	1
R1, R2, R3, R4, R11, R12, R13, R14	Resistor	10 kΩ	8
R5, R4, R8, R7	Resistor	20 kΩ	4
D1,D2	Diode	1N4148	2
CN1	Connector, 2-pin	CON2	1
CN2	Connector, 5-pin	CON5	1
CN3	Connector, 3-pin	CON3	1
U4	IC, microcontroller, 28-pin, DIP	DSPIC33FJ128-GP802	1
U1, U2, U5, U8	IC, audio operational amplifier, 8-pin, DIP	LME49710	4
U3	IC, programmable gain amplifier, 16-pin, DIP	PGA2311	1
U7, U6	IC, microphone preamplifier, 28-pin, SOIC	PGA2500	2

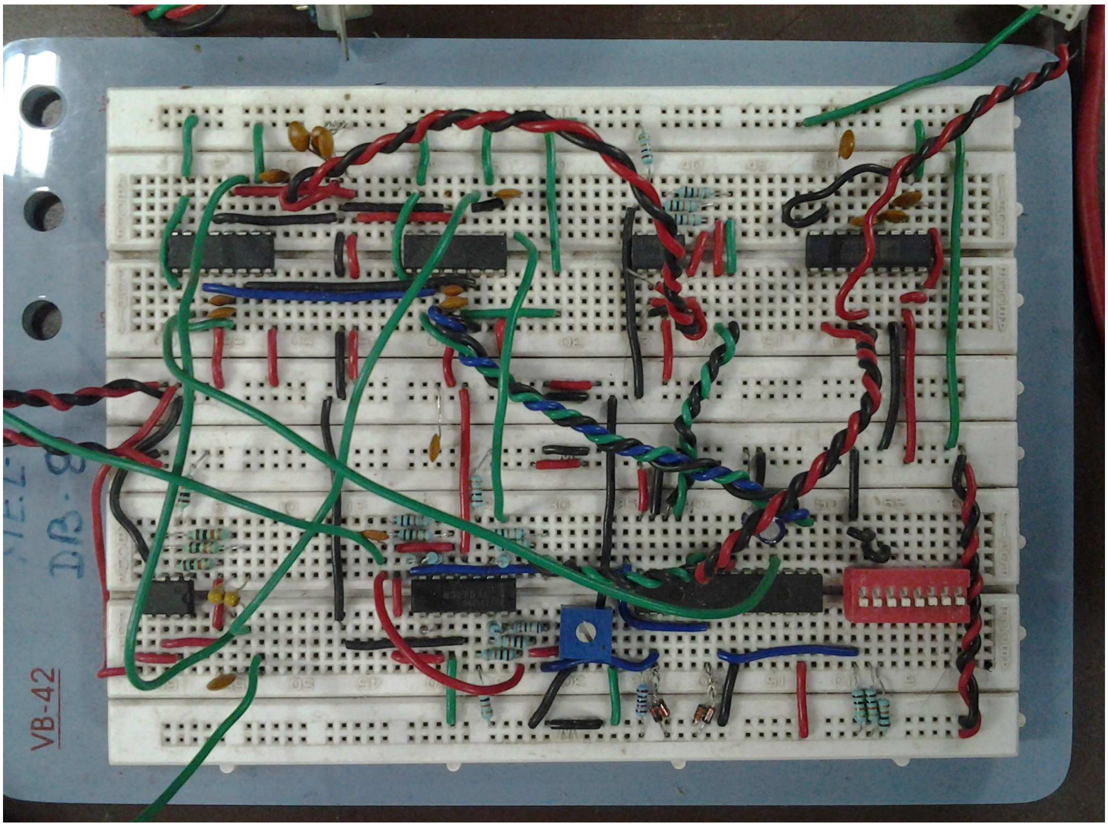

Figure A.1: Simulator hardware circuit developed on bread board.

