# IMPLEMENTATION OF AN ALL DIGITAL ENVELOPE TRACKING HARMONIC GENERATOR

By

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## PREFACE

This project explored the implementation of an all digital envelope tracking harmonic generator. The algorithm used was developed by Scheets [4]. The harmonic generator's purpose is to allow smaller bandwidths of traffic to be transmitted by generating harmonics at the receiver that increase the signal bandwidth and provide expanded auditory response to the listener. The implementation presented in this paper was developed for the personal computer using National Instrument's LabVIEW. The LabVIEW program that was written for this project takes Waveform Audio File Format (WAVE) files as an input and allows the user to select numerous different methods for examining the input data in both the time and frequency domains. The user is also given the capability to process the file using the harmonic generator algorithm and to save the modified file in the WAVE file format so that the file can be re-examined in the time and frequency domains and listened to using a standard wave file player.

First I would like to thank Dr. Scheets for providing me with the technical guidance on this project and his patience while coping with my busy work schedule. In addition, I would like to thank my employer Frontier Electronic Systems Corporation, my supervisor Scott Phillips, and my engineering peers for backing me during the completion of my studies.

Finally, I would like to thank my wife Denise and son Reece for their patience, love, encouragement and understanding during the last three years.

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## NOMENCLATURE

| | |
|---|---|
| IFFT | inverse fast fourier transform |
| FFT | fast fourier transform |
| GUI | graphical user interface |
| PCM | pulse code modulation |
| VI | virtual instrument |
| WAVE | waveform audio file format |

# CHAPTER I

# INTRODUCTION

This project implemented an all digital envelope tracking harmonic generator. The algorithm developed by Scheets [4] is designed to allow harmonics of audio signals to be stripped out at a transmitter in order to decrease the amount of bandwidth required for transmission. Once the signal is received, the algorithm would be applied and harmonics inserted based on the frequency composition of the received signal. The harmonic generating algorithm has the potential to reconstruct, with little error, signal samples of fixed frequency sinusoids.

As with most ideas, practical implementation can be difficult and pose many unforeseen challenges. Smaller, limited implementations are usually found to provide much insight into the problems of large scale implementation. The approach used in this project was to take data in the form of Waveform Audio File Format (WAVE) files. This data is already in digital format and can be played on almost any personal computer. This allowed the project to concentrate on the algorithm and not be concerned with developing a method of acquiring audio signals, converting them into digital format and then processing the data back into the analog spectra.

The other major decision in this project was to develop the algorithm using National Instrument's LabVIEW. LabVIEW is a graphical programming language specifically designed for data acquisition and manipulation. LabVIEW has many built-in functions for processing and displaying data, again allowing this project to concentrate on the algorithm and less about topics such as file handling, standard signal processing routines and user interfaces.

With these design decisions made, the implementation was desired to have the following functionality:

- Allow the user to filter a selected wavefile to simulate stripping out of harmonics at the transmitter
- Graph the pre and post processed wave files
- Examine the wave file in the frequency domain
- Allow the user to generate multiple harmonics based on multiple signal peaks in the original signal's frequency domain
- Provide a method to save the newly processed file in the wave format to allow signal playback

The remainder of this paper will discuss the basic algorithm used, the wave file format, the details of the algorithm implementation used by this author and the results found.

# CHAPTER II

# REVIEW OF THE HARMONIC GENERATOR ALGORITHM

The input to the harmonic generator algorithm is a discrete time varying signal x(i). A specific number of points (L) are extracted from the file and an L point Fast Fourier Transform is performed on the extracted points. The resultant frequency domain waveform is examined for peaks. Once the frequency peak ($f_{peak}$) has been detected, the phase angle of this peak ($P_{phase}$) is determined. The frequency peak is shifted by (N-1)*$f_{peak}$ where N is the desired harmonic. All shifted points greater than L/2 are discarded due to the characteristics of the Inverse Fast Fourier Transform (IFFT) step which will be applied later. The phase angle of the points about the shifted peak are then rotated by (N-1)* $P_{phase}$ radians. The Inverse Fast Fourier Transform is then performed on the shifted data. The new time domain signal has a user specified number of points (K) extracted from the center of the waveform. This step is used to minimize the error seen by the algorithm since it has been determined that the points in the middle of the output sample have the least amount of error when compared to theoretical data [4]. These K points then become the resultant output of the algorithm. The process is repeated for all points in the original waveform, shifting the L point window to the right, K points at a time.

# CHAPTER III

# THE WAVEFORM AUDIO FILE FORMAT (WAVE)

This section describes the Waveform format, which is used to represent digitized sound. The Waveform format, herein after referred to as WAVE, consists of two primary sections, the header and the waveform data. The header consists of 'chunks' of data used to describe the wave file. The complete header is called the format chunk and must always occur before the wave data chunk [1]. Both of these chunks are mandatory in a wave file. The wave file always starts with the ascii characters 'RIFF' encoded into 8-bit words for each letter. Next, the file size is encoded into the next four bytes of data. This is followed by the ascii characters 'WAVEfmt' encoded into the next seven bytes. Next is the wave format chunk of the header. The wave format chunk specifies the format of the wave data. The format chunk consists of the following fields [2]:

- Format category
- Number of channels
- Sampling rate
- Average bytes per second playback
- Data block size

The format category is a number indicating the WAVE format category of the file. This number controls how the waveform data is interpreted. This program was written to accept only Microsoft Pulse Code Modulation wave files as it appears to be the most common form seen since it is an open non-proprietary standard. The format category is set to one for the Microsoft PCM format. Other example wave format categories are the IBM mu-law format, the IBM a-law format and the IBM AVC Adaptive Differential Pulse Code Modulation format.

The number of channels chunk is used to define whether the wave file data is for a mono or stereo signal. The number of channels represented in the waveform data is a 1 for a mono file or a 2 for a stereo file.

The sampling rate chunk is formatted in samples per second. It indicates what rate the data was recorded and thus the rate at which each channel should be played.

The average bytes per second playback chunk provides an indication of how many bytes per second will be required to process during a given time period of one second. This is simply the sampling rate * the number of channels * the number of bits per channel (8 or 16). Playback software can estimate the buffer size using this value.

The data block size chunk indicates how many bytes of data per sample need to be processed. Playback software can use this value for buffer alignment.

Next, format specific data is provided in the header. Since we are using the Microsoft PCM format, the only format specific field is the Bits Per Sample chunk. The Bits Per Sample field specifies the number of bits of data used to represent each sample of each channel. If there are multiple channels, the sample size is the same for each channel.

Following the format specific data is the actual wave data representing the audio signal. In a single-channel wave file, samples are stored consecutively. For stereo wave files, channel 0 represents the left channel and channel 1 represents the right channel with the samples interleaved. The following diagrams show the data packing for an 8-bit mono and a 16-bit stereo wave file:

| | | | |
|---|---|---|---|
| Sample 1 | Sample 2 | Sample 3 | Sample 4 |
| Channel 0 | Channel 0 | Channel 0 | Channel 0 |

Data Interleaving for an 8-Bit Mono Wave File

| | | | |
|---|---|---|---|
| Channel 0 (left) | Channel 0 (left) | Channel 1 (right) | Channel 1 (right) |
| low-order byte | high-order byte | low-order byte | high-order byte |

Data Interleaving for a 16-Bit Stereo Wave File

This program was written to work with wave files consisting of either 8 or 16 bits of data per sample. In the wave file format, if more than one byte is required to describe the sample, then the least significant byte is stored first. The data format and maximum and minimums values for PCM waveform samples of various sizes are as follows:

Sample Size: 8 Bits

Data Format: Unsigned Integer

Maximum Value: 255 (0xFF)

Minimum Value: 0

Midpoint Value: 128 (0x80)

Sample Size: 16 Bits

Data Format: Signed Integer

Maximum Value: 32767 (0x7FFF)

Minimum Value: -32768 (-0x8000)

Midpoint Value: 0

Additional information can be encoded into the wave file format. This data, such as cue points, label and note information, has been ignored for the purpose of this project as it was not deemed necessary.

# CHAPTER IV

# PROGRAM DESCRIPTION

## 4.0 Program Design

The Envelope Tracking Harmonic Generator was written using National Instrument's LabVIEW graphical programming language. LabVIEW allows users to program in block diagram notation as will be seen in much greater detail in later sections as the program code is described. The program was written by developing smaller programs similar to subroutines. National Instruments calls these programs virtual instruments (VIs). By linking VIs together, larger programs can be developed. This facilitates re-use of code and simplifies software code testing since it can be accomplished at the VI level. VIs have three main parts: the front panel, the diagram and the icon. The front panel is the user interface. Since LabVIEW is a graphical programming language, it is perfectly suited for producing graphical user interfaces (GUIs). The main menu and all other user interfaces are front panels to VIs written or integrated during the development of this program.

The diagram is where the program functionality is defined. User selectable components are simply wired together in the order required for desired program functionality. Most of the components are LabVIEW primitives which are functions that are provided with the LabVIEW programming language package. VIs can be connected together in the diagram by wiring the VI icon into the program flow. The VI icon has defined connectors for passing data to and from the VI, much like a computer integrated circuit (IC) chip has interface pins on the outside of the IC package.

## 4.1 Main Menu

The main menu appears when the program is first started. It consists of nine operator selectable options and a loaded file text box (see figure 4.1.1).

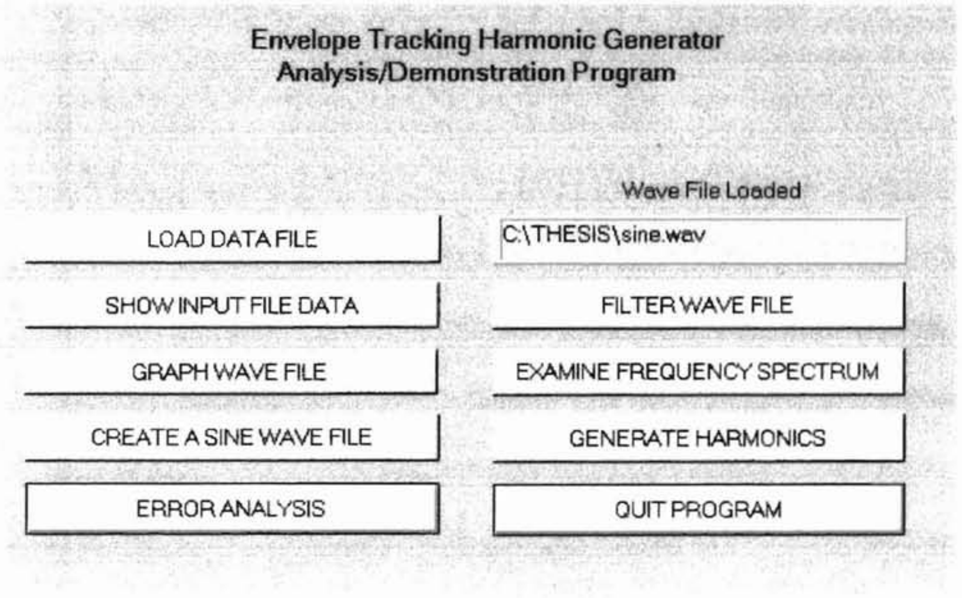


Figure 4.1.1: Program Main Menu

The user selectable options are:

<u>Load Data File</u>: Pressing this button will prompt the operator to select an existing WAVE file to load into the program memory for processing. The operator must select a file for processing before any signal processing can be accomplished. The Wave File Loaded indicator located to the right of the Load Data File button provides the name and the full directory path to the file loaded. <Not A Path> is indicated when no file is loaded.

<u>Show Input File Data</u>: Pressing this button will provide information about the WAVE file loaded such as sampling rate, bits per sample, number of channels, etc.

Graph Wave File: Pressing this button will provide a graph of the wave file data. This will appear as a time domain graph of the audio signal by user selectable left or right channel if stereo or by a single channel if mono.

Create A Sine Wave File: This selection allows the operator to create a single frequency sinusoid wave file. It can be used to create reference signals for examining the correctness of the harmonic generator algorithm.

Filter Wave File: Since wave files can cover the complete audio spectrum, the need to lowpass filter these files in order to test the audio performance was determined. Selecting this option allows the wave file to be lowpass, highpass, bandpass or bandstop filtered with the cutoff frequencies user selectable.

Examine Frequency Spectrum: Selecting this button will take the user to a screen where the time domain and frequency domain of the wave file can be examined. This is accomplished by selecting the number of samples to examine at a time and then performing a Fast Fourier Transform on the samples.

Generate Harmonics: This option opens the window that is the main focus for the program. This is the window that allows the user to select the sample size, desired harmonic to be generated and other specifics. This window is used to create new wave files using the envelope tracking harmonic generator algorithm.

Error Analysis: For use only with 8-bit, single sinusoid reference waveforms that have been processed with the Harmonic Generating algorithm. This function allows the results of the algorithm to be compared to a baseline and examined for error.

Quit Program: Selecting this button stops program execution and returns the user to the windows operating system.

The main menu VI diagram is shown in figure 4.1.2. This VI monitors the buttons located on the main menu for selection. When pressed, the appropriate VI is executed. Note that the OPEN VI is located outside of the Case Loop function. This means that when the program is first started, the user is asked to select a file to open. Once a file is selected, the functions located inside of the loop can be performed on the WAVE file. The functions are located in a sequence structure. The first sequence is 0 and the last is 7, providing 8 different functions, one per button, except for the "Quit Program" button on the main menu and one to update the Wave File Loaded indicator. The functions in the sequence are continuously monitored until the Load Data File button is selected. Once selected, the Case Loop is halted, the open file VI is called and a new wave file can be loaded into program memory.

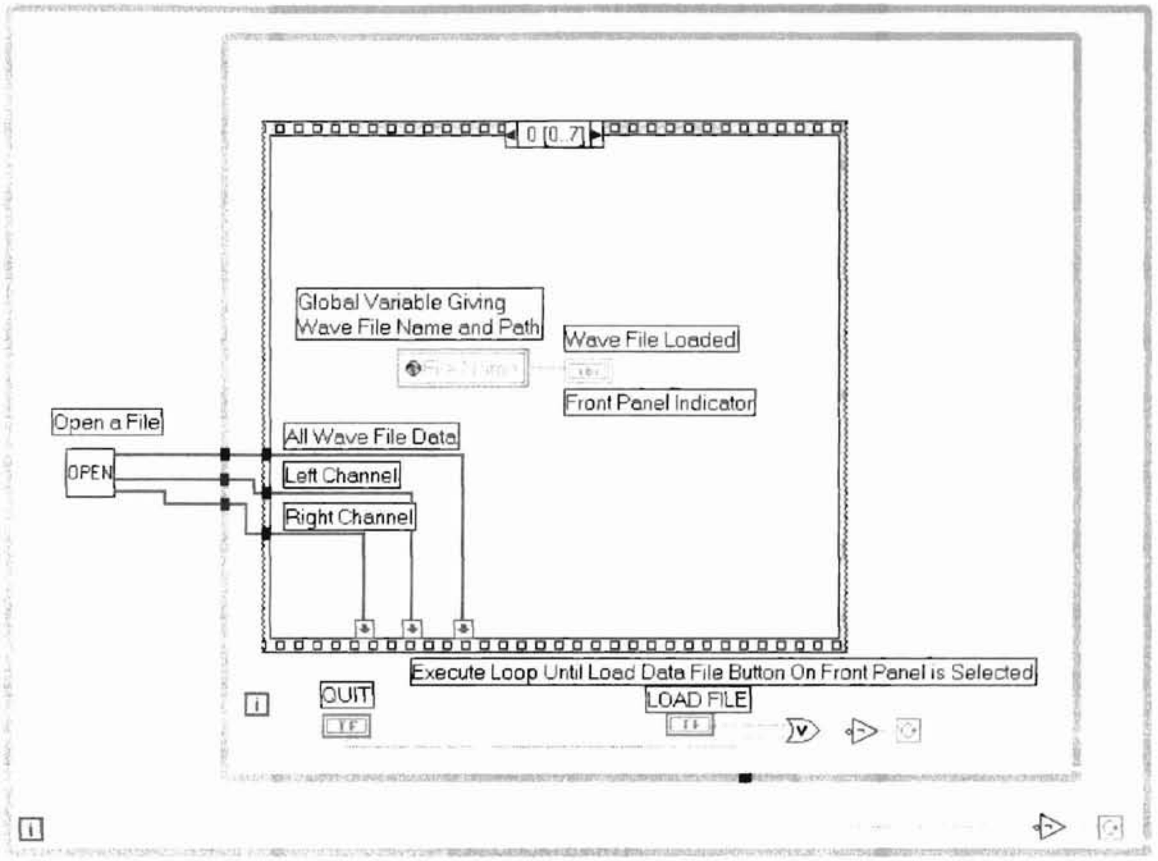


Figure 4.1.2: Main Menu Diagram Showing Sequence 0

Sequence 0 of the sequence structure updates the front panel of the main menu with the file name and path of the wave file loaded into system memory. Sequences 1 through 7 are shown in figures 4.1.3 through 4.1.9. A description of each sequence is not provided since the diagram figures are self explanatory.

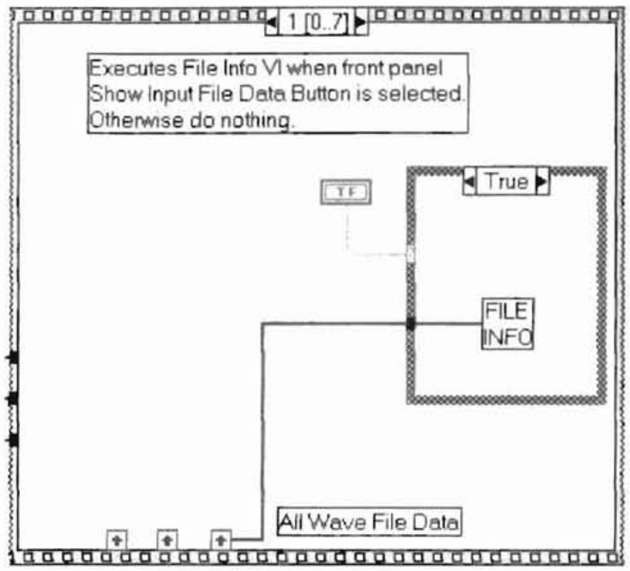

1 [0..7]
Executes File Info VI when front panel
Show Input File Data Button is selected.
Otherwise do nothing.
TF
True
FILE INFO
All Wave File Data

Figure 4.1.3: Main Menu Diagram Sequence 1

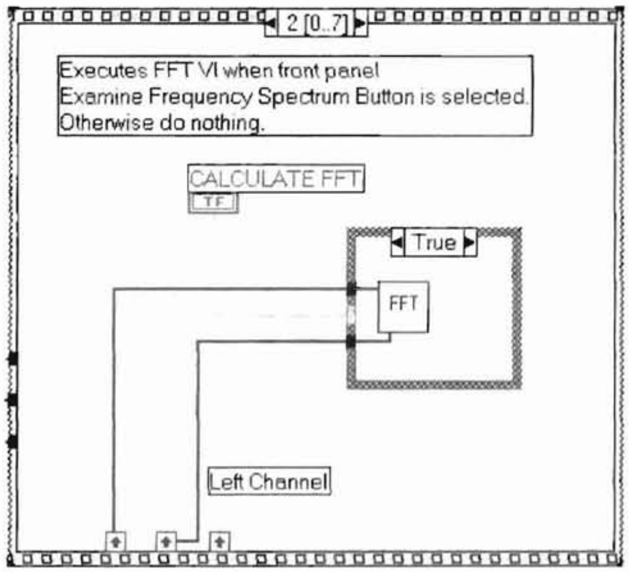

2 [0..7]
Executes FFT VI when front panel
Examine Frequency Spectrum Button is selected.
Otherwise do nothing.
CALCULATE FFT
TF
True
FFT
Left Channel

Figure 4.1.4: Main Menu Diagram Sequence 2

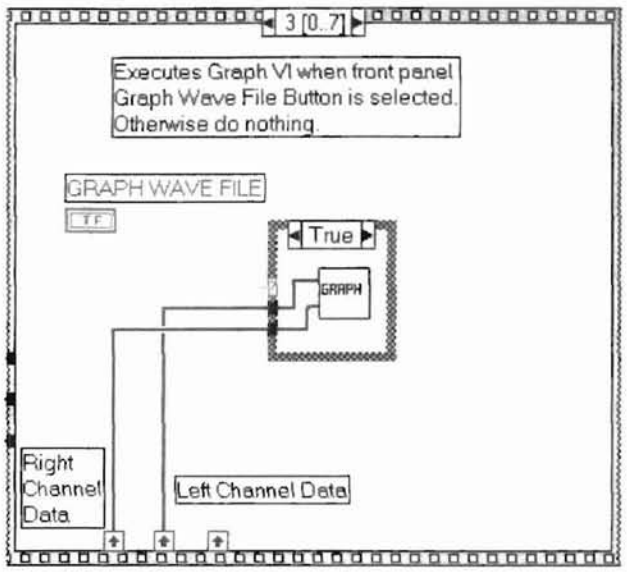

3 [0..7]
Executes Graph VI when front panel Graph Wave File Button is selected. Otherwise do nothing.
GRAPH WAVE FILE
TF
True
GRAPH
Right Channel Data
Left Channel Data

Figure 4.1.5: Main Menu Diagram Sequence 3

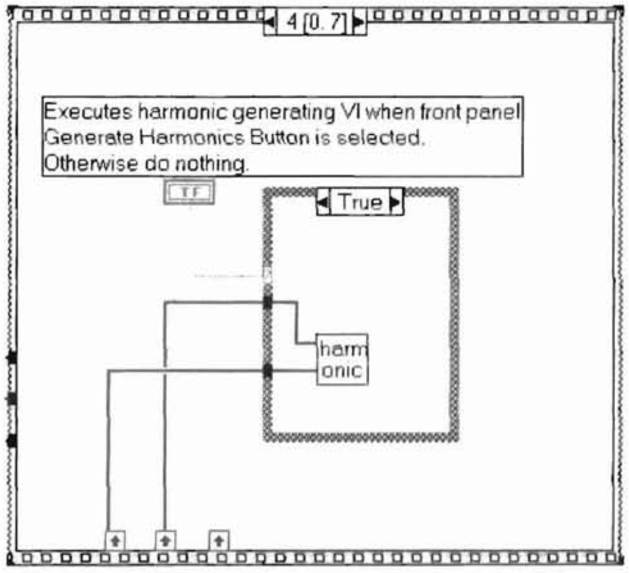
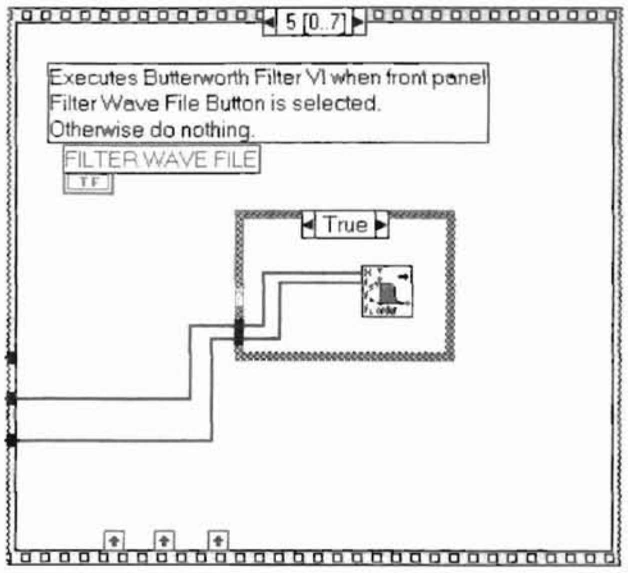
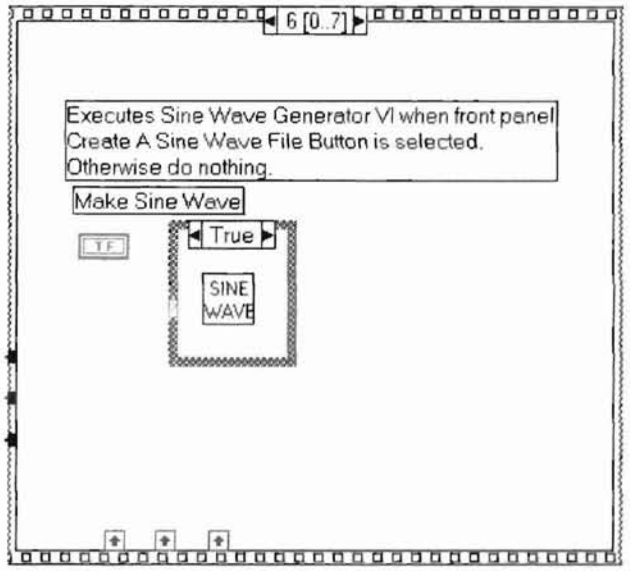

4 [0..7]
Executes harmonic generating VI when front panel Generate Harmonics Button is selected. Otherwise do nothing.
TF
True
harmonic

Figure 4.1.6: Main Menu Diagram Sequence 4



Executes Butterworth Filter VI when front panel
Filter Wave File Button is selected.
Otherwise do nothing.

Figure 4.1.7: Main Menu Diagram Sequence 5



Executes Sine Wave Generator VI when front panel
Create A Sine Wave File Button is selected.
Otherwise do nothing.

Figure 4.1.8: Main Menu Diagram Sequence 6

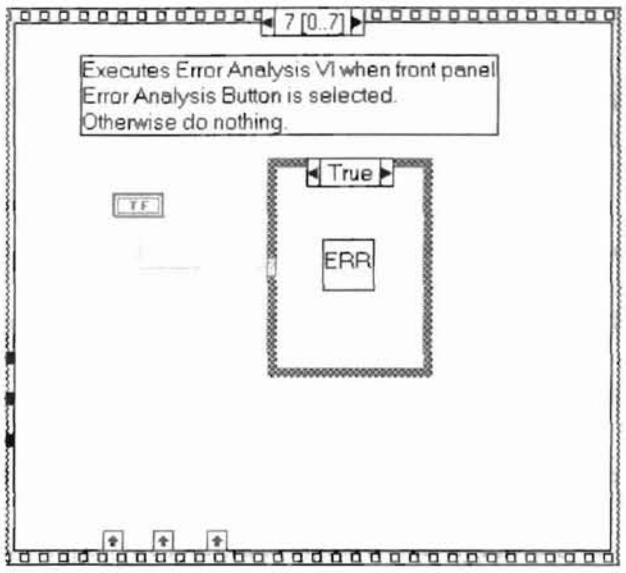

7 [0..7]
Executes Error Analysis VI when front panel Error Analysis Button is selected. Otherwise do nothing.
True
TF
ERR

Figure 4.1.9: Main Menu Diagram Sequence 7

## 4.2 Open VI Description

The Open VI is called when the program is first started and when the Load Data File button on the main menu is selected. The function of the Open VI is to extract data from the wave file for use in the processing of the wave file. Many of the parameters extracted are placed into global variables that can be used by all of the VIs in the system. These global variables typically have a single value, but a few consist of an array of values. The complete wave file is available as an output of this VI as well as left and right channel data. The Open VI Diagram is shown in figure 4.2.1. The VIs called within the Open VI are described in Appendix A of this document. The Open VI does not have an accessible graphical user interface.

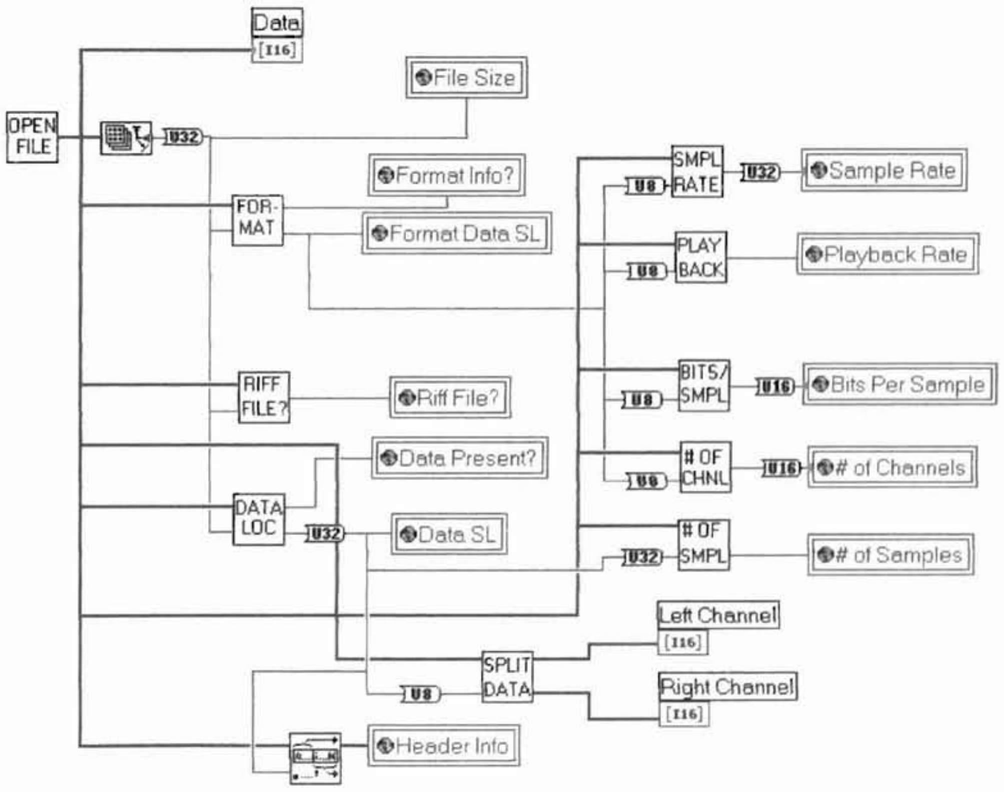
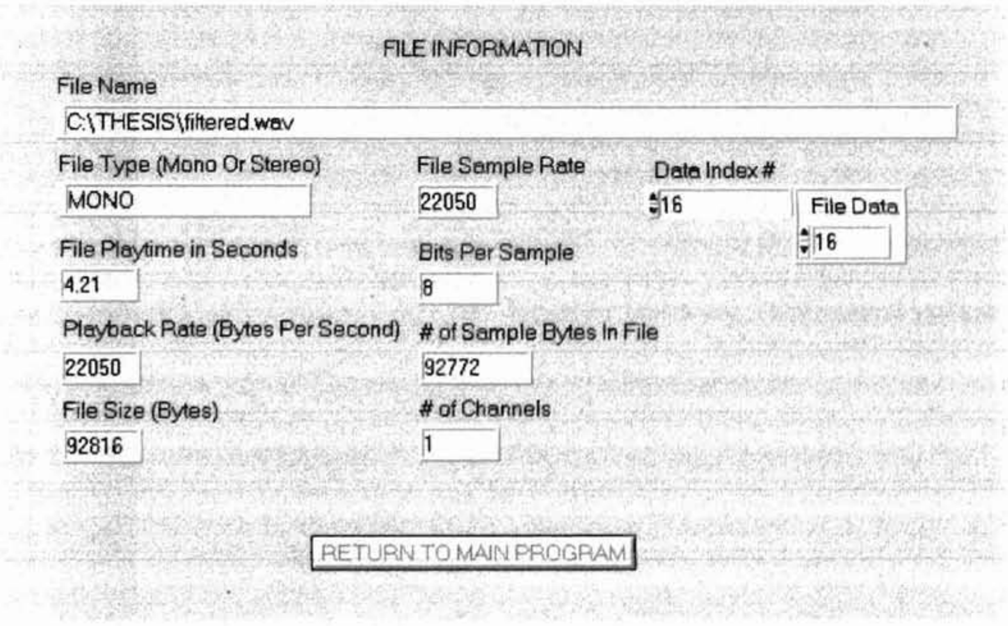


Figure 4.2.1: Open File VI Diagram

## 4.3 File Info VI Description

The File Info VI is called when the Show Input File Data Button is selected on the Main menu. This VI provides a user screen that provides format information about the loaded wave file (see figure 4.3.1). There are no operator interactions with this front panel other than to be able to scroll through the wave file data eight bits at a time and to select the return to the main menu button. This VI examines the RIFF FILE? global variable output of the Open VI to determine if a valid wave file was selected. If an invalid wave file is selected then a false condition exists and the corresponding false case structure is executed (see figure 4.3.2). The operator is returned to the main menu once the OK button is selected on the pop-up dialog box.


FILE INFORMATION

File Name: C:\THESIS\filtered.wav

File Type (Mono Or Stereo): MONO

File Sample Rate: 22050

Data Index #: 16

File Data: 16

File Playtime in Seconds: 4.21

Bits Per Sample: 8

Playback Rate (Bytes Per Second): 22050

# of Sample Bytes In File: 92772

File Size (Bytes): 92816

# of Channels: 1

Figure 4.3.1: File Info VI Front Panel

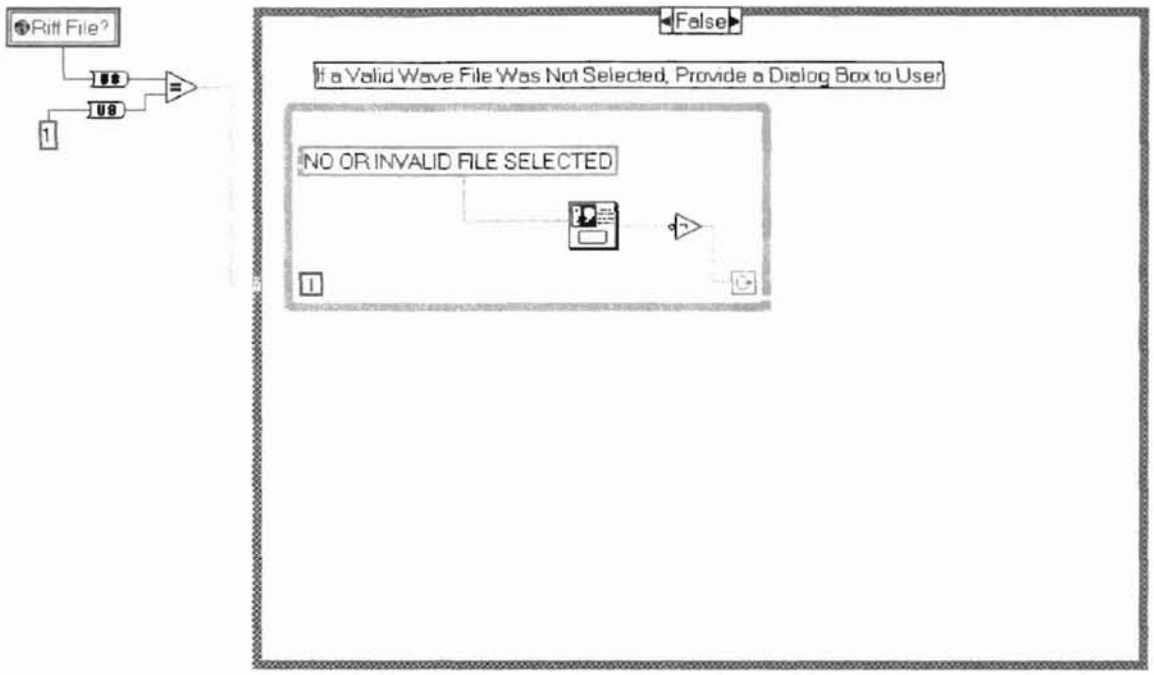

Riff File?
False
If a Valid Wave File Was Not Selected, Provide a Dialog Box to User
NO OR INVALID FILE SELECTED

Figure 4.3.2: File Info VI False Case Structure Diagram

When the false case structure is executed the operator is prompted with a message window that indicates that either no file has been loaded or that the loaded file does not contain valid wave file information. In the case that a valid file is loaded, the true case structure is executed (see figure 4.3.3). The true case takes the information loaded into global variables by the Open VI and displays the information on the front panel of the File Info VI. Two calculations are performed by the true case structure. The file playtime duration is calculated by dividing the number of wave file samples by the playback rate and the number of channels is examined to provide a stereo or mono indication on the front panel. The operator is returned to the main menu when the Return to Main Program button is selected.

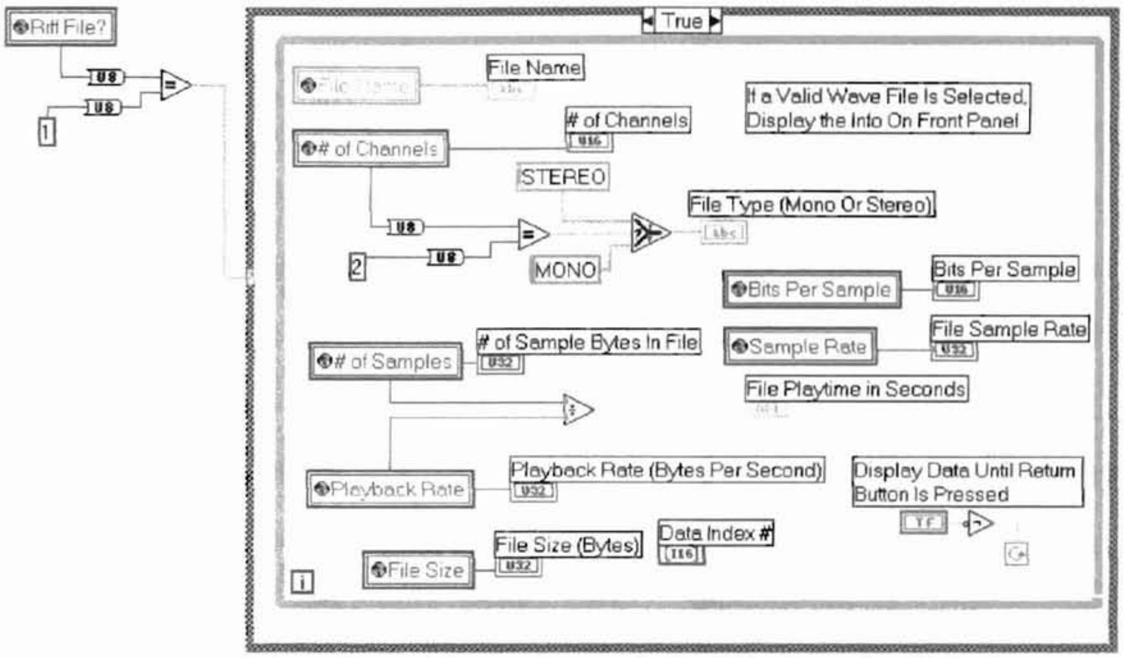

Riff File?
True
File Name
If a Valid Wave File Is Selected, Display the Info On Front Panel
# of Channels
STEREO
File Type (Mono Or Stereo)
MONO
Bits Per Sample
File Sample Rate
Sample Rate
# of Samples
# of Sample Bytes In File
File Playtime in Seconds
Playback Rate
Playback Rate (Bytes Per Second)
Display Data Until Return Button Is Pressed
File Size
File Size (Bytes)
Data Index #

Figure 4.3.3: File Info VI True Case Structure Diagram

## 4.4 FFT VI Description

The FFT VI is executed when the Examine Frequency Spectrum button on the main menu is selected. The front panel (see figure 4.4.1) provides a graphical display of the original waveform data in both the time and frequency domain. The operator must select the number of wave file samples to process in a single FFT pass. The wave file data is divided by the number of samples to process and an indication of the total number of times the FFT process will occur is indicated in the # of Passes display in the upper left hand corner of the menu. Once the number of samples is selected, the user selects the Press Here To Perform FFT button and the wave file frequency spectrum is determined in blocks of data determined by the number of samples selected by the operator. The FFT of each block of data is sequentially performed until the end of the file is reached. The diagram of the FFT VI is shown in figure 4.4.2

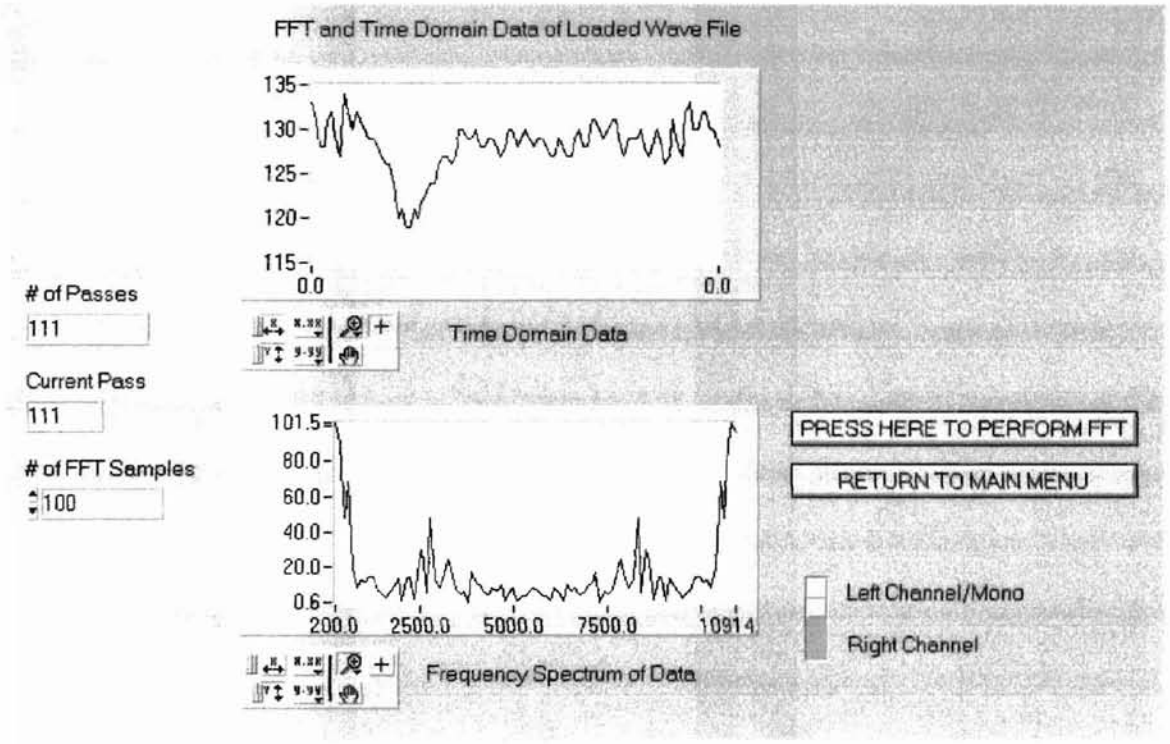


Figure 4.4.1: FFT VI Front Panel

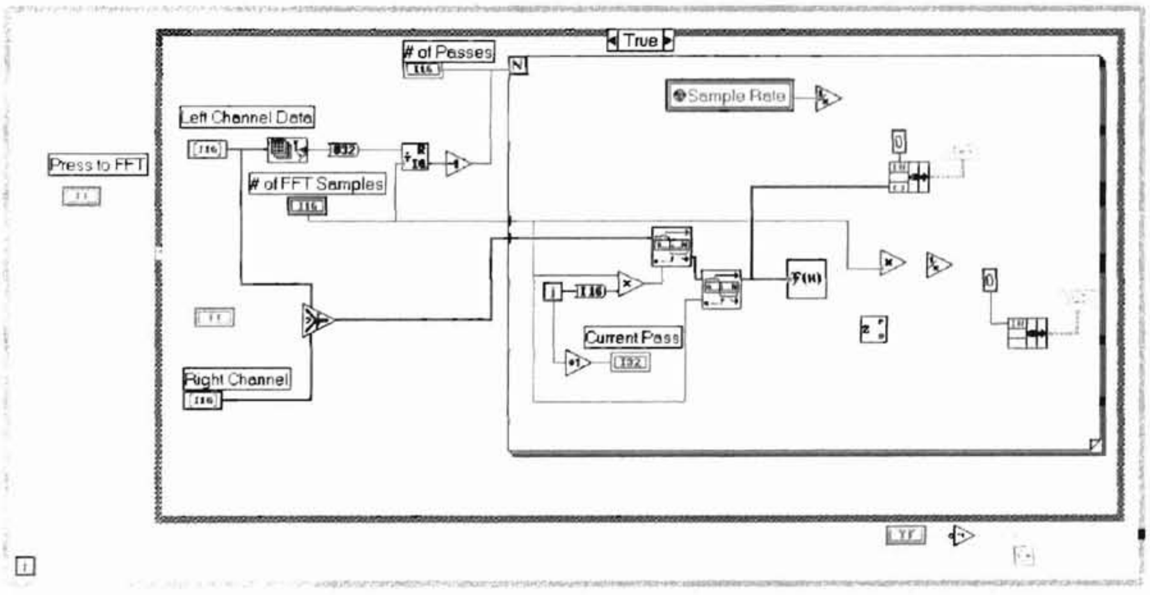


Figure 4.4.2: FFT VI Diagram

## 4.5 Graph Wavefile VI Description

The Graph Wavefile VI is called when the Graph Wave File button on the main menu is selected. This VI simply takes the wave file data provided by the Open VI and displays the data using a graph window. The graph is shown by sample number versus sample magnitude. The left and right channel data magnitude can be examined point by point through the use of two array indicators located in the lower right hand corner of the user screen. A sample graph is shown in figure 4.5.1 and the diagram is shown in figure 4.5.2.

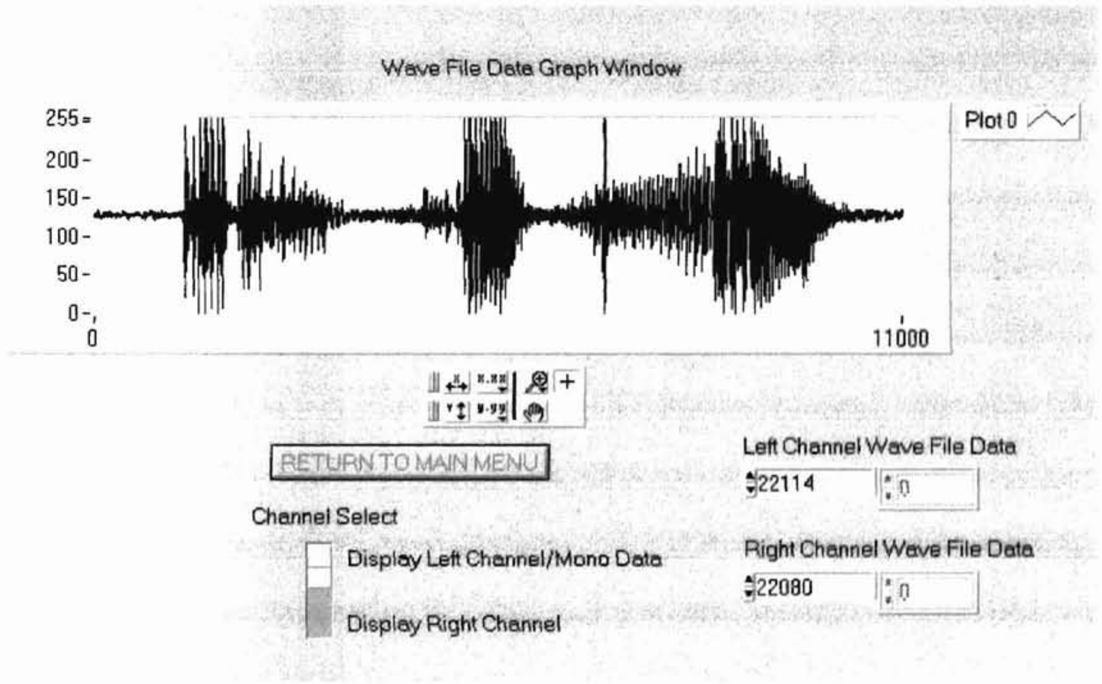

Wave File Data Graph Window
255
200
150
100
50
0
0
11000
Plot 0
RETURN TO MAIN MENU
Channel Select
Display Left Channel/Mono Data
Display Right Channel
Left Channel Wave File Data
22114
0
Right Channel Wave File Data
22080
0

Figure 4.5.1: Graph Wavefile VI Front Panel

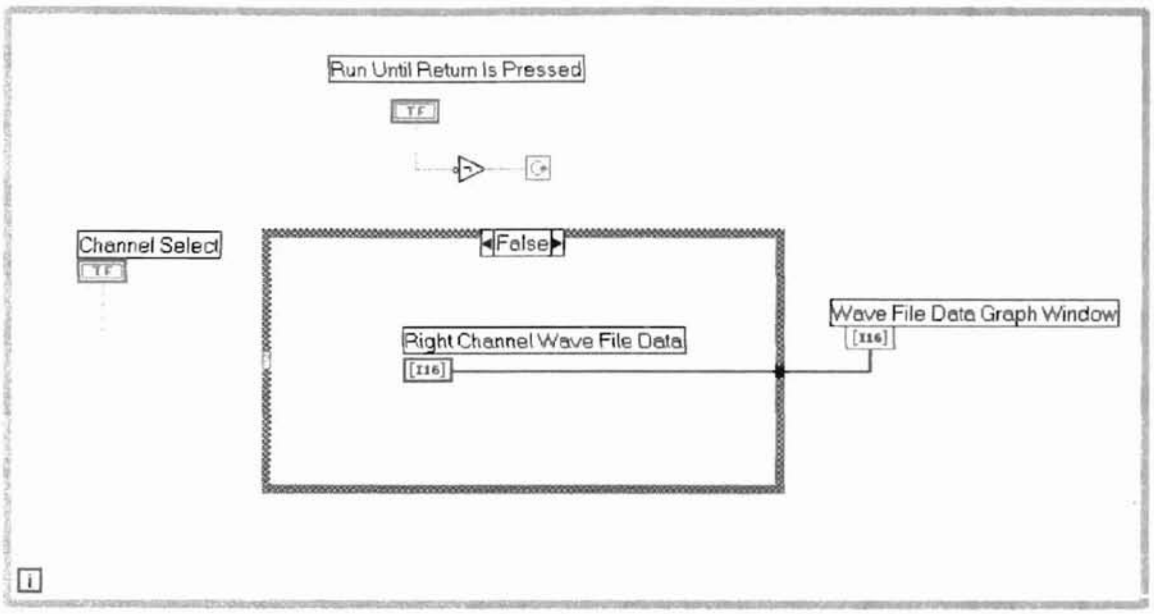

Run Until Return Is Pressed
TF
Channel Select
TF
False
Right Channel Wave File Data
[I16]
Wave File Data Graph Window
[I16]

Figure 4.5.2: Graph Wavefile VI Diagram

## 4.6 Harmonic VI Description

This VI is called when the Generate Harmonics button on the Main menu is selected. This VI contains the harmonic generator algorithm functionality and will be discussed in great detail in this section.