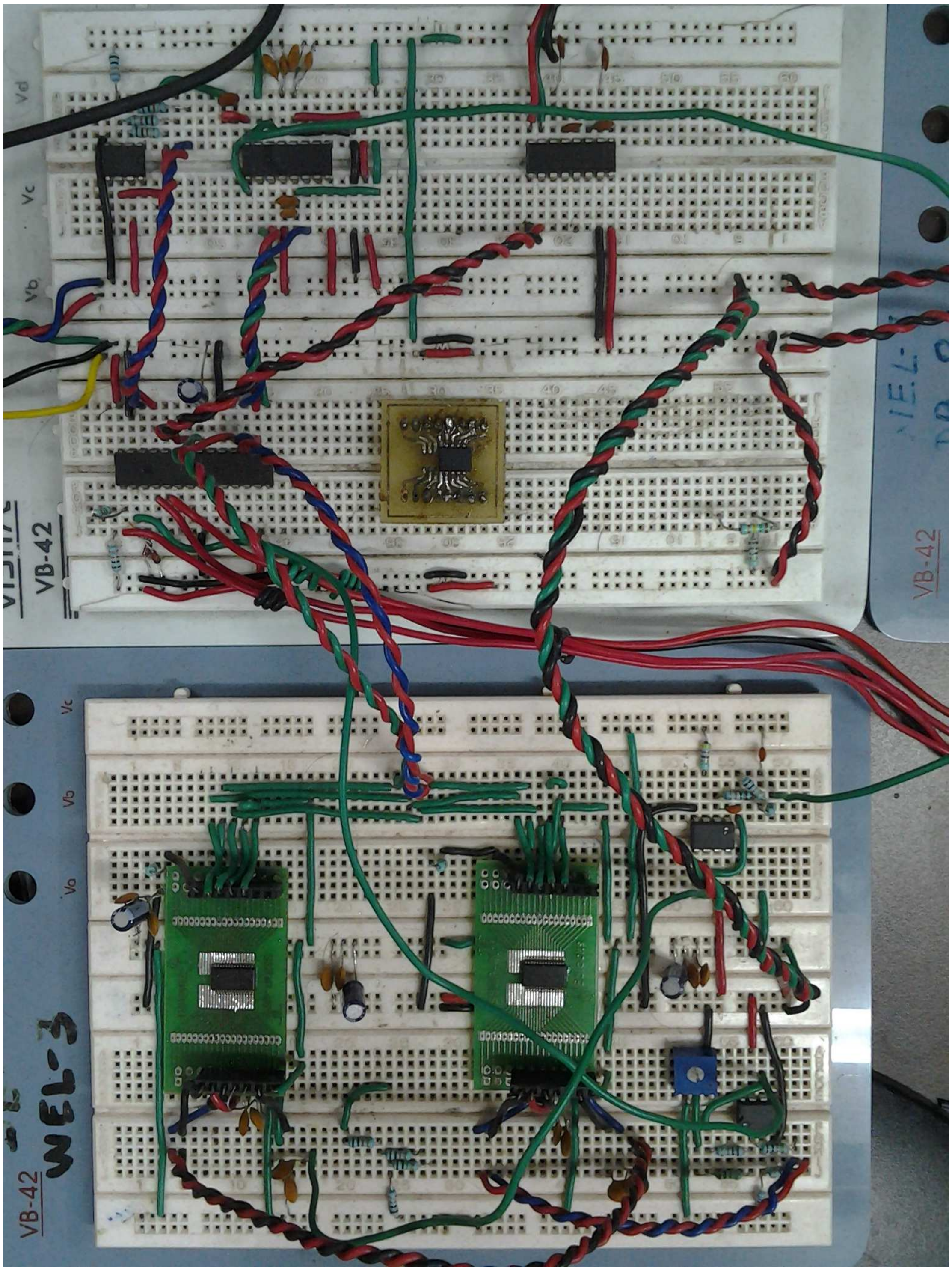

Figure A.2: TEOAE audiometer hardware developed on bread board.

Appendix B

COMERCIALLY AVAILABLE INSTRUMENTS FOR TEOAE AUDIOMETRY

B.1 Introduction

The Institute of Laryngology and Otology (ILO) of University College of London discovered OAEs. They started manufacturing TEOAE measuring instrument in collaboration with Otodynamics Ltd., UK. The first instrument was ILO88. Later other TEOAE instruments (ILO88*i* and ILO88 Echoport) have been developed with hardware upgrade, interface to PC, and software with enhanced features. This appendix provides a brief description of ILO88 and some other commercially available instruments and probe for TEOAE audiometry.

B.2 ILO88 device

"ILO88 Otodynamic Analyzer" as shown in Figure B.1 performs an auditory screening objectively. The test is to check the normal functioning of the cochlea and the middle ear. The device is connected to PC and is operated using software "ILOV5". The device has a transducer probe to deliver acoustic stimulus to ear canal and to record the delayed response. A band pass filter is used to separate required signal from noise. The device measures only signals delayed by 2.5 ms after the stimulus delivery. By removing residual linear acoustic artifact, the device is sensitive to signals which have originated at the cochlea and passed through the middle ear to the ear canal. Good middle ear function is essential for the passage of signals to and from the cochlea. Detection of TEOAE response represents good middle ear function. The same way the device cannot be used assessing the middle ear function. Following steps represent sequence of events during the automated operation of ILO88 to test an ear.

1. The OAE transducer probe is inserted to an ear canal with a disposable plastic tip.
2. The fitting of the probe with the ear canal is checked by monitoring the standard stimulus waveform and its spectrum. The user can adjust the fit of the probe for proper measurement.
3. After checking the probe fit, the test is begun. The stimulus is delivered and the delayed response is acquired repeatedly. Consecutive waveforms are added to separate buffers (A and B). Filtering followed by windowing is used to separate low level cochlear signal from the background noise. Cross-correlation of buffer A and B is used to test the validity of TEOAE response. The acquired waveform, waveform

Figure B.1: ILO88 OAE measurement system [14].

spectrum and SNR value of each selected bands will be displayed on termination of acquisition.

The TEOAE waveform can be obscured by improper functioning of OAE probe, improper fitting of probe with ear canal, excessive noise, middle ear dysfunction and cochlear dysfunction. The spectrum of TEOAE response and noise level is provided separately in GUI. The presence of TEOAE can be identified by examining both the waveform and its spectrum. Sometimes low-frequency noise in TEOAE response makes the waveform uninterpretable, and spectral analysis helps in locating TEOAE components at high frequency. Failure to detect TEOAE indicates dysfunction of cochlea or middle ear or both. OAEs are not normally present in ear with hearing loss greater than 20–30 dB SPL at stimulus frequency. ILO88 is not useful in finding the level of impairment and it is at best useful as a frequency-specific screener.