### 4.6.1 Harmonic VI Front Panel

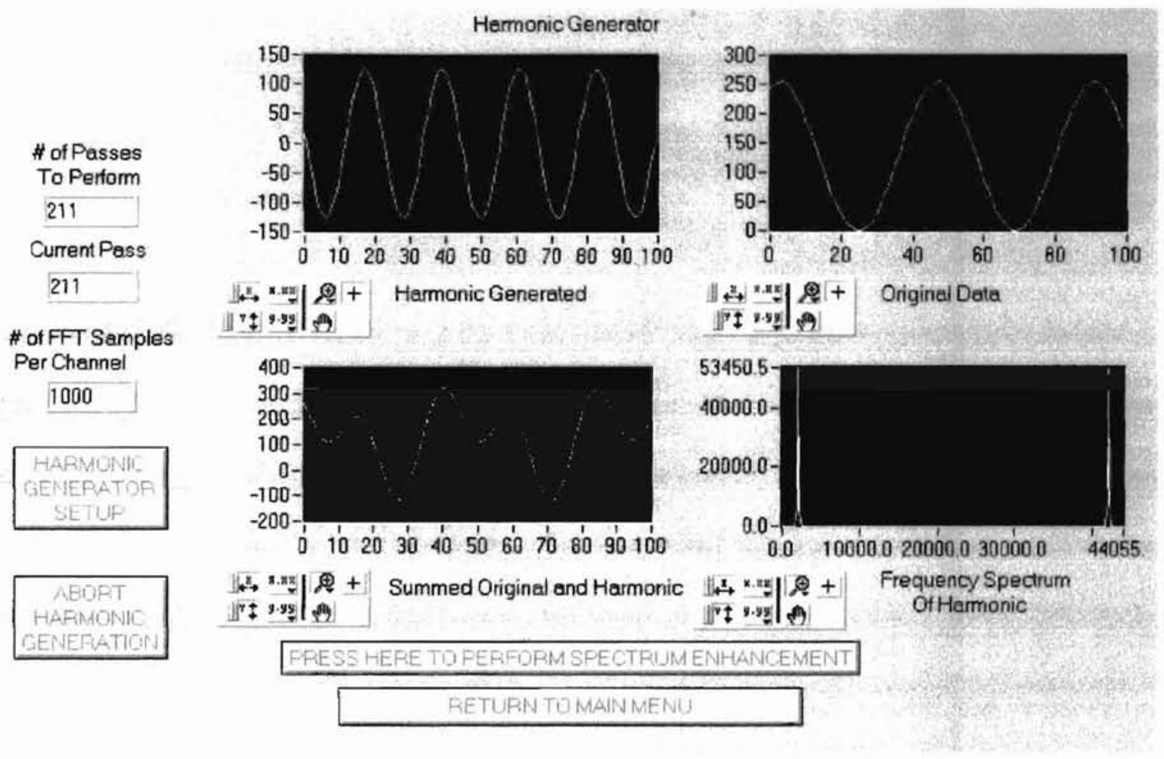
The Harmonic VI front panel, see figure 4.6.1, provides four separate graph windows for side-by-side examination of the different steps performed by the harmonic generator algorithm. A detailed description of each window is provided:

Original Data Window: This graph is located in the upper right hand side of the panel and shows a time domain slice of the original waveform before processing. The x-axis is sample number and the y-axis is the sample value. The example shown is a graph of a 1KHz signal. This window is updated for every block of samples examined as determined by the # of FFT Samples control of the Harmonic Generator Setup button on left hand side of the panel.

Frequency Spectrum Harmonic Window: This graph is located in the lower right hand side of the panel. This graph shows the frequency domain of the shifted waveform data for the current sample. The data has been shifted N-1*$f_{peak}$ bins to the right where N is the desired harmonic as provided by the Harmonic Generator Setup control located on the left hand side of the panel. The x-axis of this graph shows the shifted data bins with respect to frequency. The y-axis is the magnitude of the complex data generated by the FFT algorithm.

Harmonic Generated Window: This window displays the time domain signal generated by performing the Inverse Fast Fourier Transform (IFFT) on the shifted frequency peak data. The data x-axis range is determined by the # of samples to keep control in the Harmonic Generator Setup panel. These samples have been taken from the center of the generated waveform since it has been shown that this is where errors in the harmonic signal are minimal [4]. The magnitude of the y-axis is controlled by the actual value of the signal peaks calculated by the IFFT algorithm which is multiplied by the Scaling Factor control of the Harmonic Generator Setup panel. In this example, the scaling factor has been set to 1.

Summed Original and Harmonic Window: This window is the numerical result of adding the two windows, Harmonic Generated and Original Data, together. Only the current processed data snapshot of values determined by the # of Samples to Keep control are visible.

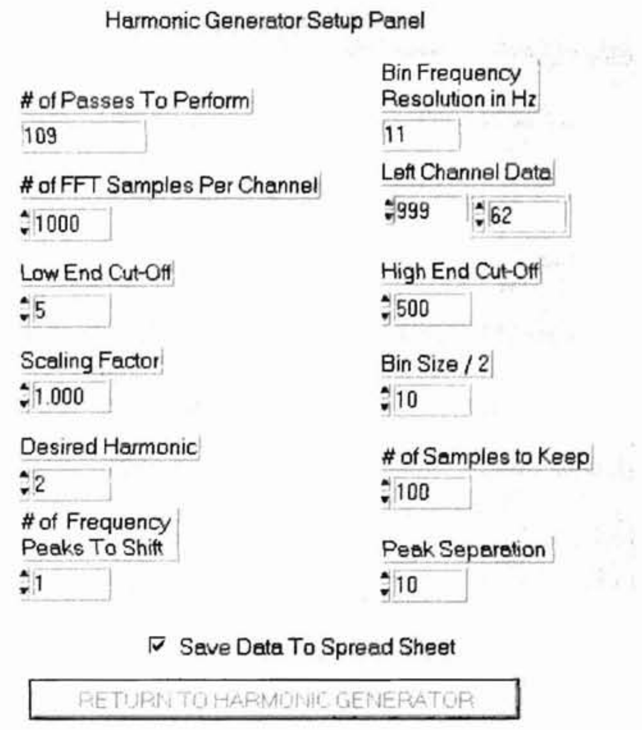

Harmonic Generator
Harmonic Generated
Original Data
Summed Original and Harmonic
Frequency Spectrum Of Harmonic
HARMONIC GENERATOR SETUP
ABORT HARMONIC GENERATION
PRESS HERE TO PERFORM SPECTRUM ENHANCEMENT
RETURN TO MAIN MENU

# of Passes To Perform: 211

Current Pass: 211

# of FFT Samples Per Channel: 1000

Figure 4.6.1: Harmonic Generator VI Front Panel



| | |
|---|---|
| # of Passes To Perform | 109 |
| # of FFT Samples Per Channel | 1000 |
| Low End Cut-Off | 5 |
| Scaling Factor | 1.000 |
| Desired Harmonic | 2 |
| # of Frequency Peaks To Shift | 1 |
| Bin Frequency Resolution in Hz | 11 |
| Left Channel Data | 999 62 |
| High End Cut-Off | 500 |
| Bin Size / 2 | 10 |
| # of Samples to Keep | 100 |
| Peak Separation | 10 |

[✓] Save Data To Spread Sheet

Figure 4.6.2: Harmonic Generator Setup Panel

In addition to the four graphical displays on the front panel, there is the Harmonic Generator Setup control button, the Abort Harmonic Generation button and three indicators. The # of Passes to Perform indicator shows how many FFT passes per channel must be performed on the wave file. This number is directly affected by the file size and the number of samples kept per pass which is set using the Harmonic Generator Setup panel. The current pass indicator provides a status to show the user which pass is currently being performed. If the Abort Harmonic Generation button is pressed when the Harmonic Generating Algorithm is being processed, the current pass indicator will be incremented to the end of the file and the operator will be prompted to save the processed file. Only the data processed prior to the point of when the Abort button was selected will be written to the file. When the Harmonic Generator Setup button is selected, the setup panel of figure 4.6.2 appears on the screen and has the following controls and indicators.

Controls

# of FFT Samples Per Channel: User defines the number of samples, N, to perform an N-Point FFT during the algorithm processing.

Low End Cut-Off: This control allows the user to ignore data below this bin number during the execution of the peak search algorithm. Since wave files have a natural DC offset during moments of silence, there can be a substantial DC peak which will be automatically ignored by this program. The Low End Cut-Off can be associated to a specific frequency using the following equation:

Low end cutoff frequency = (sampling rate/# of FFT samples per channel)*Low End Cut-off Value

Scaling Factor: This control allows the user to determine how much to scale the harmonic signal before being added to the original waveform. In many cases, the IFFT values are fairly small in comparison to the original signal due to the number of bins being shifted together, the number of FFT samples used, etc. In addition, if the harmonic signals generated are too large, they can overwhelm the listener with harmonics that are not subtle enough. In this case, making the scaling factor a smaller number would help.

Desired Harmonic: User defines the desired harmonic. Must be an integer value greater than 1.

# of Frequency Peaks to Shift: Used to allow the operator to select the number of different frequency peaks to shift per pass. For example, if the operator wants to shift two discrete peaks per pass, the algorithm will determine where the greatest peak occurs in the wave file data being examined. The program will then exclude that peak and the number of bins above and below that point as specified in the Peak Separation Control. The program will then examine for a new peak to include for shifting in the algorithm, excluding the first peak and surrounding values.

Left Channel Data: Allows the operator to scroll through the left channel of a stereo wave file data provided to the Harmonic Generator Algorithm. This control shows all wave file data for a mono wave file.

High End Cut-Off: This control allows the user to ignore data past this bin number during execution of the peak search algorithm. This was added to allow the operator to omit higher frequency signals if their harmonics were deemed undesirable. This value can be associated to a specific frequency using the following equation:

High end cutoff frequency = (sampling rate/# of FFT samples per channel)*High End Cut-off Value

Bin Size / 2: This control allows the user to determine how many adjacent data bins to move with the peak. Increasing the Bin Size can increase the spectral content of the harmonic signal generated. The value in the control must be multiplied by a factor of two and adding one in order to determine how many bins are actually being shifted together. This was done in order to eliminate the possibility of an odd number of adjacent bins being shifted.

# of Samples to Keep: Used to define the number of samples to keep from the IFFT processed waveform. This waveform, which is the harmonic signal, consists of a number of discrete samples which is equivalent to the # of FFT Samples defined earlier. Since these samples have larger amounts of error at both ends of the sample range than those in the middle, selecting a smaller number here than is found in the # of FFT Samples helps minimize the amount of error seen in the algorithm. This control also sets the number of times that the algorithm must be performed on the wave file. This forces the program to keep a certain number of samples, K, per pass. This creates a sliding window of data processing, where the window is incremented K points deeper into the

file at the end of each pass. The # of Passes to Perform equals the number of samples per channel divided by K.

Peak Separation: Used to define how many bins to each side of a peak are ignored in subsequent peak calculations when the # of Frequency Peaks to Shift control is larger than 1.

Save Data to Spread Sheet: Marking this box will save the first pass results of the harmonic generating algorithm to a tab delimited file. The data found in this file is the frequency spectrum data before and after the harmonic generating algorithm was applied. It was used to provide the test data of appendix C.

Indicators

# of Passes To Perform: This indicates the number of passes that the harmonic generating algorithm must perform in order to process the whole file. The same indicator is provided on the Harmonic Generator VI front panel

Bin Frequency Resolution in Hz: This provides the user with the frequency between bins. This is affected by sampling rate and the number of FFT samples. It is calculated using the following equation and can be helpful in determining the Low and High End Cut-off Values: Bin Resolution=(Sampling Rate/# of FFT Samples)

## 4.6.2 Harmonic Generator Diagrams

The Harmonic Generator diagrams have been broken down into functional sequence structures. The major functions in this VI are: FFT performance on the left and right channels, frequency peak calculations, bin shifting, saving data in spread sheet format, IFFT performance on the shifted left and right channel data, summation of the harmonic and original data and  creation of the modified wave file. The first sequence is shown in figure 4.6.3. This sequence calculates the number of FFT passes that will have to be performed in order to process all of the data in a single channel. It also ensures that the number of FFT passes does not go past the end of the file due to the overlapping nature of the algorithm with only the center portion of the waveform kept for use. Since the left channel will always have data, it is used for this calculation. The calculation is performed by determining the maximum number of passes to be performed by dividing the channel array size by the number of samples to keep. For example, if the wave file left channel contains 1000 bytes of data and we are to keep 50 samples from each round of calculations, then there would have to be a maximum of 1000/50=20 passes to perform. If we were performing a 50 point FFT initially, then 20 passes would be required. But, if we desire to perform a 100 point FFT, then 20 passes would cause the last FFT to have data that goes past the end of the data file. Because of this, a second calculation is performed to reduce the number of passes performed to prevent the FFT from containing invalid data. This calculation takes the FFT sample size and the number of samples to keep, dividing each number by two, and subtracting the two numbers from each other. The numbers are divided by two because only half of the data sample size can overlap for each pass past the end of the wave file. With the difference calculated, then we calculate the number of times an FFT would be performed on data past the end of the wave file by dividing the difference by the half of the number of samples to keep. This number is then subtracted from the original maximum number of FFT passes required. This value is the number of passes that will be performed by the harmonic generation algorithm per channel.

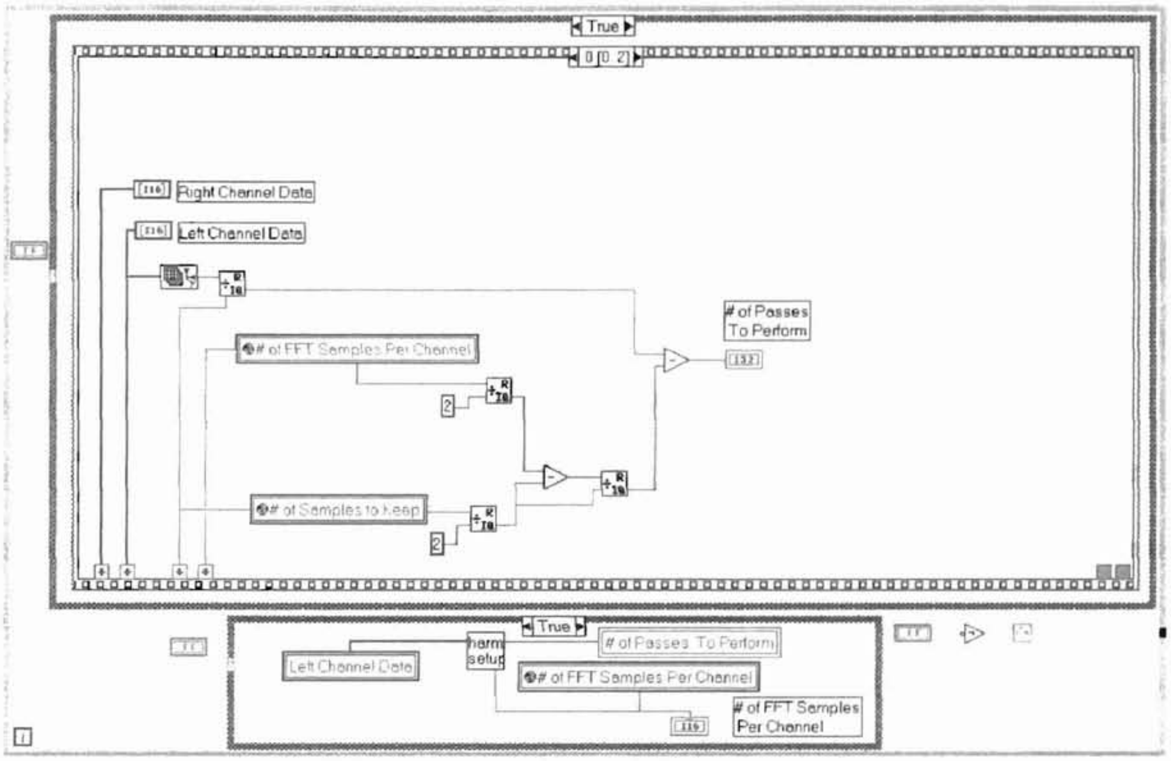


Figure 4.6.3: Harmonic Generator VI Sequence 0 Diagram

4.6.2.1 Sequence 1, Sub-sequence 0 and 1 Diagrams

Sequence 1, see figure 4.6.4, is performed after sequence 0. A For Loop is executed within sequence 1 the number of times calculated by sequence 0 mentioned above. Shown on the left hand side of sequence 1 is the method for generating the sliding window of data that will be processed by the harmonic generating algorithm. The user selectable number of FFT samples per channel provides that number of points at a time to the FFT algorithm seen in sub-sequence 0. After each pass of processing the desired number of FFT samples, the index point of the array supplied to the algorithm is incremented by the number value of the Samples to Keep control of the Harmonic Generator Setup panel. This causes the window to increment towards the end of the file by that number until the number of passes to perform has been met. For example, if we have a file size of 2000 samples, we could perform a 100 point FFT during each pass, initially looking at samples 1 through 100. If we desire to keep only 10 samples from the resultant processed file, we would slide the window by 10 samples during the next FFT process, looking at samples 10 through 110. We would be required to perform a maximum of 2000/10=200 passes, executing a 100 point FFT each time. Sequence 1 contains a subsequence of 12 frames. Subsequences 0 and 1, shown in figures 4.6.4 and 4.6.5 take the shifted data for the left and right channels and calculates the N-point FFT of each channel, where N equals the number of FFT samples per channel as set using the Harmonic Generator Setup panel.

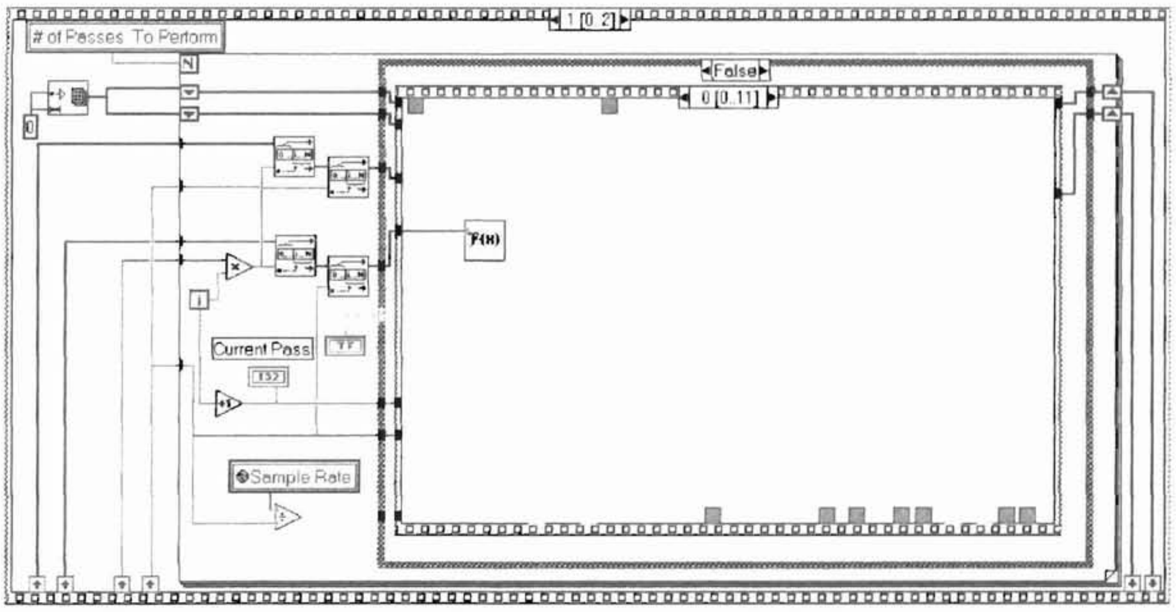


Figure 4.6.4: Harmonic Generator VI Sequence 1, Sub-sequence 0 Diagram

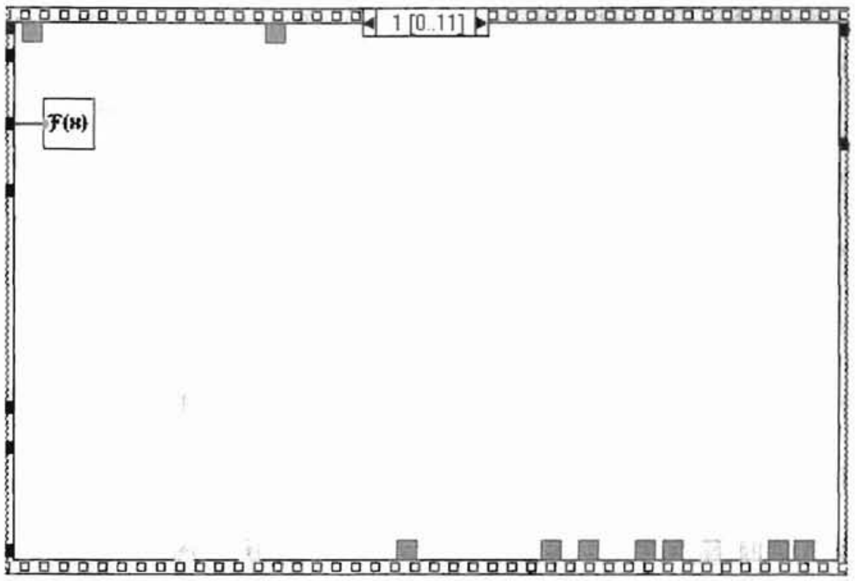


Figure 4.6.5: Harmonic Generator VI Sequence 1, Sub-sequence 1 Diagram

4.6.2.2 Sub-sequence 2 and 3 Diagrams

Sub-sequences 2 and 3 are shown in figure 4.6.6 and 4.6.7. These sequences are where the frequency peaks of the data are determined. Subsequence 2 is for the left channel data and subsequence 3 is for right channel data. Each use the Peak Find VI described in detail in Appendix A. First, the data from the FFT is converted from rectangular to polar mode. The data is then split based on the desired harmonic. For example, if the harmonic desired is the second, then the data is split in half, since only the lower half of the data can be examined for peaks. The reason for this is that a peak in the upper half of the data range cannot be shifted up to the desired second harmonic because it would be placed outside of the Nyquist frequency range. Once the data has been split, the data is cut down to a smaller sampling by eliminating the bins below the Low End and above the High End cut-off values as set by the Harmonic Generator Setup panel. With the data reduced down to the working set, a peak finder algorithm processes the remaining data and provides the index of the peak value. Based on the harmonic selected, N, the # of Bins to Shift is calculated using the following equation:

# of Bins to Shift = (N-1)*(Peak Index + Low End Cut-Off + 1)

The value of (Peak Index + Low End Cut-Off + 1) is equivalent to $f_{peak}$ as described on page 3. The algorithm used calculates the peak index after the low end cut-off has been applied to the data. This means that the peak index has been offset by the low end cut-off value. Also, since the first value in a LabVIEW array is referenced as index 0, we must account for this by adding one back into the equation for the # of Bins to Shift. The process is repeated for the number of frequency peaks that are desired for shifting as provided by the user in the Harmonic Generator Setup panel, ignoring data around the first and subsequent peaks set by the peak separation control.