B.2.1 Modes of ILO88

Six modes are present in ILO88, five modes are used to measure OAE and one mode is used to check the device function. *Original ILO88 test mode (Nonlinear check):* It is a nonlinear click evoked TOAE measurement. After each stimulus, the response is recorded for 20 ms. It is used to capture both high frequency and low frequency component of TEOAEs. It is a robust TEOAE measurement technique where OAEs are least affected by the stimulus artifact. In this method, a part of TOAE is rejected resulting in a reduced signal to noise ratio and test specificity. *Linear click mode*: The device performs TEOAE acquisition using a constant click stimulus. The response will be recorded for 20 ms after the onset of stimulus. An experienced user selects this method to obtain higher signal-to-noise ratio and test specificity. The user must recognize the presence of stimulus artifact in the earlier part of the response. This method is used when the stimulus level is below 75 dB SPL. *QuickScreen* *mode:* It uses stimulus containing medium frequencies of 1 to 4 kHz. The response is recorded for 12.5 ms after the stimulus onset. This mode is used to get fastest TEOAE response. This mode is applicable for newborn screening where the probe fit and acoustic characteristic of ear canal results in attenuation of signal below 1 kHz. The noise present in newborn screening is of low frequency and obscures the OAE components below 1 kHz. *Tone pip stimulus mode*: A tone pip stimulus at frequency specified by the user is used to perform a tone burst OAE. If no response is obtained in a frequency range using click stimulus method, tone burst at frequency of missing is used to confirm the absence of TEOAE. *SOAE activity check mode:* It performs a search for SOAE component, which is recorded along with TEOAEs response. Probe cavity test: This test is used to check the functioning of instrument.

B.2.2 TOAE testing procedure

After completing the documentation of patient and placing OAE probe at the ear canal, the device goes through sequence of procedure such as Check fit, monitoring the noise level, delivery of stimulus, acquisition of response, validation of TEOAE response along with displaying of waveform and its spectrum.

ILO88 performs an acoustic probe fit test called Checkfit. This test is performed to find the fit of probe with the ear and monitor the stimulus waveform. Two window panels such as waveform panel and spectrum panel are used to display the stimulus monitored in the ear canal. This provides a real-time feedback so that the effect of adjustments to the probe can be seen. If a click stimulus is being applied, the stimulus waveform should be as short as possible and normally less than 1 ms in duration. The sensed stimulus waveform can be used to detect ear fitting problem. Figure B.2 shows an example of good adult probe fit. For newborns, some acceptable oscillation is found as shown in Figure B.3. Influence of excessive noise during testing procedure can be detected and Figure B.4 shows an example of stimulus waveform with noise. This effect may be due improper fit, allowing external noise to enter the ear canal or the noise induced by the patient movement. The device continuously monitors the noise since a small amount of noise can affect OAE recording. By default, the maximum acceptable noise plusTEOAE response level is 48.8 dB SPL (5.5 mPa) and it is displayed in the noise panel. The acceptable value can be adjusted by the user. As the test proceeds the number of accepted (ΣN) and rejected response (XN) is shown in the noise panel. As described earlier, there are six modes of testing present in ILO88. By completing Checkfit test and setting rejection threshold value at noise panel, a menu of test modes appears. After selecting test mode, the device delivers stimulus sequence to acquire OAE response. The noise is continuously monitored and the test is halted if excessive noise is detected. The stimulus is also monitored in between acquisition of response and gives warning if stimulus exceeds 94 dB SPL. Termination of test can be done either manually or

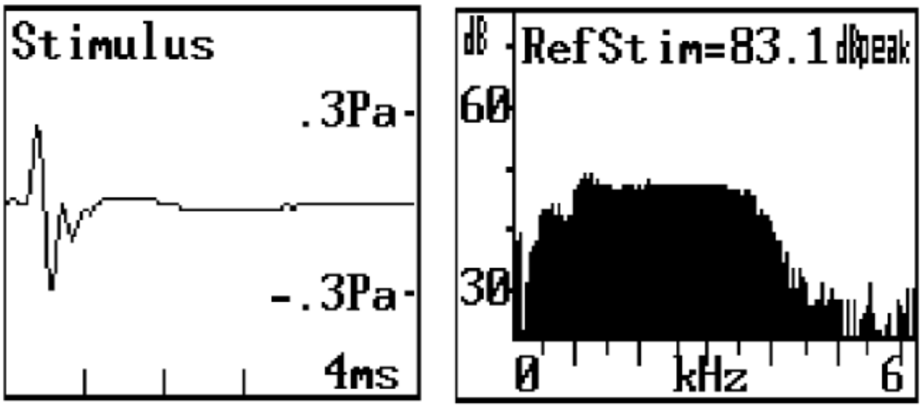

Figure B.2: Stimulus waveform and its spectrum of a good adult probe fit, based on [14]

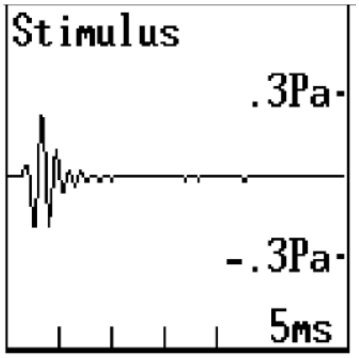

Figure B.3: Acceptable oscillation of stimulus at newborn ears [14].

automatically. Whenever Enter key is pressed, test will be terminated and response can be analyzed. The waveform and its spectrum will be displayed in the screen. Automatic termination is based on stop logic protocol set by the user. The response acquisition procedure will be terminated whenever boundary condition is met. For example, the settable parameters such as LONGTIME: The maximum allowed time for the test to run, MANYREJECTS: The test will be terminated if this number of responses is rejected due to noise. MUCH DATA: The test will be terminated if this number of responses is collected for averaging. HIGHSTIMULUS: The test will be terminated if level of three consecutive stimuli is greater than this value, etc.

ILO88 has a number of features to eliminate the artifact from the OAE response such as the noise reject system, low frequency cut filter, data window to exclude stimulus oscillations from response and the non-linear detection system. Nonlinear stimulus protocol used is to separate nonlinear response, from linear artifacts. An experienced user may select the linear stimulus option to record the whole OAE signal, if he/she is able to interpret the extended oscillation of ear canal response. The user has to be cautious during linear method when the Checkfit process shows an unsatisfactory probe fitting or when higher stimulation levels are used. The minimum number of stimulus sweeps has to be greater than 60 to get

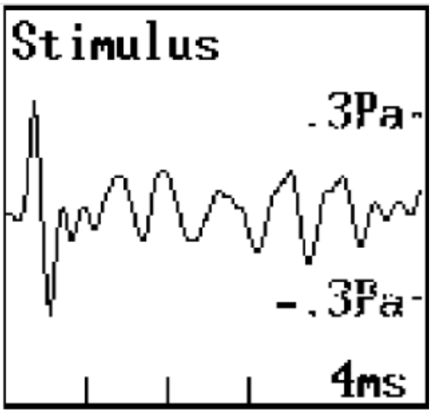

Figure B.4: Stimulus waveform influenced by excessive noise [14].

valid OAE and to avoid misinterpretation of noise as OAE response. The noise present in the test room can be analyzed by recording TOAE by keeping the probe open to the room. If the noise level prevents further recording procedures, we can conclude that the noise level is high and OAE recording is not possible in that environment. When a test is possible, there is an option to monitor the waveform of the noise. Periodic patterns generally indicate electrical interference, which can be eliminated by proper grounding of the instrument. The acoustic noise in the room is also displayed. Acoustic noise can be distinguished from electrical noise by blocking the probe firmly with the finger. This method eliminates all acoustic noise. Spontaneous OAEs appears as an acoustic interference. An option is present to distinguish between SOAEs and acoustic interference. The SOAE Check option is used to detect the presence of SOAE components. SOAE detection is done by measuring TEOAE with a reduced click stimulus intensity and rate. This method records only OAEs which are sustained for long periods after stimulation (the click rate is reduced by a factor of four and data is recorded from 20 to 80ms post stimulus time). Since spontaneous OAEs are readily synchronized with this stimulus but external interferences are not. Thus the presence of TEOAEs can be confirmed if the unusual waveform and spectrum is due to the presence of SOAEs.

The response panel as shown in Figure B.5 is an example for the TEOAE measurement of a healthy adult ear. Panel A shows the stimulus waveform at the ear canal. This waveform lasts for 1 ms if the probe is properly fitted with the ear. The spectrum of stimulus at ear canal is shown in Panel B. The spectrum shows a decrease towards 6 kHz due to the characteristic of ear canal. Panel C shows the peak of the stimulus (84 dB SPL) and the stability of the stimulus (64 %). The stability value of 64 % represents that the peak value of stimulus sweep is changed by 36 %. Panel D shows the name of test e.g., ILO88 slandered nonlinear and also shows the reason for the termination of OAE test e.g., GOOD SNR!. Stimulus dB gain value represents change to the default gain of stimulus. Panel E shows noise parameters such as noise input, the rejection threshold, number of accepted sweeps (ΣN), and

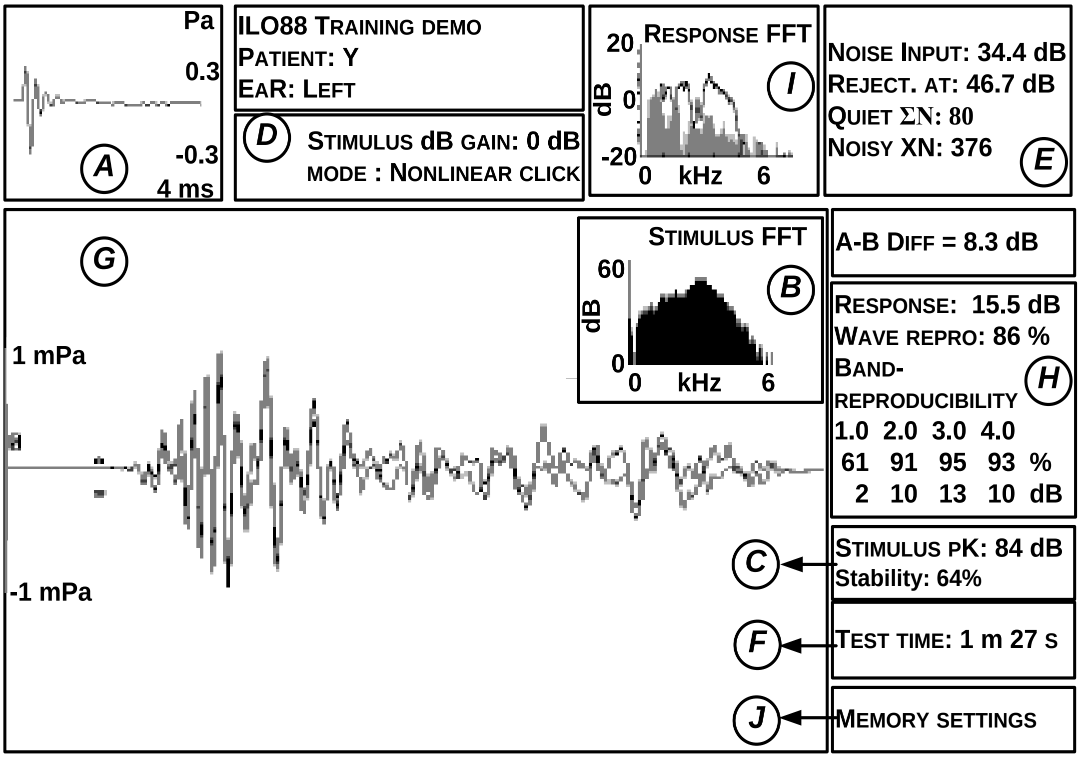

Figure B.5: An adult ear TEOAE response assessment using ILO88, based on [14].

number of rejected sweeps (XN). Noise input represents mean noise level in the ear canal. Sweep is rejected when the ear canal signal level is above Reject. At (46.7 dB SPL). Total number of accepted sweeps and rejected sweeps after thresholding is given in the panel. A-B DIFF represents noise level after averaging of responses. The total test time is shown in panel F. The response waveform is shown in panel G, contains two superimposed waveforms A and B.