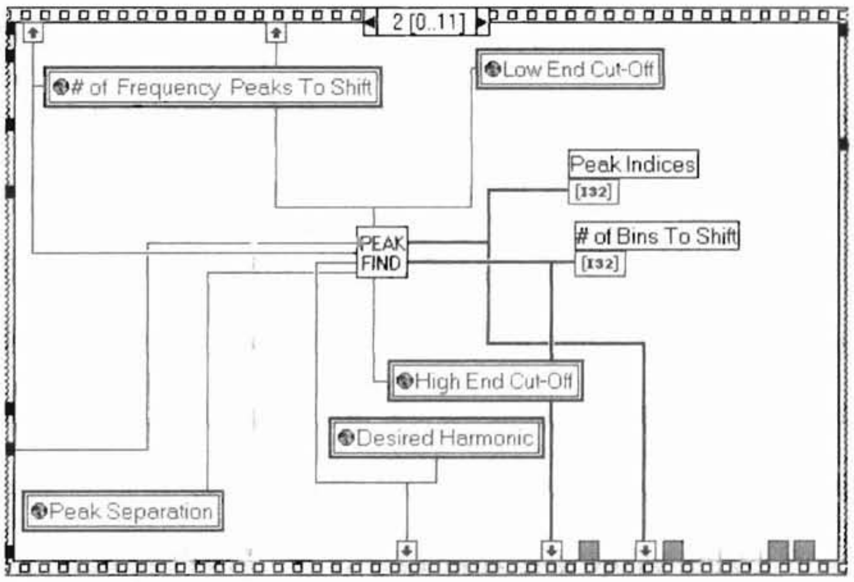


Figure 4.6.6: Harmonic Generator VI Sequence 1, Sub-sequence 2 Diagram

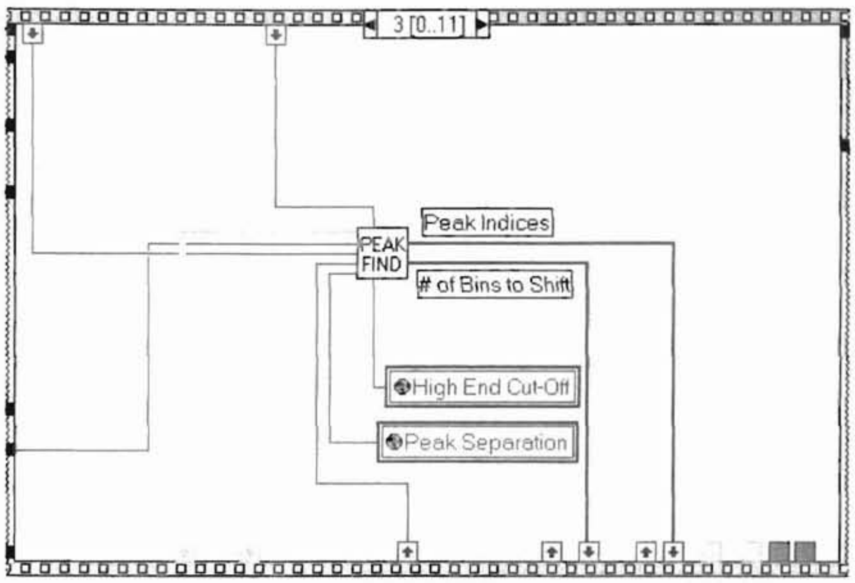


Figure 4.6.7: Harmonic Generator VI Sequence 1, Sub-sequence 3 Diagram

4.6.2.3 Sub-sequence 4 and 5 Diagrams

These sequences, see figures 4.6.8 and 4.6.9, are used to shift the FFT data to the right by the Number of Bins to Shift result from the previous sequence. These sequences use the Shift VI described in Appendix A. The Shift VI determines and rotates the individual phases of each frequency bin by $(N-1)*P_{phase}$ where N is the desired harmonic and $P_{phase}$ is the phase of the Peak bin data. Each bin shifted with a associated peak has its phase rotated by this peak's phase per this equation. The number of bins shifted with a particular peak is set by the Bin Size/2 control in the Harmonic Generator Setup panel. If the value for this set set to 10, that means that 21 bins will be shifted, 10 for the bins below the peak, 10 for the bins above the peak and one for the peak itself. All 21 bins will be shifted up in frequency as determined by the # of Bins to Shift calculated in the previous sequence. Lastly, this sequence pads the new array created with all zeroes except for the desired shifted frequency peaks and adjacent bins.

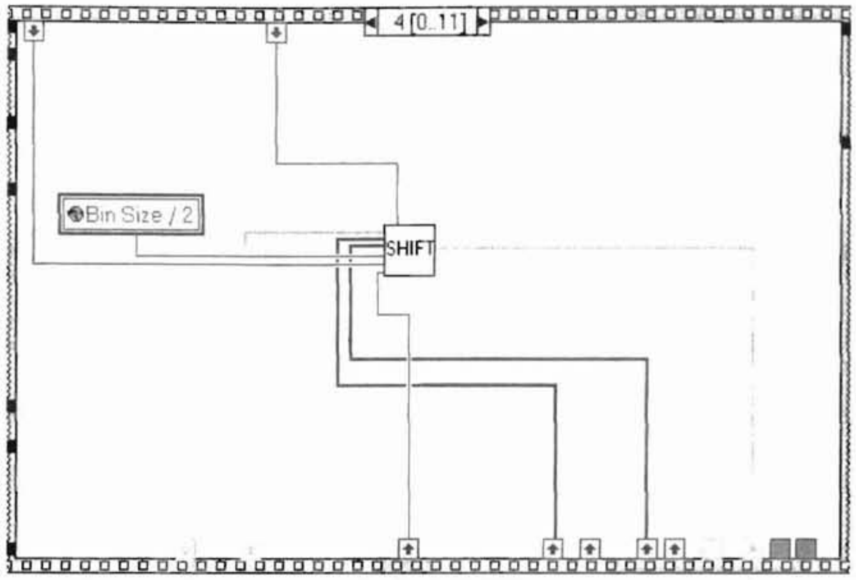


Figure 4.6.8: Harmonic Generator VI Sequence 1, Sub-sequence 4 Diagram

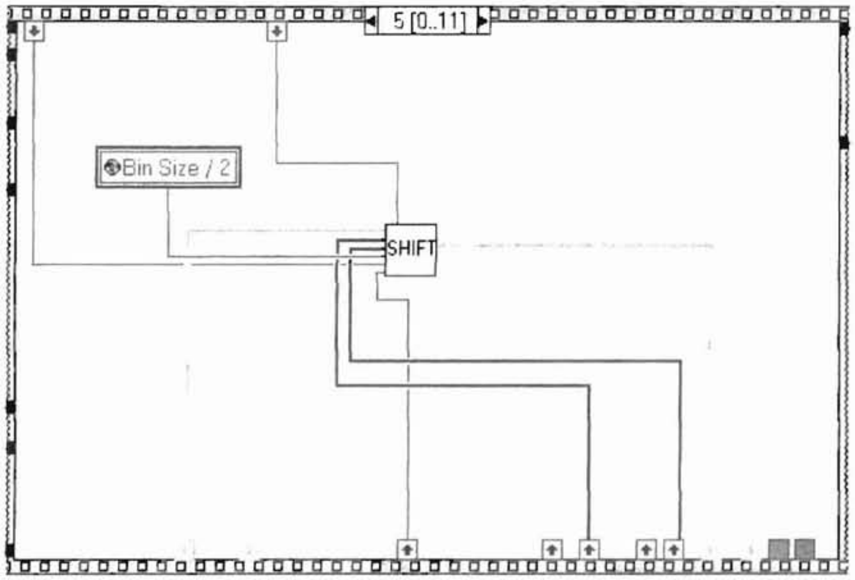


Figure 4.6.9: Harmonic Generator VI Sequence 1, Sub-sequence 5 Diagram

### 4.6.2.4 Sub-sequence 6 and 7 Diagrams

The sequences of 6 and 7 are used to take the first pass of data and save it to disk for access by spread sheet applications. The data is the pre and post shifting algorithm frequency data. Each data point represents a value in a specific bin represented in magnitude and phase. Sub sequence 7 will only execute if a stereo wave file is loaded.

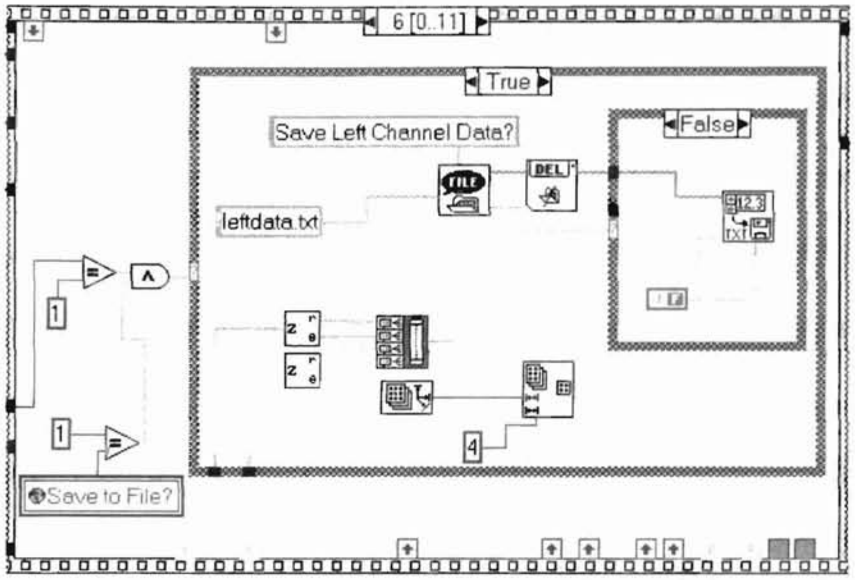


Figure 4.6.10: Harmonic Generator VI Sequence 1, Sub-sequence 6 Diagram

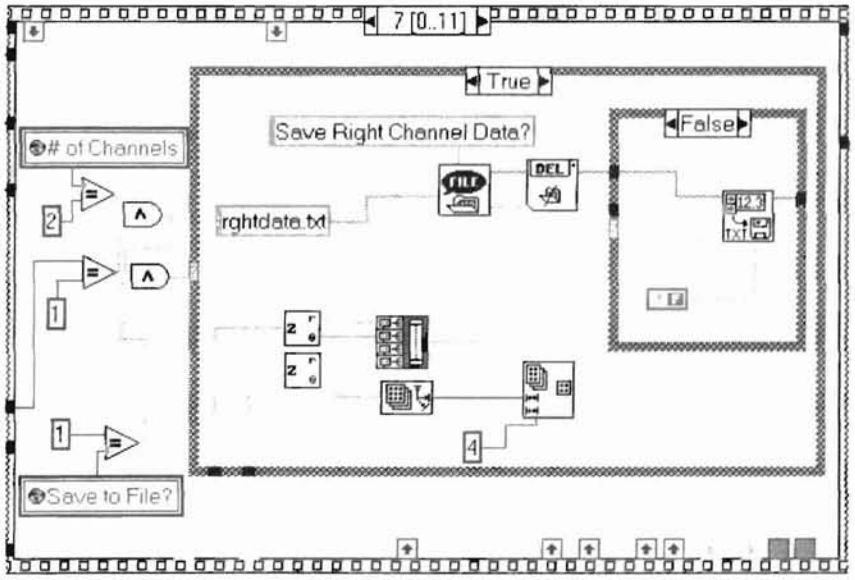


Figure 4.6.11: Harmonic Generator VI Sequence 1, Sub-sequence 7 Diagram

4.6.2.5 Sub-sequence 8 and 9 Diagrams

Sequences 8 and 9 are where the left and right channel shifted data is processed using an IFFT algorithm. The IFFT algorithm is explained in detail in Appendix A. The basic functionality of the IFFT algorithm is to perform the inverse fast fourier transform and to strip out the desired section from the middle portion of the generated harmonic waveform. This downsized version of the data is then multiplied by the scaling factor set on the front panel. Once scaled appropriately, the data is then summed with the original waveform, creating the new audio signal desired. Sequences 8 and 9 diagrams are shown in figures 4.6.12 and 4.6.13.

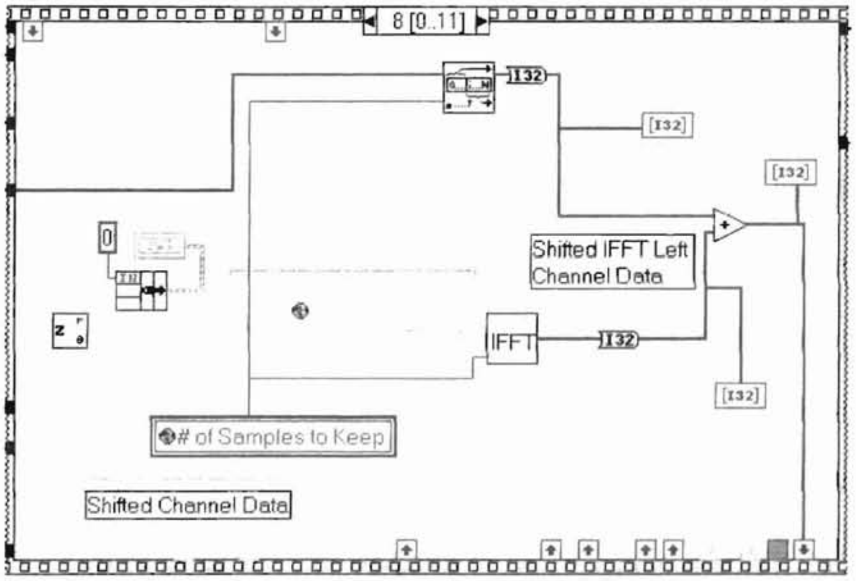


Figure 4.6.12: Harmonic Generator VI Sequence 1, Sub-sequence 8 Diagram

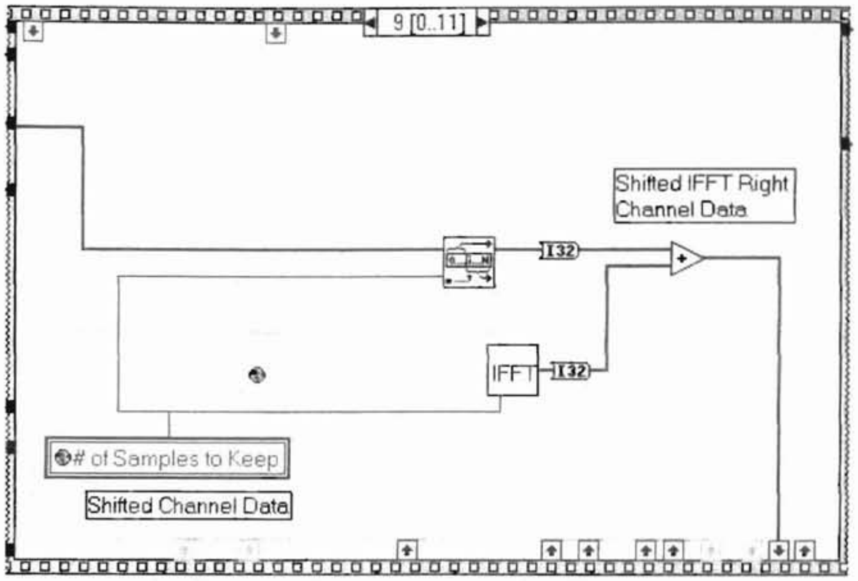


Figure 4.6.13: Harmonic Generator VI Sequence 1, Sub-sequence 9 Diagram

4.6.2.6 Sub-sequences 10 and 11 Diagrams

Sub-sequences 10 and 11 are used to create a complete array of data to be stored in the final step of the Harmonic Generator VI process. These two sequences simply append the new left and right channel data from the current pass, to the data created in earlier passes. The diagrams for sequences 10 and 11 are shown in figures 4.6.14 and 4.6.15. Once these two sequences are complete, the diagram of figure 4.6.16 is processed. This sequence prompts the operator to save the processed file. The Save Processed File VI is called in this sequence and is explained in great detail in Appendix A. This VI saves the newly created waveform audio data into a useable wave file. If the operator desires to save the file, the appropriate wave file header is applied to the data and then stored to disk. Whether the operator desires to save the file or not, the operator is returned to the Harmonic Generator VI panel.

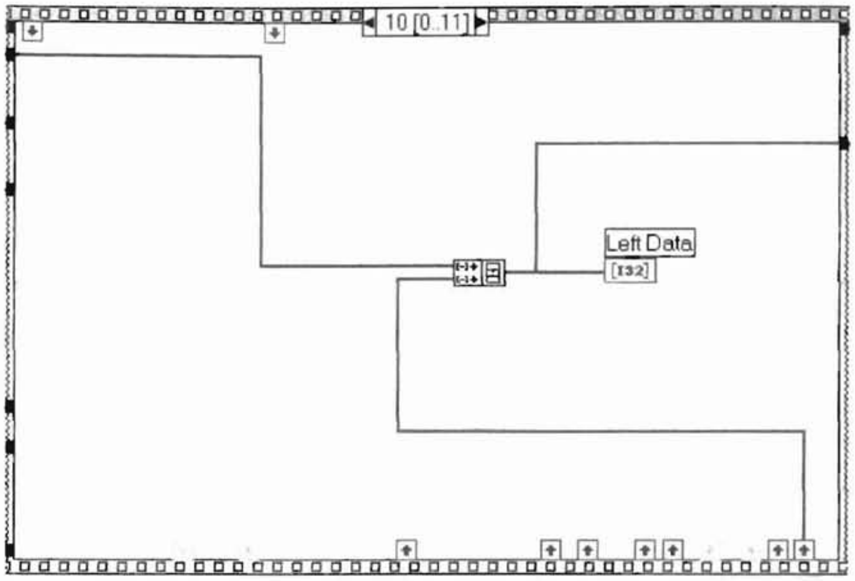


Figure 4.6.14: Harmonic Generator VI Sequence 1, Sub-sequence 10 Diagram

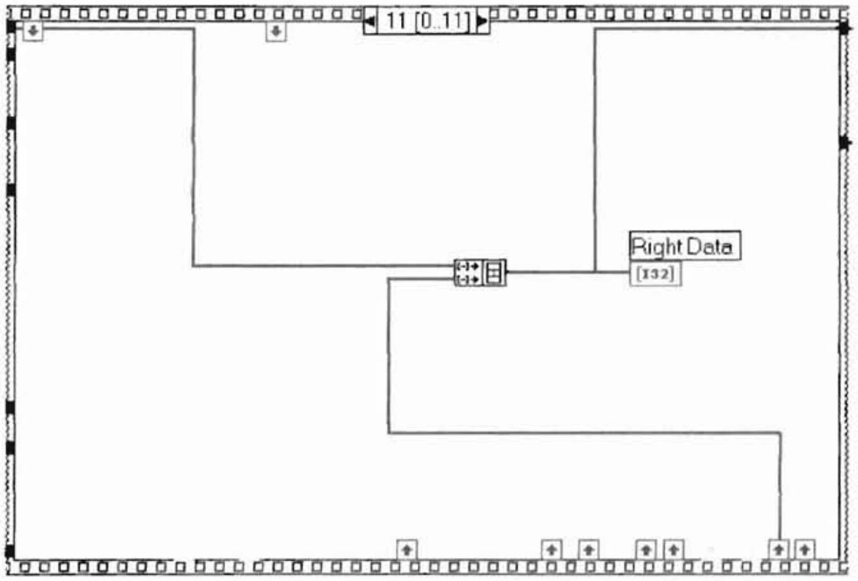


Figure 4.6.15: Harmonic Generator VI Sequence 1, Sub-sequence 11 Diagram

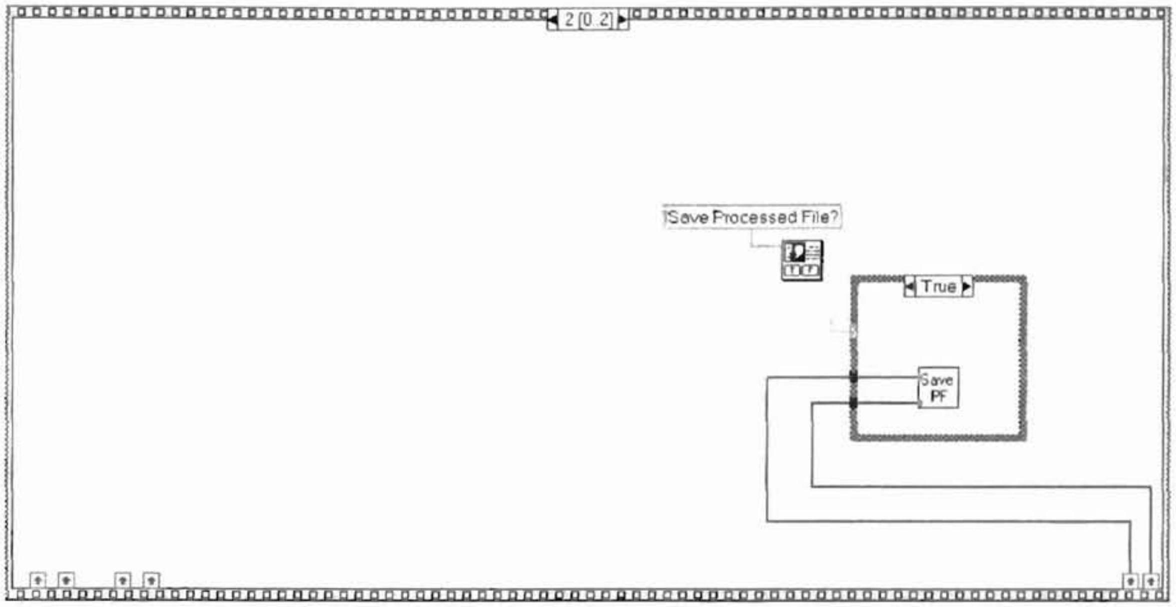
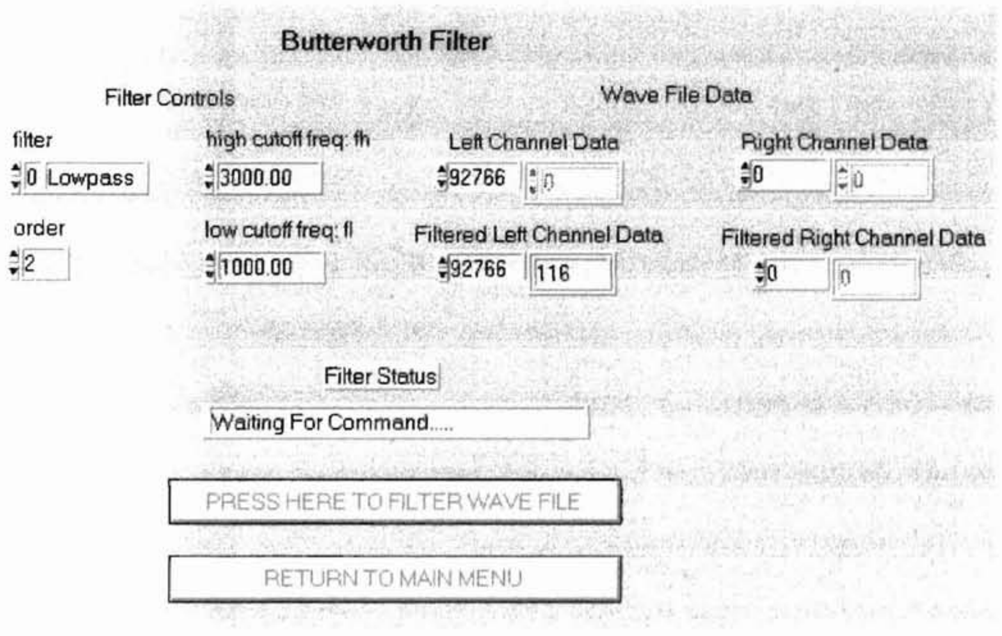


Figure 4.6.16: Harmonic Generator VI Sequence 2 Diagram

## 4.7 Filter VI Description

This VI is executed when the Filter Wave File button on the Main menu is pressed. This VI is a modified Butterworth Filter VI provided with National Instrument's Advanced Analysis package for LabVIEW [3]. It has been modified to accept the data format used by this program, to view the results of the filtering algorithm on a point-by-point case and to provide messaging during the filtering process. The front panel is shown in figure 4.7.1.


Butterworth Filter

Filter Controls

filter: 0 Lowpass

order: 2

high cutoff freq: fh: 3000.00

low cutoff freq: fl: 1000.00

Wave File Data

Left Channel Data: 92766 | 0

Right Channel Data: 0 | 0

Filtered Left Channel Data: 92766 | 116

Filtered Right Channel Data: 0 | 0

Filter Status: Waiting For Command.....

Figure 4.7.1: Filter VI Front Panel

The front panel has six controls. The first control is to select the type of filtering desired. The available options are lowpasss, highpass, bandpass and bandstop. The second control is to select the order of the filter. The order must be greater than zero. The third control is the low cutoff frequency. The low cutoff frequency must be greater than zero but less than one half of the sampling rate. This is required to meet the Nyquist criterion. The fourth control is the high cutoff frequency. The VI ignores this parameter when the filter type is set to lowpass or highpass. The fifth control is the filter wave file control button. Once the previous four controls have been set, the operator will select this button to execute the filtering algorithm. Once the filtering has been completed, the user can save the wave file to disk for use. The last control is the return to main menu button that exits the filter wave file VI and returns the user to the Main menu.

In addition to the controls, there are five data displays. The first two displays are the Left and Right Channel Data displays, which allow the user to scroll through the left and right channel data that is input to the filter. The third and fourth displays are the Filtered Left Channel Data display and the Filtered Right Channel Data display which allow the user to scroll through the filtered left and right channel channel data. The last indicator is the Filter Status window. The window provides the user with a status of "Waiting for Command" prior to and after filtering is complete and with a status of "Processing" when the filtering algorithm is executing.

The diagrams for the Filter VI are shown in figures 4.7.2 through 4.7.5. Figure 4.7.2 shows the first sequence of the algorithm where the filtered left channel data array is initialized to all zeroes.