Panel H provides absolute OAE energy in dB SPL and its value is 15.5 dB in this test. The noise energy is given by A-B DIFF value. The signal-to-noise ratio can be calculated by taking difference between energy of signal and noise. SNR is found to be 7.3 dB in this case. The energy of OAE is related to the characteristic of ear, and SNR is related to the quality of measurement. Correlation between two waveforms A and B (consecutive responses are stored in buffer A and B) is calculated and showed as a wave reproducibility parameter in percentile. A value greater than 50 % is considered as true TEOAE response.

The response spectrum is shown in panel I, shaded region is the spectrum of noise and unshaded region represents spectrum of OAE. The related signal-to-noise level of TOAE at four frequency bands centered at 1, 2, 3, and 4 and their reproducibility is shown in panel H. A SNR value of greater 3 dB or a reproducibility value greater than 65 % is considered as valid OAE component. It is recommended that the valid OAE detection in at least 2 bands is necessary for a pass.

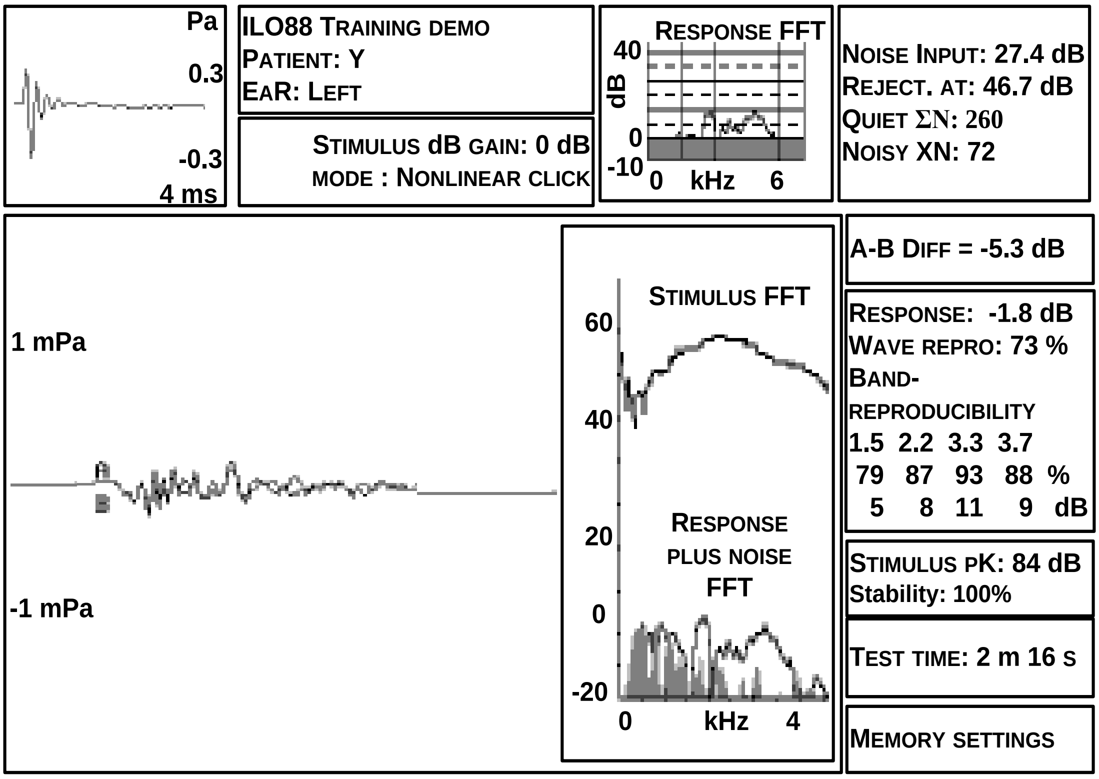

Figure B.6: Weak TEOAE response detection QuickScreen method, based on [14].

An example for the interpretation of weak OAE using Quickscreen method is shown in Figure B.6. Examination of the same ear earlier had found the total OAE intensity level to be less than that of noise level and a reproducibility of 38 %. But careful examination of OAE component around 3 to 4 kHz showed +3 db signal-to-noise ratio. So the cochlea is functioning over a limited frequency range. To confirm the marginal response, either we can repeat standard measurement or Quickscreen to enhance SNR. OAE could be recorded with a level of –1.8 dB by using Quickscreen method with noise level of –5.3 dB. The stimulus energy is concentrated more in the frequency range of 1– 4 kHz as shown in stimulus spectrum panel. The reproducibility at frequency bands 1.5, 2.2, 3.0, and 3.7 kHz are greater than 65 %. This example shows the importance of Quickscreen method in checking cochlear status in marginal cases.

B.2.3 System checks

Otodynamics provides a special test program to test each hardware part of the instrument. The probe is tested with the help of probe check option. A test cavity is provided with ILO system where the probe is inserted and checked. After inserting probe into the probe cavity, the Checkfit test is done to check the fit. The probe check option is selected after getting a good fit. The instrument delivers a synthesized OAE containing three tonebursts through the speaker and recorded through the microphone. The recorded response will be compared with the reference to check any variation. The variation in the recorded response shows the

Figure B.7: SNS newborn TEOAE probe [14].

Figure B.8: SGS general purpose TEOAE probe [15].

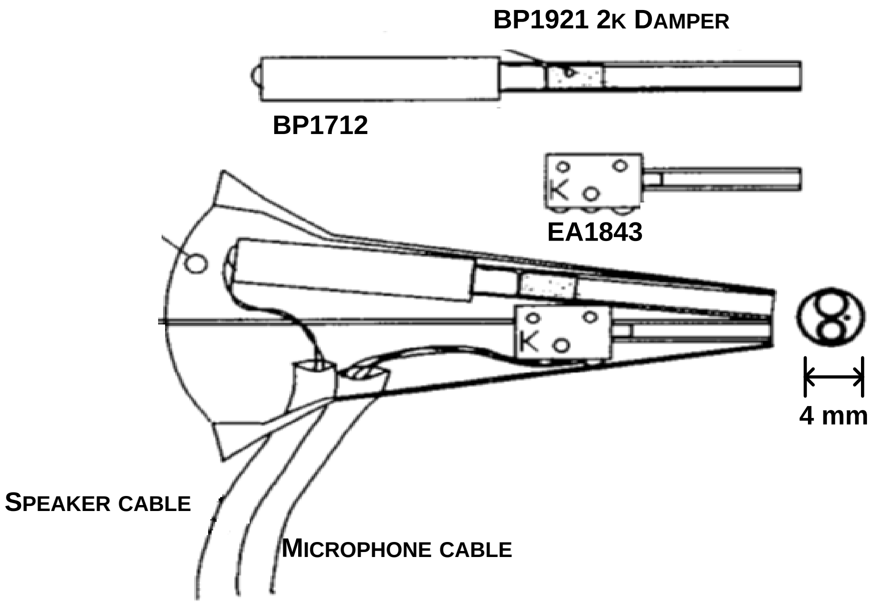

Figure B.9: Probe assembly of TEOAE adult probe [4].

deterioration of the probe. OAE probe from Otodynamics is used for response acquisition. For neonatal application, SNS Newborn TEOAE probe as shown in Figure B.7 is used and adult application, SGS general purpose TEOAE probe as shown in Figure B.8 is used. Otodynamics designed all probes by keeping both simulator and microphone transducer inside a assembly. Figure B.9 shows an adult OAE probe with BP series micro speaker and EA series microphones from Knowles Ltd. Probes are assembled in a Heine disposable type plastic speculum unit. The ground lines of both transducer and microphone are not joined in

Figure B.10: Echocheck device from Otodynamics [15].

the probe. The assembly is filled with settling filter plastic. The cabling is soft and flexible to reduce noise transmission to the probe.

B.3 Echocheck

It is a handheld TEOAE measurement device based on ILO88 from Otodynamics as shown in Figure B.10. It is designed to detect TEOAEs from healthy ears of all age in the speech bandwidth 1.5 – 3.0 kHz. A clear OAE response represents the normal middle ear and cochlear function in the frequency range of 1.5 – 3.0 kHz. The acquisition procedure of TEOAEs includes nonlinear stimulus delivery, averaging, cross-correlation, and spectral analysis. The frequency content of stimulus below 1.5 kHz is filtered out thus higher stimulus rate (97 per second) is achieved. Features include automatic pass/fail test results, indicators for noise level, stimulus/probe stability, automatic stimulus adjustment and fitting (similar to the Chekfit test of ILO88). The device can detect OAE signal in the range of –6 dB SPL to 55 dB SPL. Echocheck follows nonlinear stimulus mode of ILO88 with 80 µs square pulse. The microphone signal is monitored to maintain a constant stimulus level. The default stimulus level at the ear canal is 84 +/-3 dB SPL and recording of response starts after achieving this level. The test is terminated if the stimulus level changes from the default level. The response will be rejected if the level of windowed, filtered and averaged response from each block exceeds above 50 dB SPL. Consecutive averaged response from each block is stored in buffer A and buffer B. Averaging continues only when noise and stimulus are within the range mentioned. The cross-correlation of buffer A and B is used to find reproducibility and the difference between A and B is used to find mean noise level. Presence of OAEs is confirmed

Figure B.11: Otoportscreener TEOAE measurement system[16].

when SNR of 6 dB in at least 2 frequency bands. Technical specifications: The frequency response of the response filter: 1.6 to 2.8 (3 dB points) 4^{th} order butter worth filters, Stimulus amplifier: 0 to 6 kHz, Acoustic stimulation levels: 84 **+/-** 3dB SPL peak equivalent. Stimulus delivery: Nonlinear mode. Probe: ECP probe from Otodynamics. It consists of a TPC coupler and a BGS probe body/lid and miniature 6 pin DIN connector.

B.4 OtoportScreener

OtoportScreener, shown in Figure B.11 is a compact handheld device for screening application. It has three types of TEOAE testing: TEOAE screening, TE OAE1, and TE OAE2. Each test is started with Checkfit test as mentioned in ILO88 section. In addition to the indication of stimulus level at the ear canal, Checkfit panel indicate size of ear canal. Checkfit screen is shown in Figure B.12 where the bar on the left of the screen gives an estimate of ear canal size. The size is estimated by finding the level of click stimulus required to give 84 dB SPL in the ear canal. Larger ear canal requires a larger stimulus and small ear canal requires small stimulus. Noise level is also indicated on the same panel at the right on the screen. The shaded bar moves in response to changes in noise. The user can adjust the rejection threshold of noise. The response sample is rejected if the response plus noise level is greater than the threshold value. Stimulus level at the ear canal is displayed numerically and using needle indication. The level of stimulus is monitored and the device adjusts the stimulus to required level. Vertical position of needle indicates that stimulus level is at 84 dB SPL as shown in Figure B.12. The sytem can automatically identifies the fit of probe with the ear. It indicates presence of noise and oscillation of stimulus in the ear canal during Checkfit proess. A seperate display panel shows the waveform and spectrum of stimulus as in the case of ILO88.