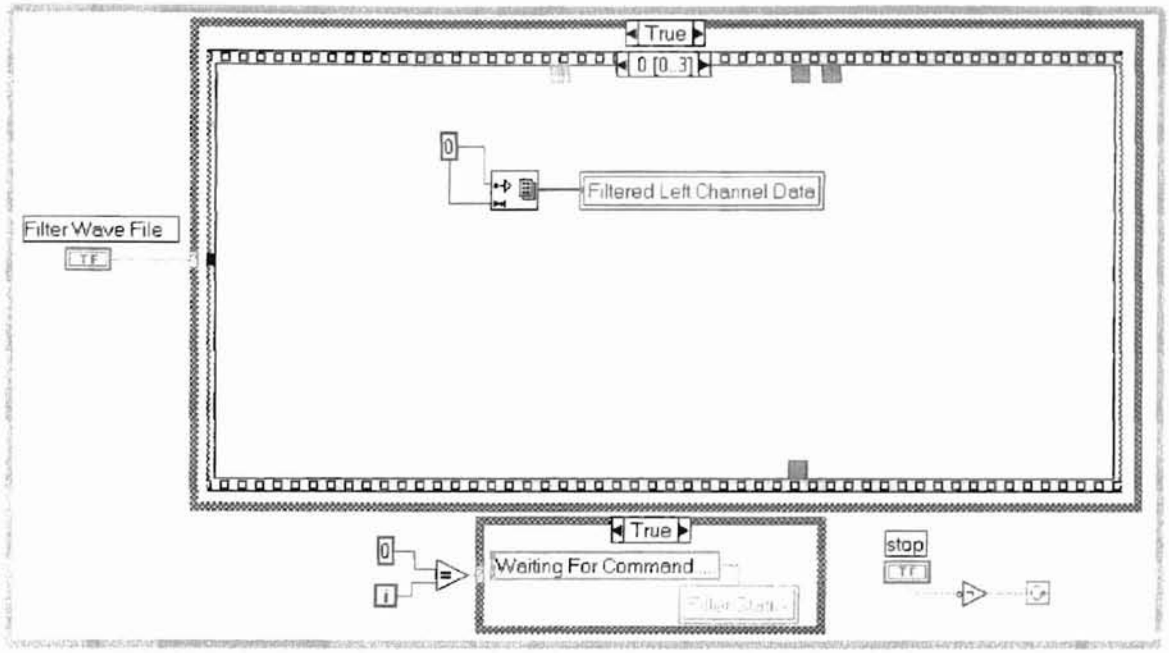


Figure 4.7.2: Filter VI Sequence 0 Diagram

Sequences 1 and 2, shown in figures 4.7.3 and 4.7.4 take in all of the left and right channel data and perform the Butterworth filtering algorithm based on the inputs from the front panel controls. The VIs shown in these sequences are provided with the original Butterworth filter VI from National Instruments. The VI shown on the left is the Butterworth Coefficients VI. This generates the set of filter coefficients to implement an Infinite Impulse Response (IIR) filter as specified by the Butterworth filter model. These coefficients are then passed to the IIR Cascade Filter VI. The number of filtering stages performed by this VI are controlled by the filter order selected on the front panel. The different stages are cascaded together, hence the name IIR Cascade Filter.

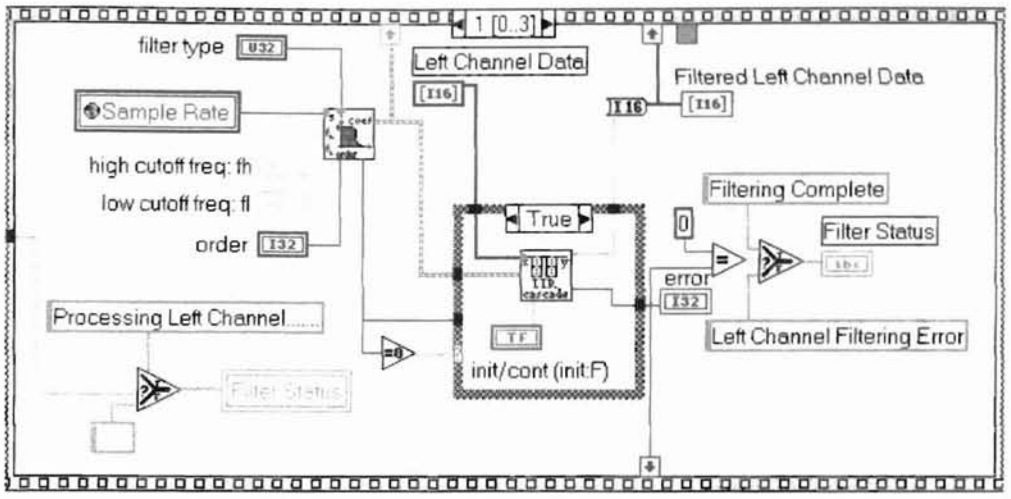


Figure 4.7.3: Filter VI Sequence 1 Diagram

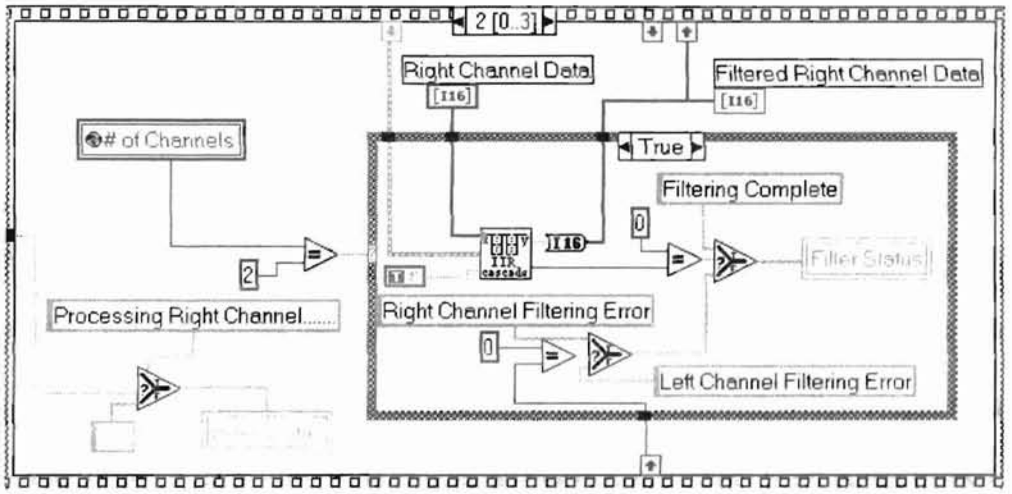

2 [0..3]
Right Channel Data
I16
Filtered Right Channel Data
# of Channels
True
Filtering Complete
IIR cascade
I16
2
0
Filter Status
Processing Right Channel......
Right Channel Filtering Error
Left Channel Filtering Error

Figure 4.7.4: Filter VI Sequence 2 Diagram

Sequence 3 combines the filtered left and right channel data and creates a filtered wave file that can be saved to disk. The Save Filtered File VI is identical to the Save Processed File VI described in Appendix A except for the fact that the wave file is not normalized since data summation is not required. Therefore, the Save Filtered File VI is not described any further.

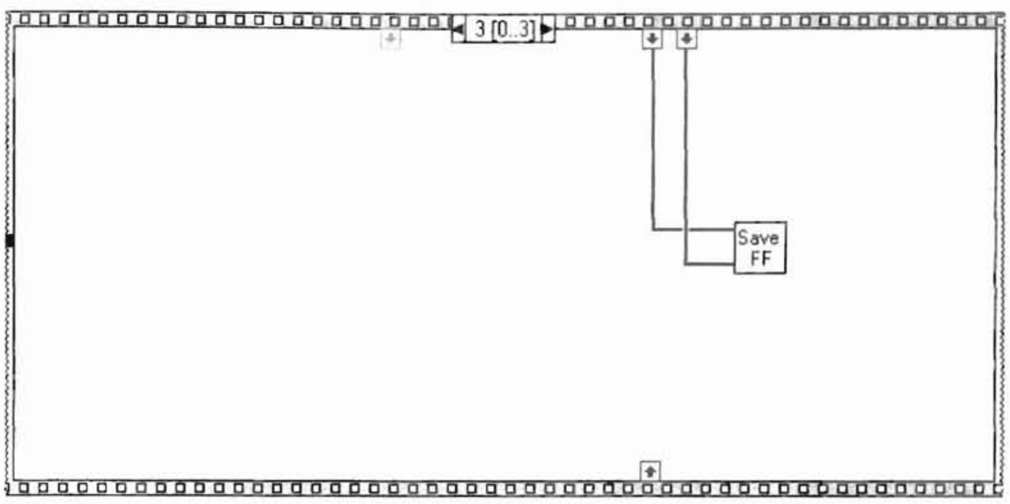
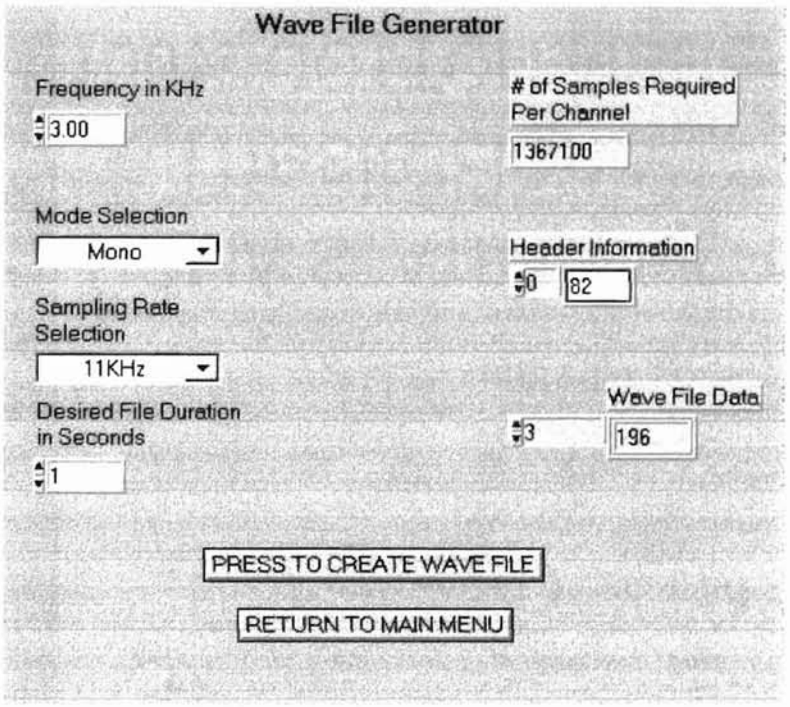


Figure 4.7.5: Filter VI Sequence 3 Diagram

## 4.8 Create Sine Wave VI Description

The Create Sine Wave VI is used to create wave files that are of a fixed frequency. This VI is called when the Create A Sine Wave File button is selected on the Main menu. This VI allows the user to create wave files for use in learning the process of the harmonic generating algorithm since it is easiest observed with a single frequency signal. The front panel of this VI is shown in figure 4.8.1. This front panel has six separate controls. The first is the 'Frequency in KHz' control which allows the user to select the wave file frequency. This value must not be greater than one half of the Sampling Rate Selection value. This again is a requirement needed to satisfy the Nyquist criterion. The second control is the Mode Selection. Currently there is only one valid option for mode selection, Mono, meaning that only left channel data will be created. The third control is the Sampling Rate Selection. Valid selections are 11, 22 and 44KHz. The fourth control is the Desired File Duration. The next control is the Create Wave File button. Once the parameters of the previously described four controls have been selected, the user presses this button and the file is created and the operator is then brought to a dialog box to select the name of the file to save. Pressing the Return to Main menu takes the user back to the Main menu and exits the Create Sine Wave VI.


Wave File Generator

Frequency in KHz: 3.00

# of Samples Required Per Channel: 1367100

Mode Selection: Mono

Header Information: 0 82

Sampling Rate Selection: 11KHz

Wave File Data: 3 196

Desired File Duration in Seconds: 1

Figure 4.8.1: Create Sine Wave VI Front Panel

The diagrams for the Create Sine Wave VI are shown in figures 4.8.2 through 4.8.9. Figure 4.8.2 shows sequence 0 which is where the LabVIEW advanced analysis package Sine Wave VI is called [3]. This VI takes in the number of samples required and the frequency of the sine wave from the data generated by the front panel controls. The output of the Sine Wave VI is equivalent to sampling a 2Vp-p signal centered around zero and recording the data into an array with a maximum value of 1 and a minimum value of -1. Since valid wave file ranges for eight bit data is 255 to 0 the output of the Sine Wave VI is multiplied by 127 and a DC offset of 127 added in order to get the correct signal range.

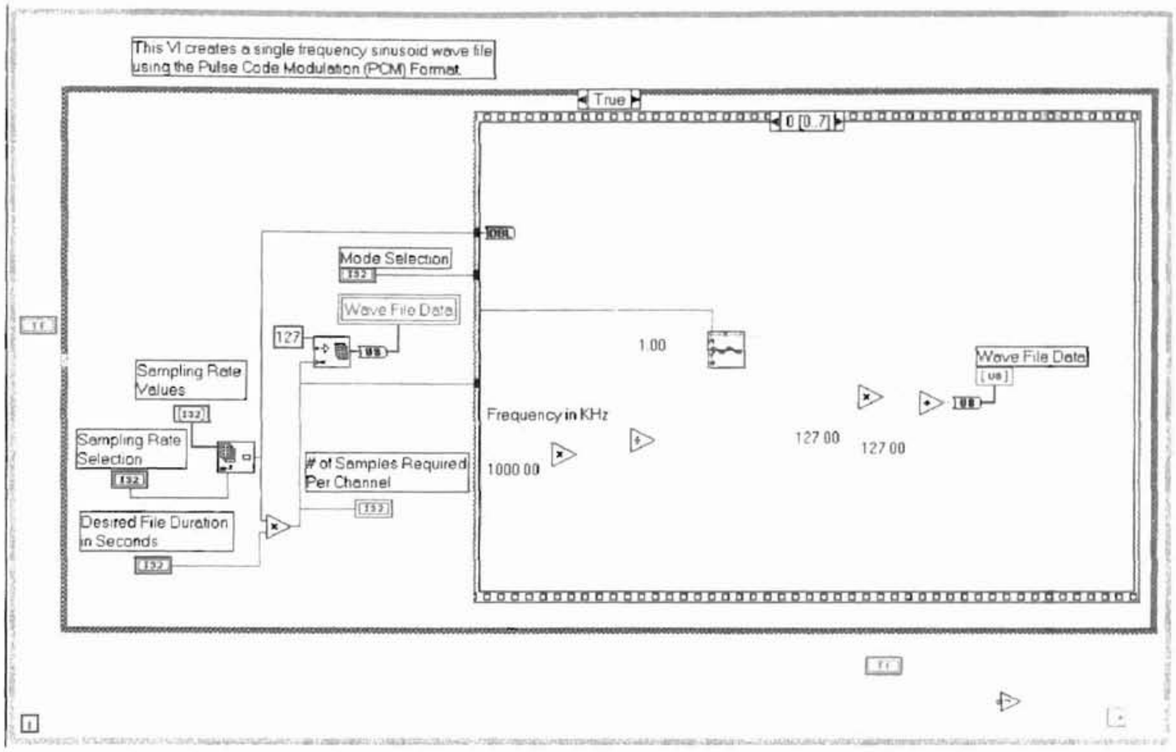

This VI creates a single frequency sinusoid wave file using the Pulse Code Modulation (PCM) Format.
True
0 [0..7]
Mode Selection
Wave File Data
127
Sampling Rate Values
Sampling Rate Selection
# of Samples Required Per Channel
Desired File Duration in Seconds
Frequency in KHz
1000.00
1.00
127.00
127.00
Wave File Data

Figure 4.8.2: Create A Sine Wave VI Diagram Showing Sequence 0

Sequences 1 through 6 are used to create the wave file header to make it compliant to the wave file format discussed in Chapter II of this document. Sequence 7 is used to combine the header of the wave file with the wave file data generated by sequence 0. During execution of sequence 7 is also where the operator is prompted to save the file to disk. This process is again very similar to the Save Processed File VI described in Appendix A and will not be described further.