Three modes are present in Otoportscreener. *TEOAE screening mode:* It can be used to detect TOAE response within small interval of time. The mode is similar to that used in the Otodynamics Echocheck or ILO Quickscreen mode. It is fully automated and stops acqusition

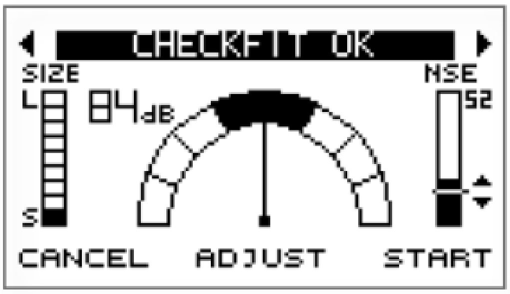

Figure B.12 Checkfit panel of OtoportScreener [16].

when an OAE component is detected. *OAE1 mode*: It is used for TEOAE measurement in the range 1 to 4 kHz. The test can be manually started and ended. Three half-octave bands with SNR of 6 dB are required for a pass. OAE2 mode: It is a replication of the settings used for the Universal Newborn Hearing Screening Programme in England. It is similar to OAE1 but requires only two half-octave bands for a pass and does not included 1 kHz among possible pass bands. The Mic Filter setting is narrower than OAE1 reducing the TEOAE recorded at 1kHz. *Factory mode:* It is provided for quality assurance purposes. More details of these modes are given in Table B.1. Stim Level indicates the default stimulus level and the stimulus level is permitted to vary at the ear canal by the value specified at Stim range. The Ring reject is the maximum allowable ratio in dB of the peak stimulus over the stimulus level recorded at 3 ms. A value greater than the specified one denotes stimulus artifact and will be misinterpreted as a geniune OAE signal. Max NLo is a test timeout function, the test is stopped when the specified number of low noise data samples (when the noise present is below the reject level) has been collected. Test time is the maximum allowded time for the test before automatic stop. Response window shows starting and ending point time window with respect to the start of stimulus, e.g. Response window of ‘3 to 13 ms’ means time window of 10 ms length is applied between 3 ms and 13 ms after the stimulus is presented. Min NLo is the minimum number of low noise data samples to be collected. Min OAE sig is the minimum energy of OAE signal needed to get a pass. Min SNR is the minmum SNR (difference between energy of OAE signal and noise). Min band sig is the minimum OAE signal required in each band. Pass bands is the total number of pass bands required to meet the overall pass criteria. Band settings provides list of bands bands which are centered at 1, 1.5, 2, 3, and 4 kHz. The minimum SNR required for each band is specified. If Autoadjust is on, then the device adjusts the stimulus level automatically to acheive default stimulus level. Otherwise the level has to be manually adjusted. If Autostart is on, the testing will be automaticlly started after Checkfit. The stimulus level at ear will be adjusted and acquisition of TEOAE response will be started. If Autostart is off, the acqusition has to be manually started after Checkfit test. If Autostop is on, the test will be terminated when the pass criteria are met and the test will be terminated after reaching NLo figure in other case. Overide otion

Table B.1: Comparative study of modes in OtoportScreener [16].

Mode Name	Screening		OAE 1		OAE 2		Factory (locked)	
TE Test Config								
Stim Level	84dB pe		84dB pe		84dB pe		84dB pe	
Stim Range	± 1dB		± 1dB		± 1dB		± 1dB	
Noise Reject	52dB SPL		52dB SPL		52dB SPL		52dB SPL	
Ring Reject	-20dB		-20dB		-20dB		-20dB	
Max NLo	260		260		260		260	
Test Time	300s		300s		300s		300s	
Response Window	3-9ms		3-13ms		3-13ms		3-13ms	
Norms	OFF		OFF		OFF		OFF	
6k Band	OFF		OFF		OFF		OFF	
TE Stop criteria								
Min NLo	30		30		40		40	
Min OAE Sig	0dB SPL		0dB SPL		0dB SPL		0dB SPL	
Min SNR	6dB		OFF		OFF		OFF	
Min Band Sig	-5dB		-5dB		-5dB		-5dB	
Pass Bands	1		3		2		3	
Band Settings	SNR	Rqrd	SNR	Rqrd	SNR	Rqrd	SNR	Rqrd
1K	6	NO	6	NO	OFF	NO	6	NO
1.5K	6	NO	6	NO	6	NO	6	NO
2K	6	NO	6	NO	6	NO	6	NO
3K	6	NO	6	NO	6	NO	6	NO
4K	6	NO	6	NO	6	NO	6	NO
TE Automation								
Autoadjust	ON		ON		ON		ON	
Autostop	ON		OFF		OFF		OFF	
Autostart	ON		OFF		ON		OFF	
Override	YES		N/A		YES		N/A	
TE Other Settings								
Mandatory	SAVE OFF		SAVE OFF		SAVE OFF		SAVE ON	
Mic Filter	1.6-3.2kHz		0.8-4.8kHz		1.2-4.8kHz		0.8-4.8kHz	

is to control the Auto start option manually. Mandatory option is used to save the response in memory. Frequency range of filters used in three modes of Otoport are shown in Mic Filter list. Technical specification are as the folloing, *Stimulus waveform*: 80 µS positive broad square wave pulse and 300 µS biphasic broadband triangular pulse. *Waveform sampling frequency*: 20 kHz. *Response buffer averaging*: The response from each stimuli are summed and averaged to reomove the artifact. These subaverages are alternatively added to two separate averages known as waveform A and waveform B. *Signal and noise calculation*: Measure of signal noise calculation is based on the correlation difference between these two waveforms. *Stimulus repetion rate*: stimulus at every 13 ms. *Response window*: 10 ms starting at 3 ms after the start of stimulus representation. and cosine filtered with with rise and fall time of 2 ms. *Response frequency band*: half octave, centerd at 1, 1.4, 2, 2.8, 4 kHz. *Response frequency range*: 841 – 4757 Hz. *Hardware processing and memory:* multiple distributed processors plus dedicated hardware DSP engine with total processor performance of 480

Figure B.13: UGS probe tip [16].