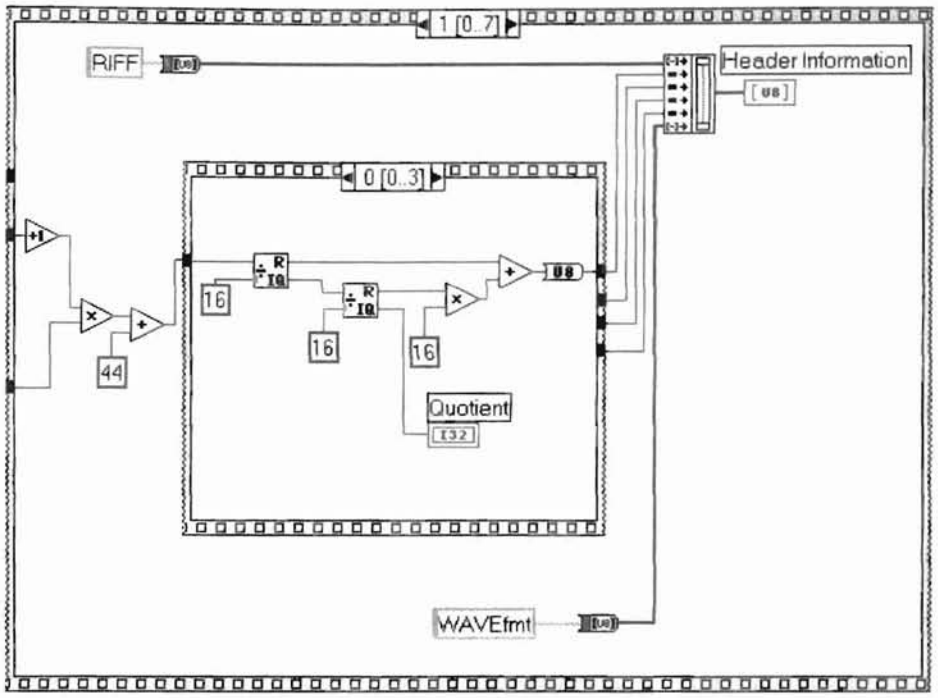


Figure 4.8.3: Create A Sine Wave VI Sequence 1 Diagram

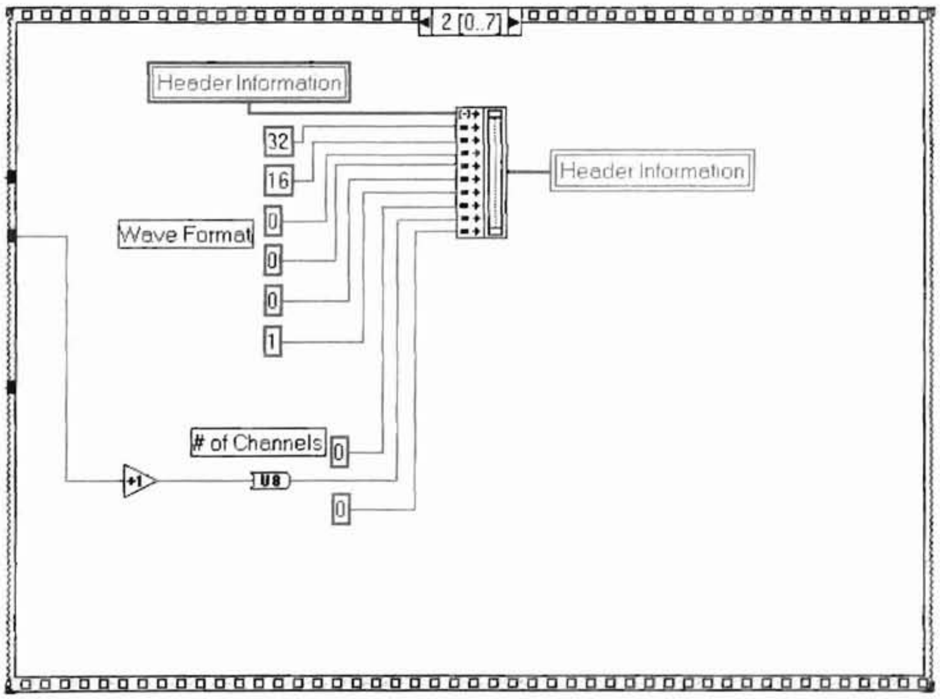


Figure 4.8.4: Create A Sine Wave VI Sequence 2 Diagram

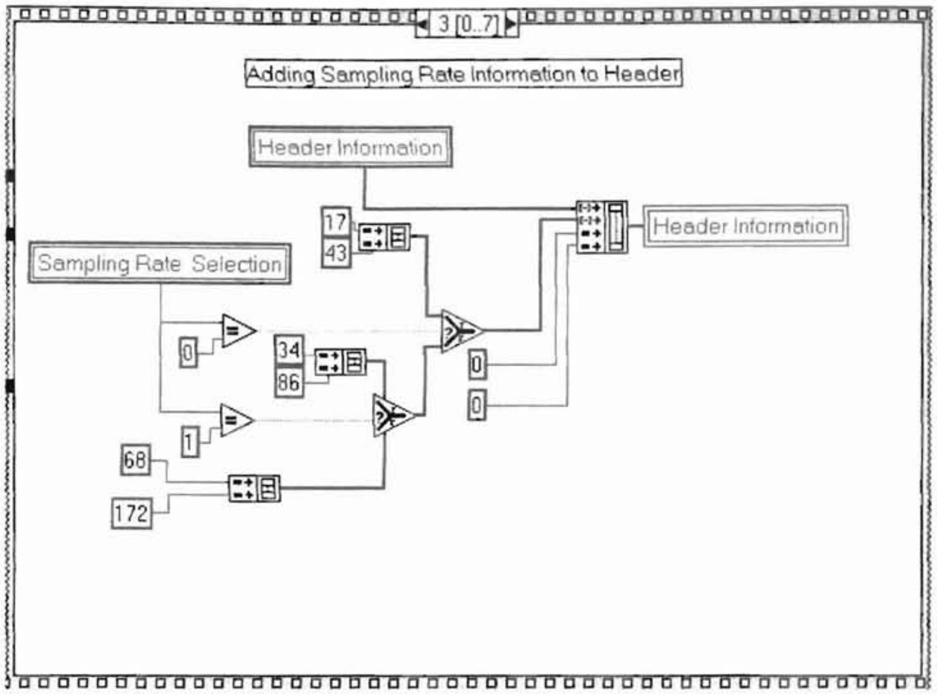


Figure 4.8.5: Create A Sine Wave VI Sequence 3 Diagram

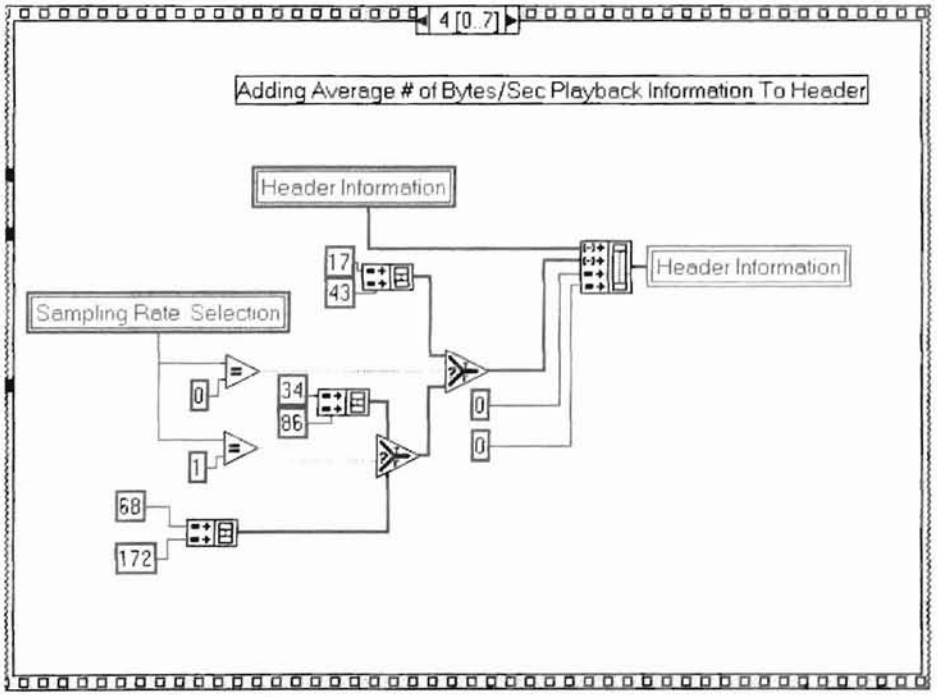


Figure 4.8.6: Create A Sine Wave VI Sequence 4 Diagram

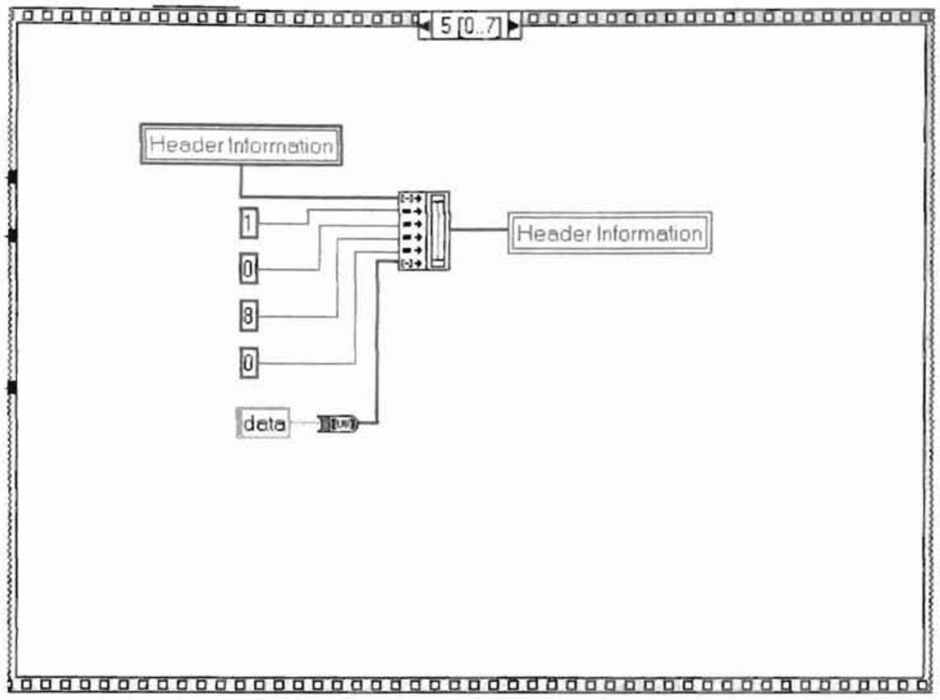


Figure 4.8.7: Create A Sine Wave VI Sequence 5 Diagram

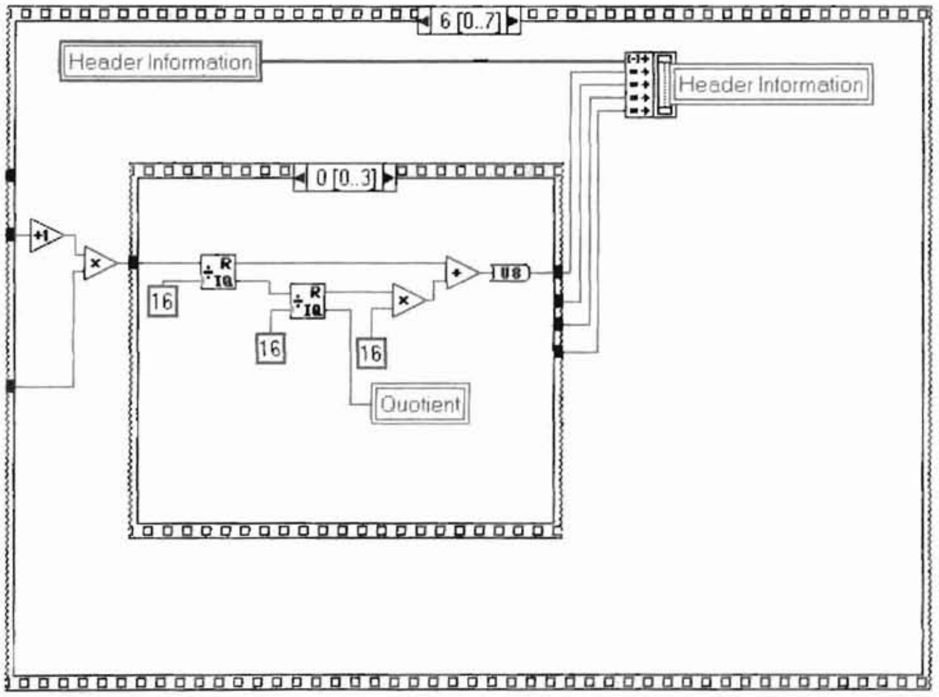


Figure 4.8.8: Create A Sine Wave VI Sequence 6 Diagram

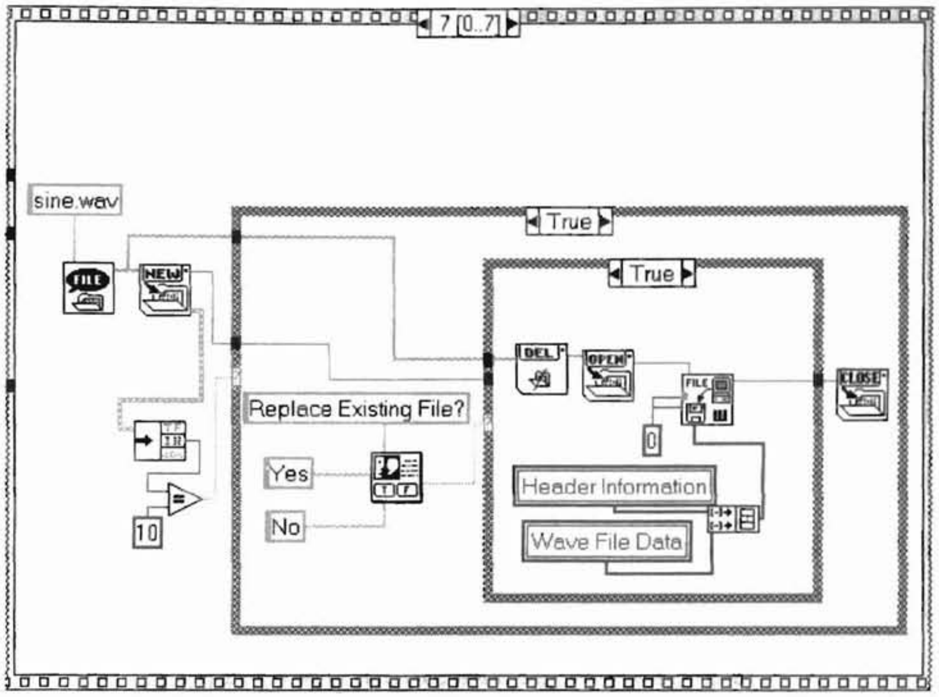
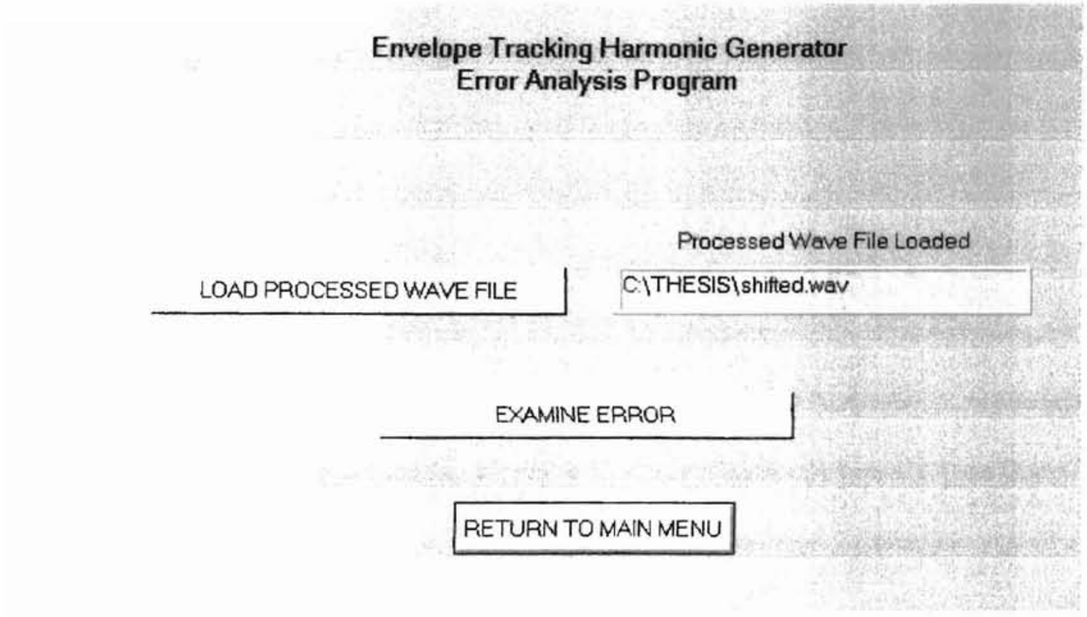


Figure 4.8.9: Create A Sine Wave VI Sequence 7 Diagram

## 4.9 Error Analysis VI Description

This VI is used to assist in determining the amount of error that can be expected. The premise behind this VI is to compare the generated harmonic waveform with an approximation of the desired results. When selected the panel of figure 4.9.1 appears. This panel allows the operator to select a specific wave file to load into memory for comparison. Once a wave file is loaded, the Examine Error button will take the user to the wave file comparison window of figure 4.9.2.


Envelope Tracking Harmonic Generator
Error Analysis Program

Processed Wave File Loaded

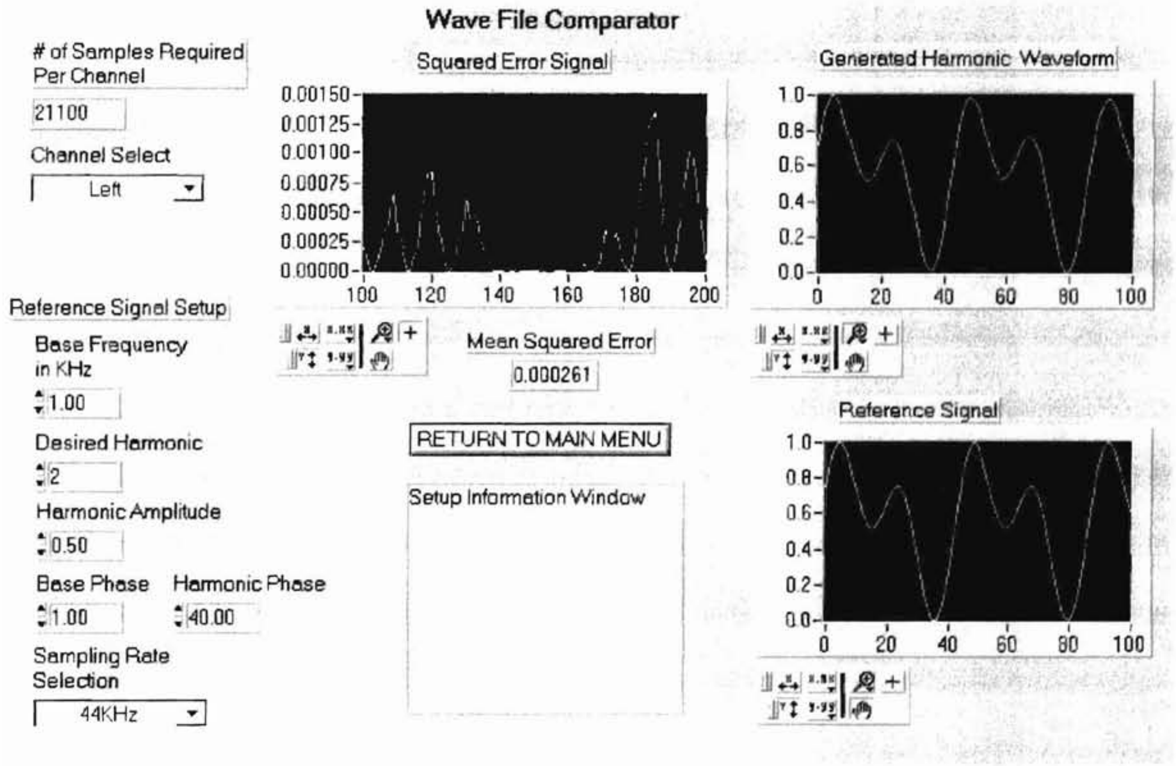
LOAD PROCESSED WAVE FILE

C:\THESIS\shifted.wav

EXAMINE ERROR

RETURN TO MAIN MENU

Figure 4.9.1: Error Analysis VI Front Panel


Wave File Comparator

# of Samples Required Per Channel: 21100

Channel Select: Left

Reference Signal Setup

Base Frequency in KHz: 1.00

Desired Harmonic: 2

Harmonic Amplitude: 0.50

Base Phase: 1.00

Harmonic Phase: 40.00

Sampling Rate Selection: 44KHz

Mean Squared Error: 0.000261

RETURN TO MAIN MENU

Setup Information Window

Figure 4.9.2: Wave File Comparison Panel

The wave file comparison window provides the user with several controls and indicators. On the left hand side of the panel are the operator controls for selecting which channel to examine if a stereo file is loaded and the reference signal setup controls, which allows the operator to select the base frequency of the signal from which a harmonic will be generated. The operator has control over the base and harmonic frequencies. This allows the operator to match the phases of the file processed by the Harmonic Generating algorithm. By carefully selecting the phase and amplitude of the reference signal, a good approximation can be created. Provided in the graphs are the representations of the Generated Harmonic waveform processed by the algorithm, the reference signal and the squared error signal. The squared error signal is the square of the difference between the processed and the reference signal point by point. Below the squared error signal graph is the mean squared error value determined for the complete duration of the file.

Figure 4.9.3 shown below shows sequence 0 of the error analysis VI. Sequence 0 is invoked when the operator first selects the error analysis VI or when the operator selects the Load Processed Wave File button on the error analysis panel. This uses the open VI referenced in appendix A and splits the wave file data into left and right channels as necessary.

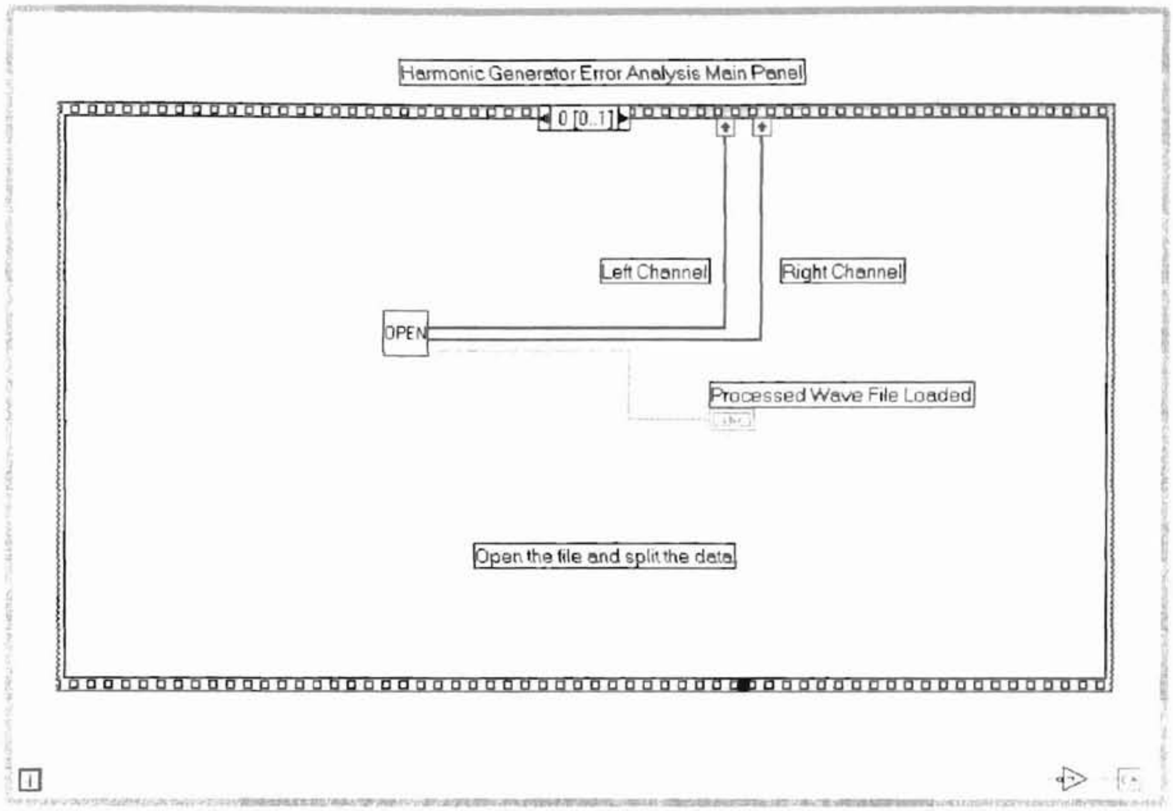

Harmonic Generator Error Analysis Main Panel
0 [0..1]
Left Channel
Right Channel
OPEN
Processed Wave File Loaded
Open the file and split the data

Figure 4.9.3: Error Analysis Sequence 0 Diagram

Sequence 1, shown in figure 4.9.4, invokes the compare wave file VI when the Examine Error button on the error analysis VI front panel is selected. When the compare wave file VI is called the sequences of figures 4.9.5 through 4.9.8 are called. Sequences 0 and 1 are used to create the reference signal that the processed wave file is compared against. Sequence 0 creates a summed sinusoid signal. Sequence 1 takes that signal and normalizes it so that the peak value does not exceed 1 and the minimum value does not go below 0. Sequence 2, normalizes the generated signal max value to 1 by dividing it by 255. Sequence 3 generates the mean squared error and the squared error signals for display on the front panel.

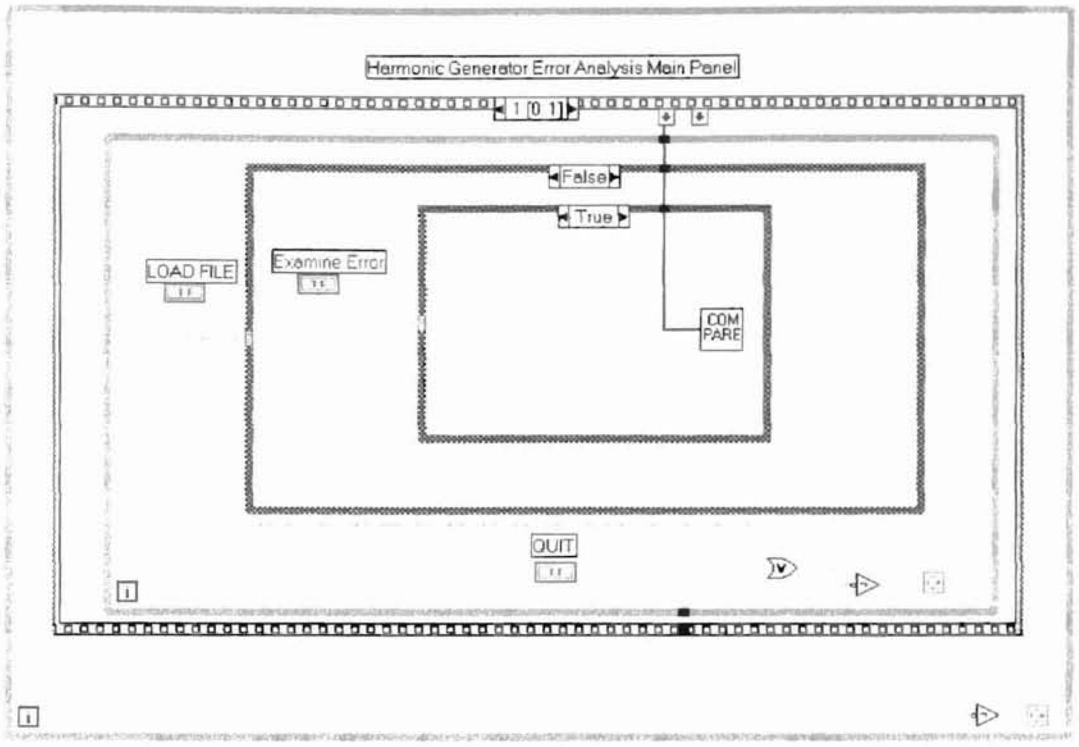


Figure 4.9.4: Error Analysis VI Sequence 1 Diagram

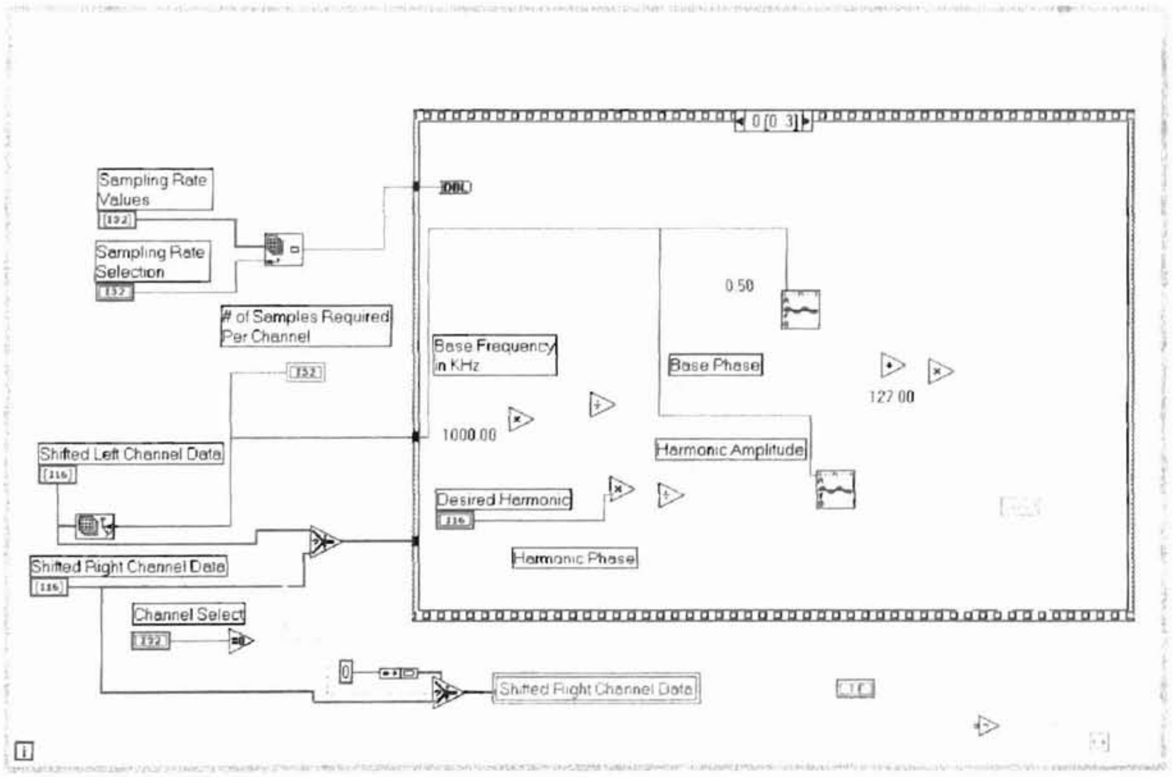


Figure 4.9.5: Waveform Comparison Panel Sequence 0 Diagram

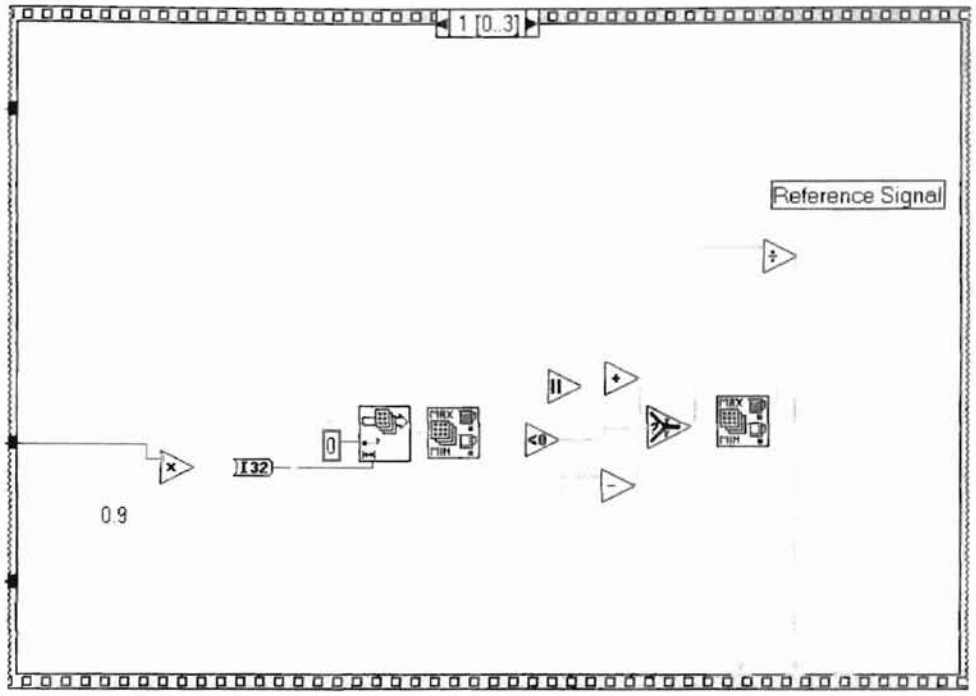


Figure 4.9.6: Waveform Comparison Sequence 1 Diagram

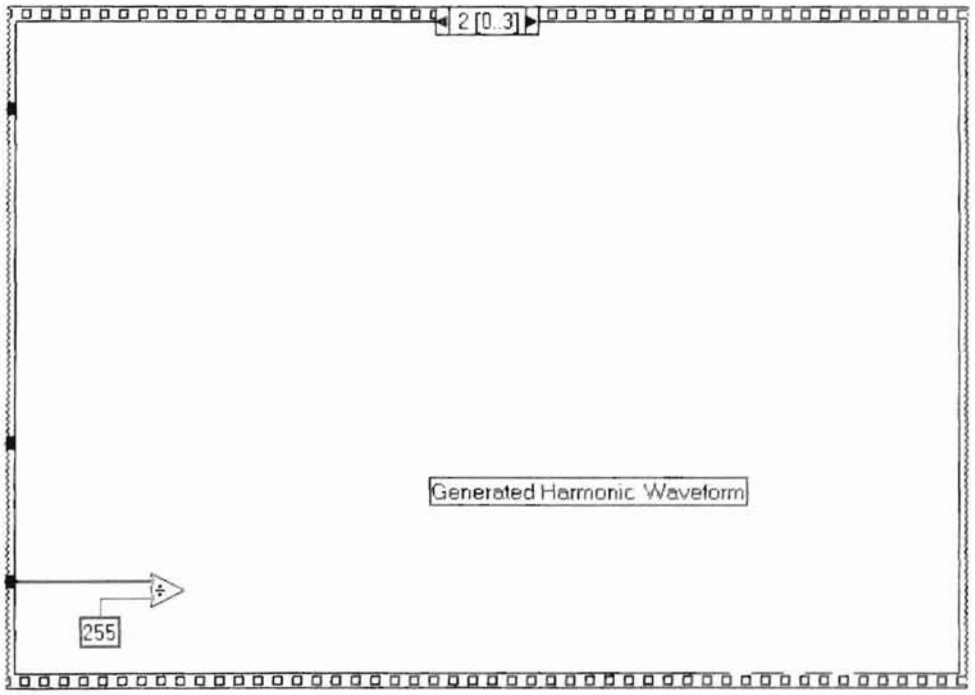


Figure 4.9.7: Waveform Comparison Sequence 2 Diagram

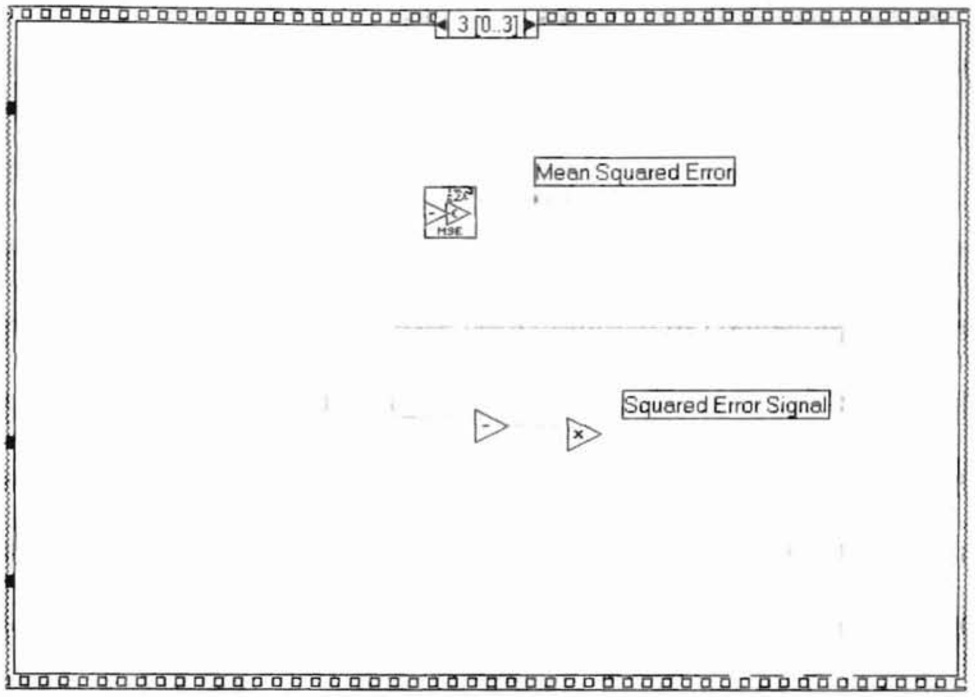


Figure 4.9.8: Waveform Comparison Sequence 3 Diagram

# CHAPTER 5

# PROJECT RESULTS

The goal of the project was to develop a LabVIEW based program that implemented the harmonic generating algorithm correctly to aid in further analysis of the algorithm. The results of this project have been evaluated by comparing the output of the harmonic generator with a reference signal created by summing two sine waves of different frequencies together. Numerous parameters were manipulated using the algorithm. The abbreviated results are shown in Table 5.1 and show that a small amount of error can be expected when using this algorithm. Representative detailed output data is provided in appendix B. From the data in Table 5.1, it is obvious that to minimize the amount of error seen, the data from the center of the generated waveform should be used where error is the least. The number of FFT samples taken at a time had a slight effect on the mean squared error (MSE), but the number of samples kept had the largest impact on error results. This concurs with Scheets results. Another driver to minimizing the number of FFT samples is throughput. Test run 23, performing 1000 point FFTs, took 7 minutes, 42 seconds to run, whereas test run 28 using 100 point FFTs took only 2 min 12 seconds. Test run 23 had an MSE of .000014 and test run 28 had an MSE of .000795. Further research could be performed to determine if this difference is negligible for this application.

The results verify that the algorithm was implemented correctly within the bounds of the scope of this project. Therefore, it can be concluded that the goal of this project has been met.

| Test Run # | Sample Rate | Base Frequency | Bits/Sample | # of FFT Samples | # of Samples Kept | Bin Size/2 | Harmonic | Min MSE |
|---|---|---|---|---|---|---|---|---|
| 1 | 44100 | 1K | 8 | 1000 | 1000 | 10 | 2 | 0.018036 |
| 2 | 44100 | 1K | 8 | 1000 | 1000 | 1000 | 2 | 0.027293 |
| 3 | 44100 | 1K | 8 | 1000 | 500 | 10 | 2 | 0.003702 |
| 4 | 44100 | 1K | 8 | 1000 | 100 | 10 | 2 | 0.000261 |
| 5 | 44100 | 1K | 8 | 1000 | 10 | 10 | 2 | 0.000011 |
| 6 | 44100 | 1K | 8 | 500 | 500 | 10 | 2 | 0.019336 |
| 7 | 44100 | 1K | 8 | 500 | 100 | 10 | 2 | 0.000849 |
| 8 | 44100 | 1K | 8 | 500 | 10 | 10 | 2 | 0.000057 |
| 9 | 44100 | 1K | 8 | 100 | 100 | 10 | 2 | 0.056642 |
| 10 | 44100 | 1K | 8 | 100 | 10 | 10 | 2 | 0.017398 |
| 11 | 11025 | 1K | 8 | 1000 | 1000 | 10 | 2 | 0.015467 |
| 12 | 11025 | 1K | 8 | 1000 | 500 | 10 | 2 | 0.003204 |
| 13 | 11025 | 1K | 8 | 1000 | 100 | 10 | 2 | 0.000245 |
| 14 | 11025 | 1K | 8 | 1000 | 10 | 10 | 2 | 0.000012 |
| 15 | 11025 | 1K | 8 | 500 | 500 | 10 | 2 | 0.020168 |
| 16 | 11025 | 1K | 8 | 500 | 100 | 10 | 2 | 0.000765 |
| 17 | 11025 | 1K | 8 | 500 | 10 | 10 | 2 | 0.000029 |
| 18 | 11025 | 1K | 8 | 100 | 100 | 10 | 2 | 0.001407 |
| 19 | 11025 | 1K | 8 | 100 | 10 | 10 | 2 | 0.000036 |
| 20 | 22050 | 1K | 8 | 1000 | 1000 | 10 | 2 | 0.020177 |
| 21 | 22050 | 1K | 8 | 1000 | 500 | 10 | 2 | 0.004972 |
| 22 | 22050 | 1K | 8 | 1000 | 100 | 10 | 2 | 0.000276 |
| 23 | 22050 | 1K | 8 | 1000 | 10 | 10 | 2 | 0.000014 |
| 24 | 22050 | 1K | 8 | 500 | 500 | 10 | 2 | 0.018016 |
| 25 | 22050 | 1K | 8 | 500 | 100 | 10 | 2 | 0.000726 |
| 26 | 22050 | 1K | 8 | 500 | 10 | 10 | 2 | 0.000032 |
| 27 | 22050 | 1K | 8 | 100 | 100 | 10 | 2 | 0.030229 |
| 28 | 22050 | 1K | 8 | 100 | 10 | 10 | 2 | 0.000795 |
| 29 | 22050 | 1K | 8 | 100 | 10 | 100 | 2 | 0.000756 |
| 30 | 22050 | 1K | 8 | 100 | 10 | 1 | 2 | 0.001312 |
| 31 | 11025 | 2K | 8 | 1000 | 1000 | 10 | 2 | 0.025098 |
| 32 | 11025 | 2K | 8 | 1000 | 100 | 10 | 2 | 0.000406 |
| 33 | 22050 | 2K | 8 | 1000 | 1000 | 10 | 2 | 0.015504 |
| 34 | 22050 | 2K | 8 | 1000 | 100 | 10 | 2 | 0.000246 |
| 35 | 44100 | 2K | 8 | 1000 | 1000 | 10 | 2 | 0.015498 |
| 36 | 44100 | 2K | 8 | 1000 | 100 | 10 | 2 | 0.000246 |
| 37 | 44100 | 1K | 8 | 1000 | 100 | 10 | 3 | 0.000858 |
| 38 | 44100 | 1K | 8 | 1000 | 100 | 10 | 4 | 0.001207 |

Table 5-1: Abbreviated Test Results

# REFERENCES

Microsoft Corporation. Waveform Audio File Format (WAVE). Web page document. HTML addres: http://keck.ucsf.edu/~jwright/RIFF-format.html, 1997.

Microsoft Corporation. WaveFormat Structure for PCM. Web page document. HTML address: http://premium.microsoft.com/msdn/library/tools/3rdparty/lernout/dl/s41bb.htm, 1997.

National Instruments Corporation. LabVIEW Analysis VI Reference Manual. Reference Manual. National Instruments Corporation, January 1996.

Scheets, George M. An all digital envelope tracking harmonic generator. Stillwater, OK: Oklahoma State University Department of Electrical Engineering, 1992.

## APPENDIX A

## SUPPLEMENTAL VI DESCRIPTIONS

The VI descriptions located in Appendix A are the ancillary VIs that were deemed unnecessary to describe in the body of this document. This may be due to similarity to other VIs that have been described in earlier sections or that their inclusion within the body of this document was not considered desirable in describing the required algorithm implementation.

A.1 Bits per Sample VI: Used to determine the number of bits per sample that the wave file was recorded in. This value is extracted as the 23rd byte in the wave file header. Valid ranges for this program are 8 and 16.

Figure A.1.1: Bits Per Sample VI Icon

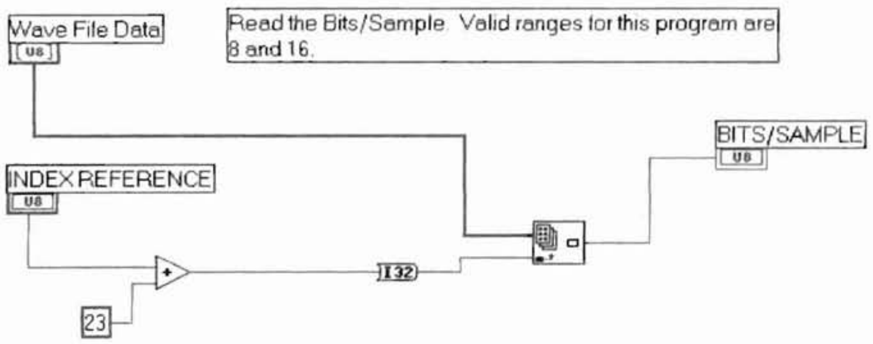

Wave File Data
U8
Read the Bits/Sample. Valid ranges for this program are 8 and 16.
BITS/SAMPLE
U8
INDEX REFERENCE
U8
+
I32
23

Figure A.1.2: Bits per Sample VI Diagram

A.2 Convert Data VI: Used to convert 16 bit wave file data words into two separate 8 bit words. All data must be converted in the 8 bit word format when the wave file is saved.

Figure A.2.1: Convert Data VI Icon

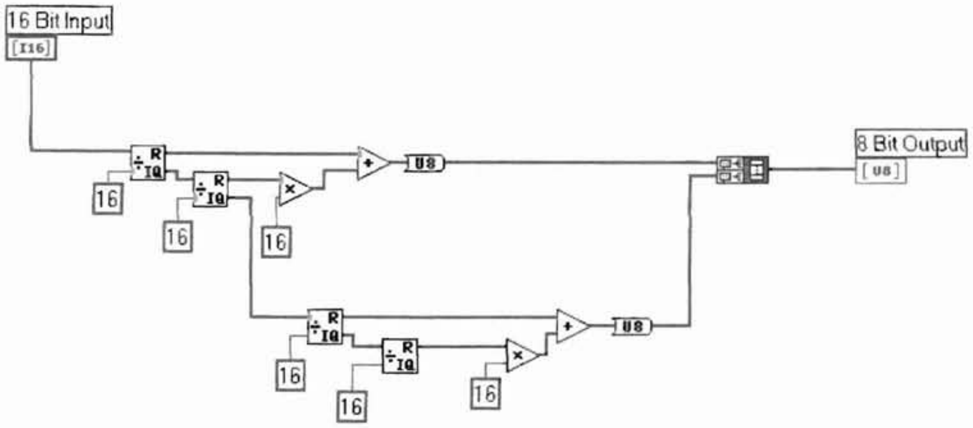


Figure A.2.2: Convert Data VI Diagram

A.3 Data Location VI: Used to find the ASCII word "data" in the wave file header. The presence of the word data indicates that data appears immediately after and is also used to verify that the wave file includes data.

Figure A.3.1: Data Location VI Icon

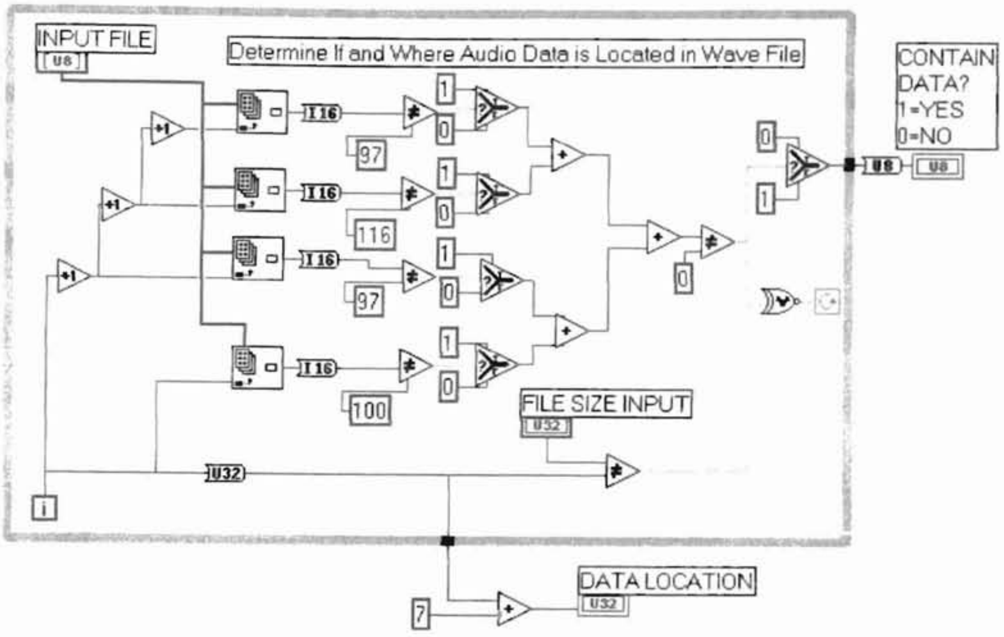


Figure A.3.2: Data Location VI Diagram

A.4 File Format VI: Used to find consecutive appearance of the ASCII letters "Efmt" in the wave file header. The presence of these letters indicates that the wave file format data appears immediately after and is also used to verify that the wave file includes a valid header.

Figure A.4.1: File Format VI Icon

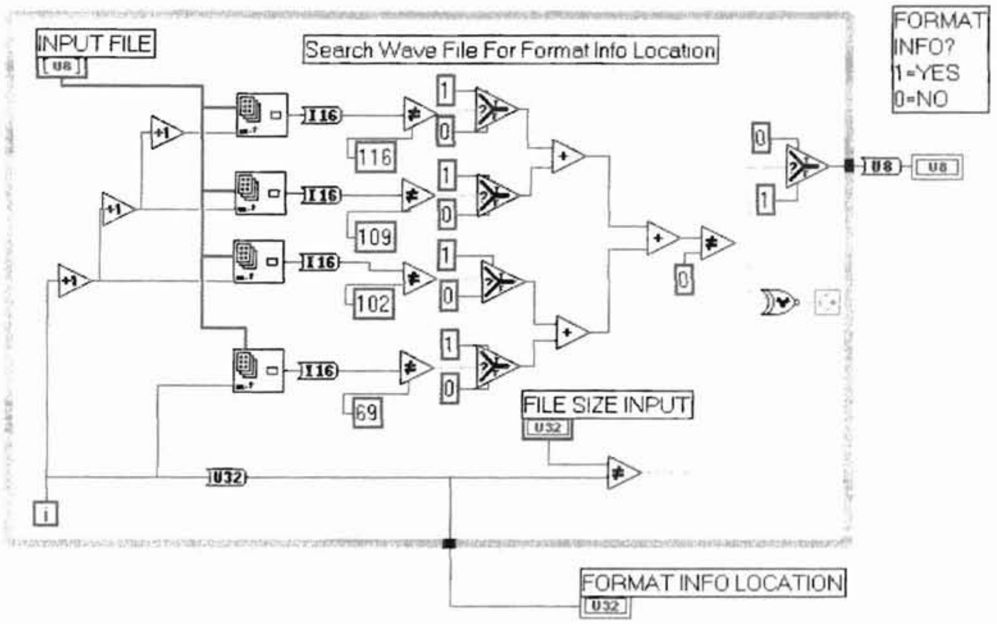


Figure A.4.2: File Format VI Diagram

A.5 Inverse Fast Fourier Transform (IFFT) VI: Used to take the data created by shifting the bins and phases per the harmonic generation algorithm and performing an inverse fast fourier transform. The IFFT VI shown in the diagram is supplied with the LabVIEW Advanced Analysis package. Once the IFFT is performed, only the number of data points from the center is kept per the input of the number of samples to keep. This is accomplished by determining the size of the original array input, subtracting the value of the number of samples to keep and then dividing the difference by two to establish the start of the number of points to keep out of the original array. The scaling factor is used to scale the harmonic data before being added back to the original data.

Figure A.5.1: IFFT VI Icon

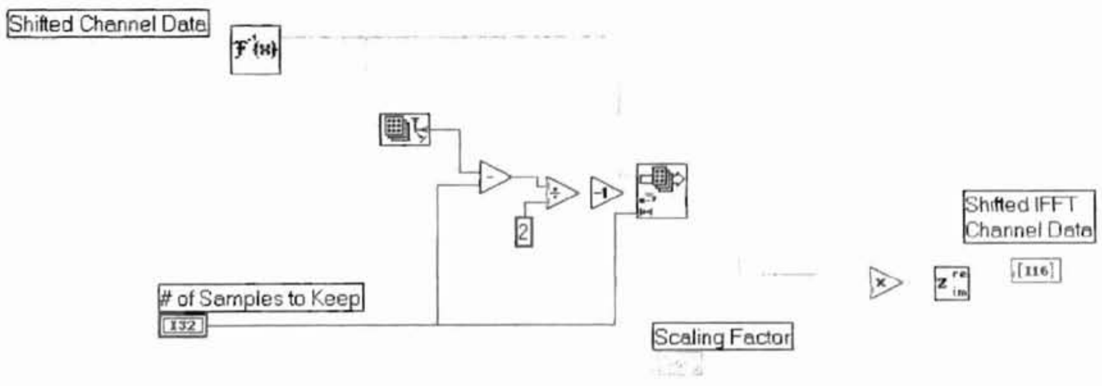


Figure A.5.2: IFFT VI Diagram

A.6 Open File VI: Used to load the wave file data into program memory for processing. The first sequence clears any resident wave file data from memory. Sequence 1 opens the wave file from the computer and stores that data into an 8-bit array. The original wave file is then closed.

Figure A.6.1: Open File VI Icon

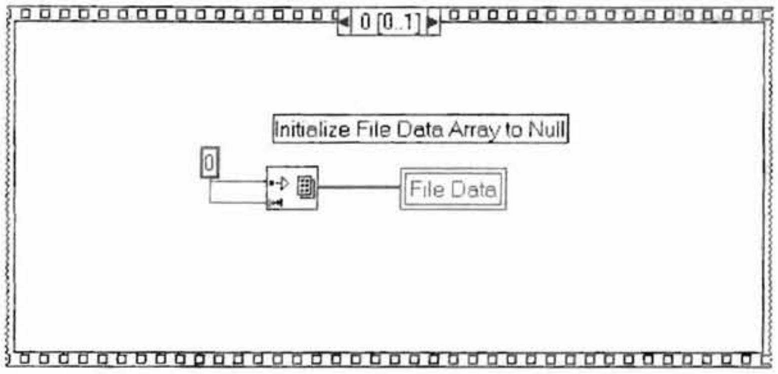


Figure A.6.2: Open File VI Sequence 0 Diagram

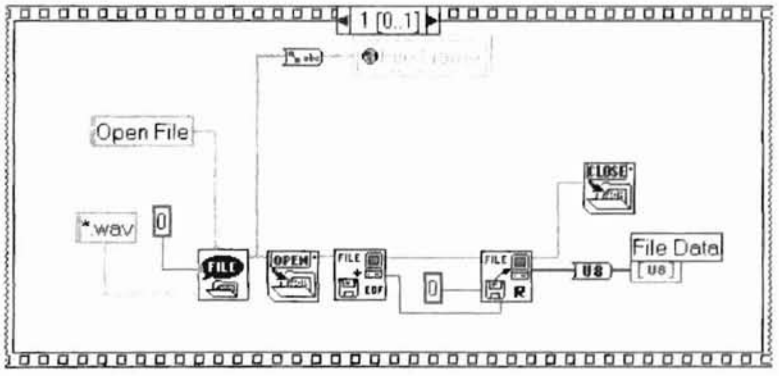


Figure A.6.3: Open File VI Sequence 1 Diagram

A.7 Peak Find VI: Used to establish where maximum frequency bins occur after the Fast Fourier Transform of the wave file has been performed. The operator provides input as to how many peaks are to be shifted, how many bins are shifted around each peak, how far to shift each peak and which peaks below and above specified values are ignored.

Figure A.7.1: Peak Find VI Icon

Figure A.7.2 shows the inputs specified above. It also shows that depending on the harmonic selected, there is an upper limit allowed for shifting since we cannot shift higher than the center point of the frequency domain. Sequence 0 clears any data from previous calculations.

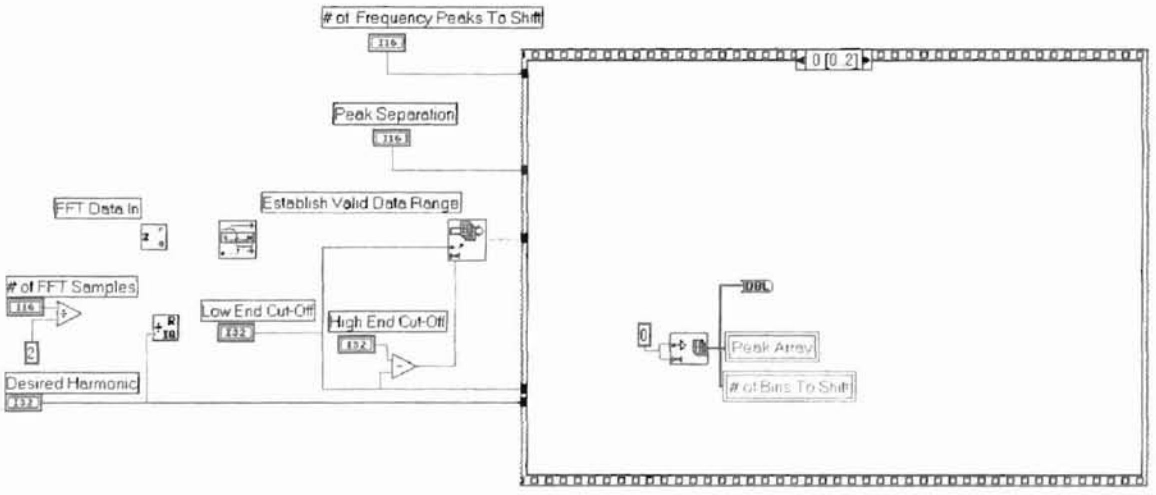


Figure A.7.2: Peak Find VI Sequence 0 Diagram

Sequence 1 stores the current array of data into a local variable for processing in sequence 2.