MIPS, 8 MB of test memory, 1.3 MB of program memory, 2x16 bit resolution output channels and 1x16 bit resolution input channel. Transducer: UGS probe as shown in Figure B.13. The probe tip hold ear piece in the ear canal. Calibration is performed in occluded ear canal simulator conforming to IEC 60711 (Bruel and Kjeartype 4157) with the probe mounted in a DB2012 adaptor using an Otodynamics probe tip.

B.5 Probe used in TEOAE measurement

Vast majority of OAE signals are low level signals, thus the sensitivity and noise floor of microphone has to be considered. Microphone can be placed either outside of ear or inside the ear canal. A connecting tube is used to guide the sound from ear to microphone in the first method and influvence of fricitional noise is unavoidable at quite condition also in this method. The second method is preferred where the sensitivity of microphone is low and noise floor is high. "ER10D" Probe is an OAE probe designed for both TEOAE and DPOAE measurement from Etymotic Research.

The probe consists of a probe head and a preamplifier module as shown in Figure B.14. The probe head has of two speakers and a miniature microphone. A flexible silicon tubing of 12 inches extends from probe head to the preamplifier module and a flexible tube of 5 inches from amplifier to mini DIN connector. Different models are available for ER10D include. The probe with preamplifier and with out preamplifier model. The models are divided based on gain of preamplifier that ranges from 0 dB to 60 dB. List of different models with part number is shown in Figure B.15, where ER10DP-3320-XXX represents the model number. The mini DIN connector is shown in Figure B.16. Pin 1 gives the output of microphone, Pin 2 and Pin 3 are postive inputs of two speakers, Pin 4 and Pin 5 have no connection, Pin 6 is the supply voltage negative terminal, Pin 7 is the virtual ground, Pin 8 is the supply voltage positive terminal. Microphone specification: Sensitivity of 50 mV/Pa, frequency response of ± 1dB at 1 kHz, ±4 dB between 200 and 12 kHz, noise floor of –15 dB SPL at 1 kHz, –28 dB SPL at 3 khz, and –15 dB SPL at 5 kHz (1 Hz band-width), maximum

Figure B.14: ER10D OAE probe.

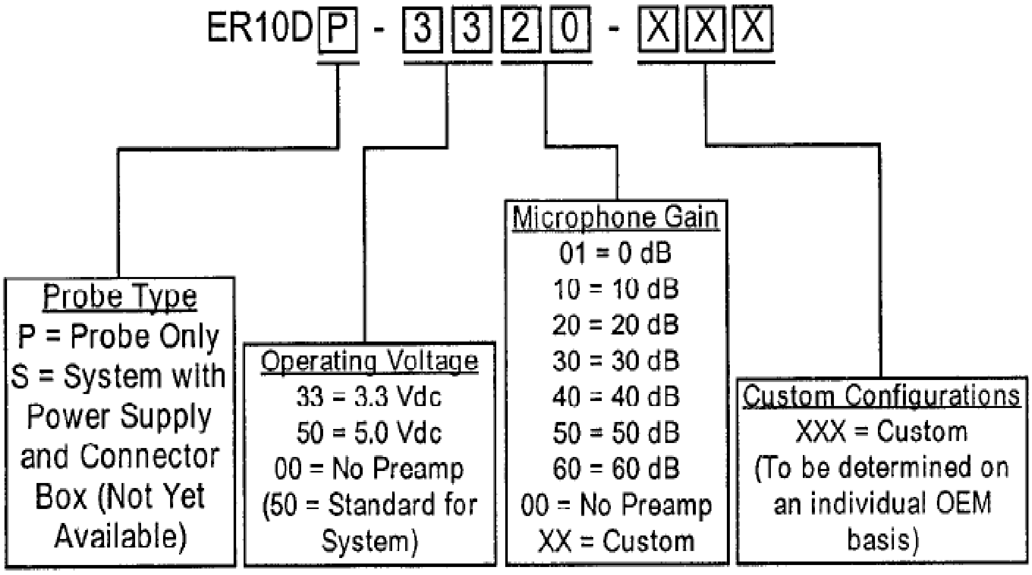

Figure B.15: Specification table of ER10D probe models.

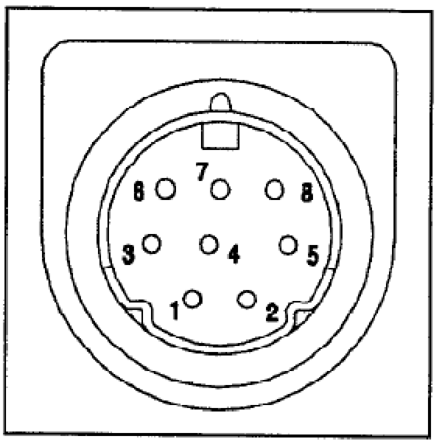

Figure B.16: Connector pin diagram of ER10 D probe.

undistorted output of 3.3 V supply is 1 Vrms (120 dB SPL), and output impedence of 100 Ω. Speaker specification: sensitivity is 86 dB SPL at 1 kHz for 1 Vrms input , maximum driver voltage is 10 Vrms, maximum output is 106 dB, and an input impedance of 1 kΩ. The maximum current consumption at 3.3 V is less than 3 mA.

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I dedicate my work to my family members.

Antony Rajan

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