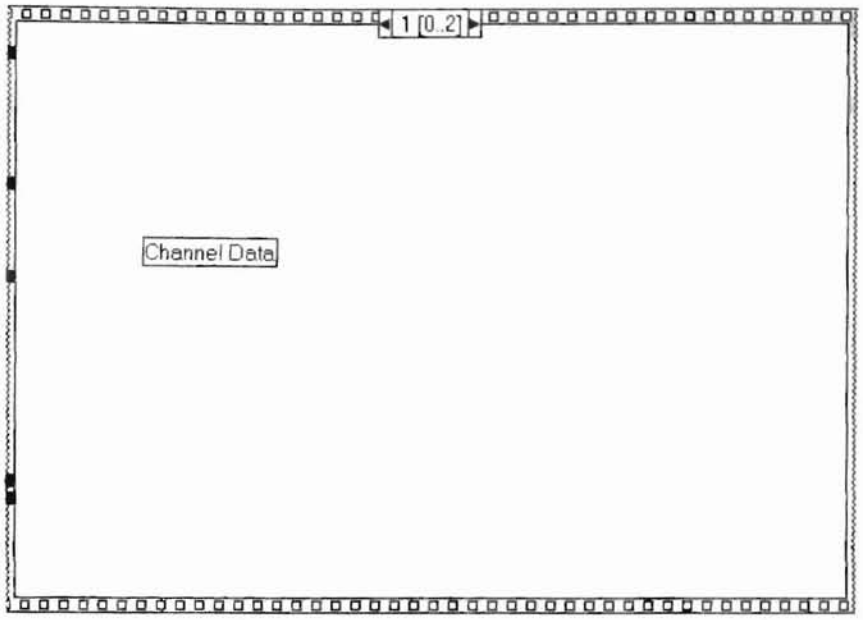


Figure A.7.3: Peak Find VI Sequence 1 Diagram

Sequence 2, shown in figure A.7.4 is executed n times where n is the number of frequency peaks to shift. For each desired peak, the array of data stored by sequence 1 is examined for maximum peaks. Once the peaks are determined, they are stored in an array used as an input to the shifting algorithm described in A.14 of this appendix.

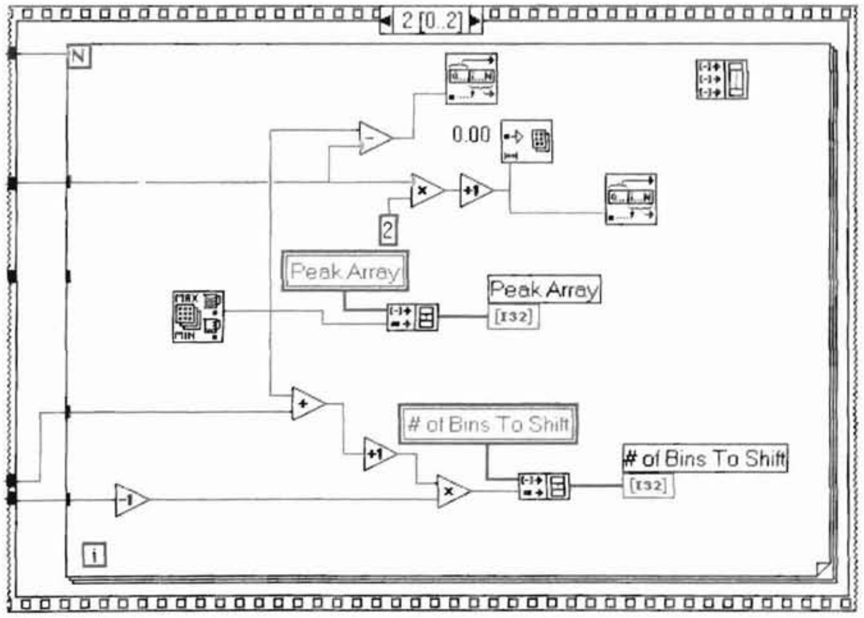


Figure A.7.4: Peak Find VI Sequence 2 Diagram

A.8 Play Back Rate VI: Used to determine the number of samples per second play back required by the wave file. This data is available in the 17th through 20th bytes of data in the wave file header.

Figure A.8.1: Play Back Rate VI Icon

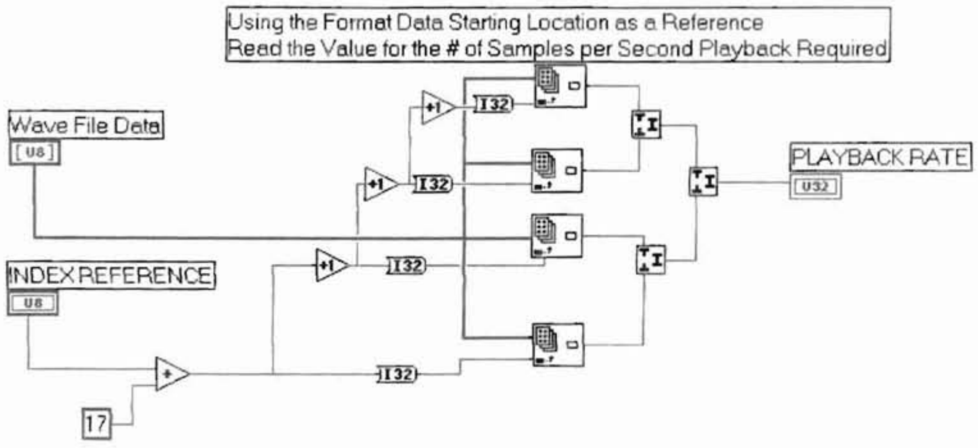
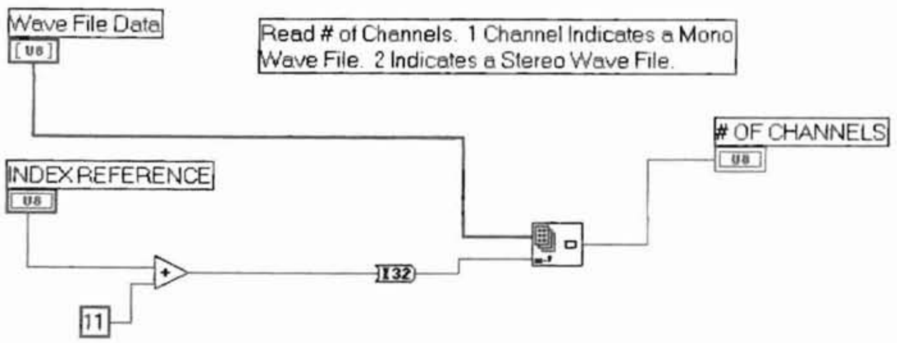

Using the Format Data Starting Location as a Reference
Read the Value for the # of Samples per Second Playback Required
Wave File Data
U8
INDEX REFERENCE
U8
17
+1
I32
PLAYBACK RATE
U32

Figure A.8.2: Play Back Rate VI Diagram

A.9 Number of Channels VI: Used to determine the number of channels in the wave file. This data is available in the 11th byte of data in the wave file header. A one indicates a mono wave file and a 2 indicates a stereo wave file.

Figure A.9.1: Number of Channels VI Icon

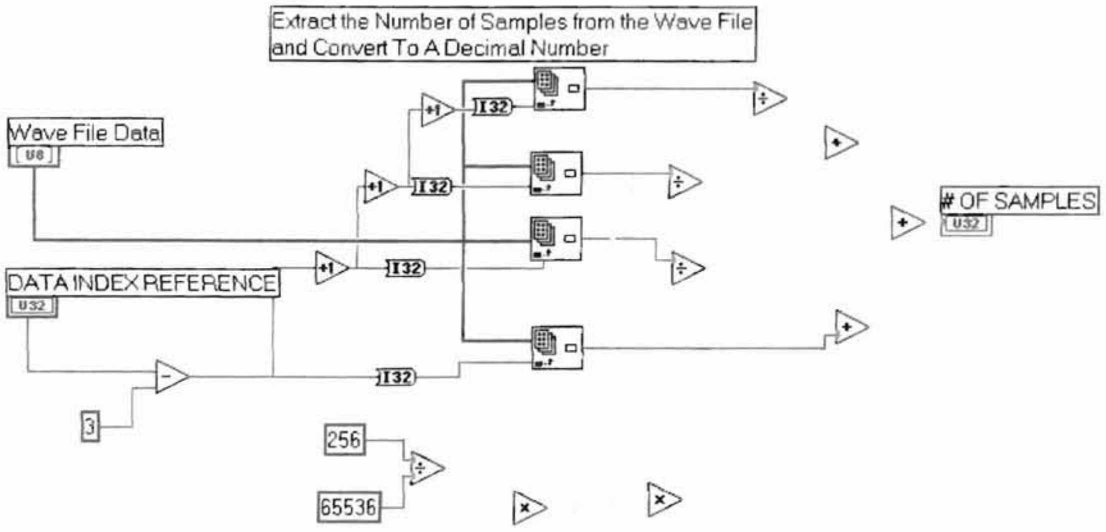


Wave File Data
U8
Read # of Channels. 1 Channel Indicates a Mono Wave File. 2 Indicates a Stereo Wave File.
# OF CHANNELS
U8
INDEX REFERENCE
U8
+
I32
11

Figure A.9.2: Number of Channels VI Diagram

A.10 Number of Samples VI: Used to calculate a decimal value for the number of 8-bit samples located in the wave file. This data is recorded in bytes 3 through 6 of the wave file header.

Figure A.10.1: Number of Samples VI Icon



Extract the Number of Samples from the Wave File and Convert To A Decimal Number
Wave File Data
U8
DATA INDEX REFERENCE
U32
# OF SAMPLES
U32
I32
3
256
65536

Figure A.10.2: Number of Samples VI Diagram

A.11 RIFF File Present? VI: Used to find consecutive appearance of the ASCII letters "RIFF" in the wave file header. The presence of these letters indicates that a wave file has been loaded into memory. It does not verify that valid wave file data is present in the file.

RIFF
FILE?

Figure A.11.1: RIFF File Present? VI Icon

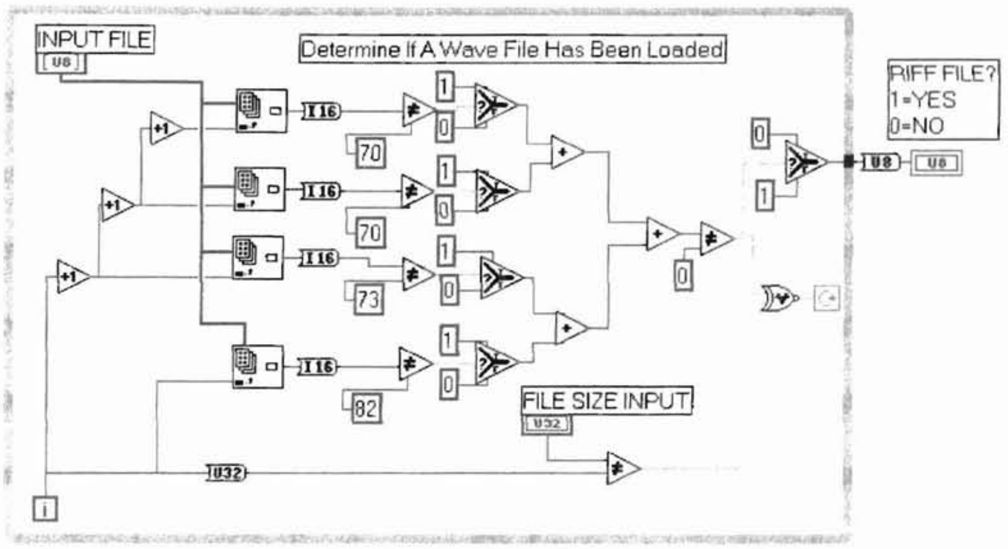


Figure A.11.2: RIFF File Present? VI Diagram

A.12 Sample Rate VI: Used to extract and calculate the decimal value for the sample rate information of the wave file.

SMPL
RATE

Figure A.12.1: Sample Rate VI Icon

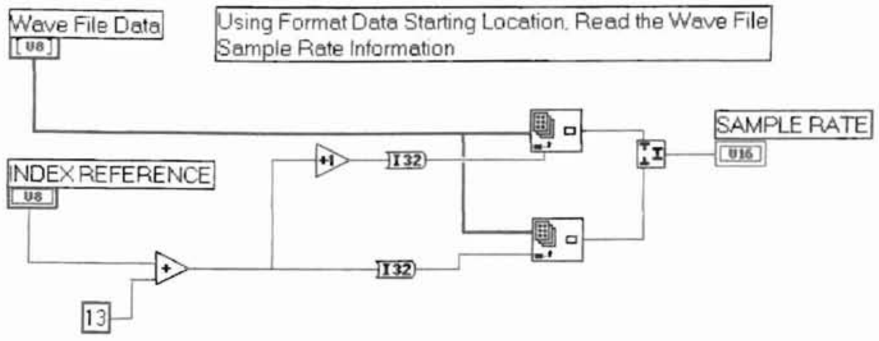

Wave File Data
U8
Using Format Data Starting Location, Read the Wave File Sample Rate Information
INDEX REFERENCE
U8
I32
I32
13
SAMPLE RATE
U16

Figure A.12.2: Sample Rate VI Diagram

A.13 Save Processed File VI: Used to create a wave file after the harmonic generation algorithm has been performed. Figure A.13.2 shows the shifted left and right channel data as input arrays to this VI. In sequence 0, the file is processed to determine the maximum value of any sample. The maximum allowed size for a 16 bit sample is 32767 and for an 8 bit sample is 255. Since the shifted data is a sum of the original and the harmonic, we must ensure that these values are not exceeded. In sequence 0, the maximum value of the input array is determined. All values in the array are divided by this value in order to normalize the wave file data to a maximum value of 1. All data is then multiplied by the scale factor of 32767 for 16 bit files and 255 for 8 bit files. Figures A.13.2 and A.13.3 show this process for the left channel and figures A.13.4 and A.13.5 show this process for the right channel.

Save PF

Figure A.13.1: Save Processed File VI Icon

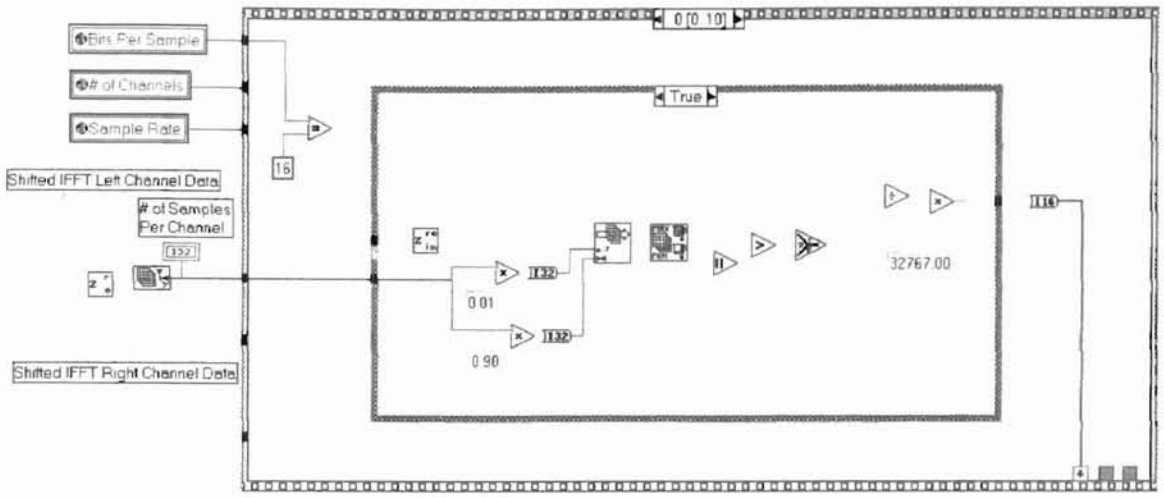


Figure A.13.2: Save Processed File VI Sequence 0, Subsequence True Diagram

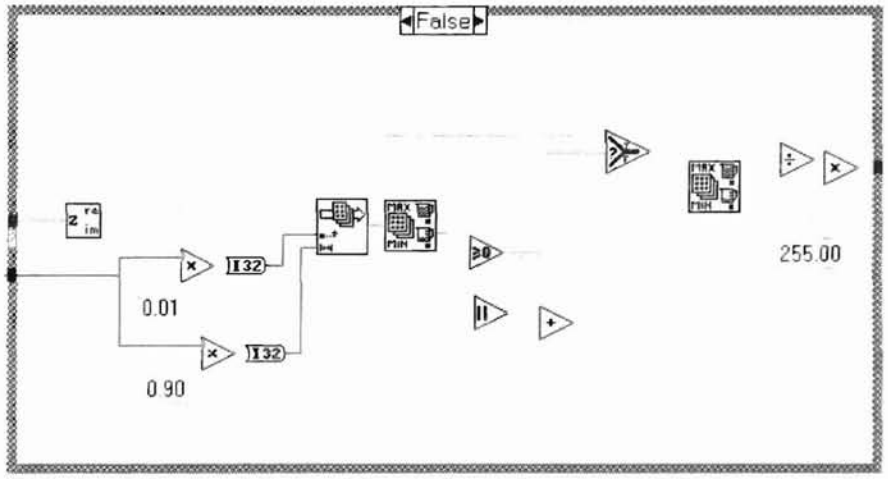


Figure A.13.3: Save Processed File VI Sequence 0, Subsequence False Diagram

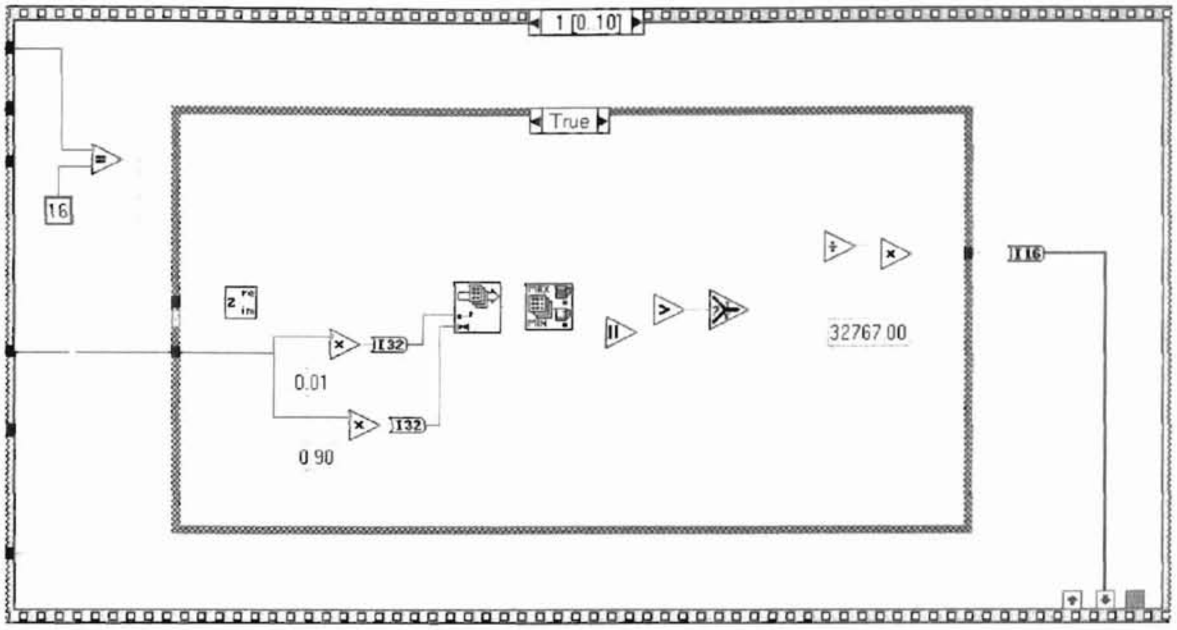


Figure A.13.4: Save Processed File VI Sequence 1, Subsequence True Diagram

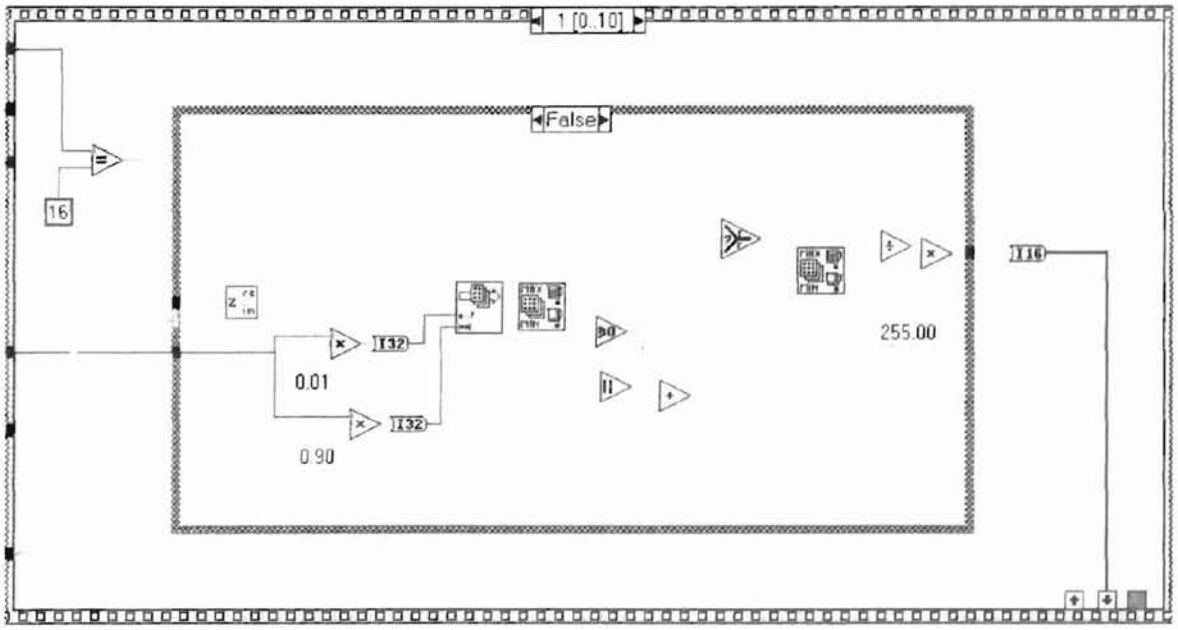


Figure A.13.5: Save Processed File VI Sequence 1, Subsequence False Diagram

Figure A.13.6 shows the interleaving of stereo wave data into a single output array of data. If the wave file has only one channel, then the false case of figure A.13.7 occurs and the data is simply passed through the frame unmodified.

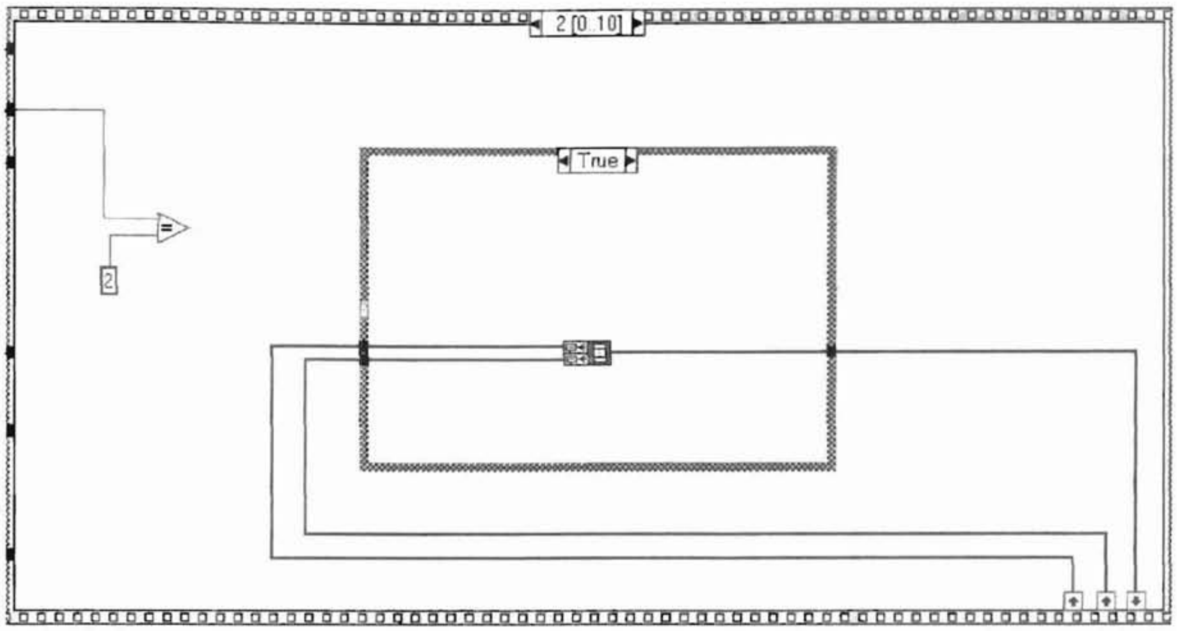


Figure A.13.6: Save Processed File VI Sequence 2, Subsequence True Diagram

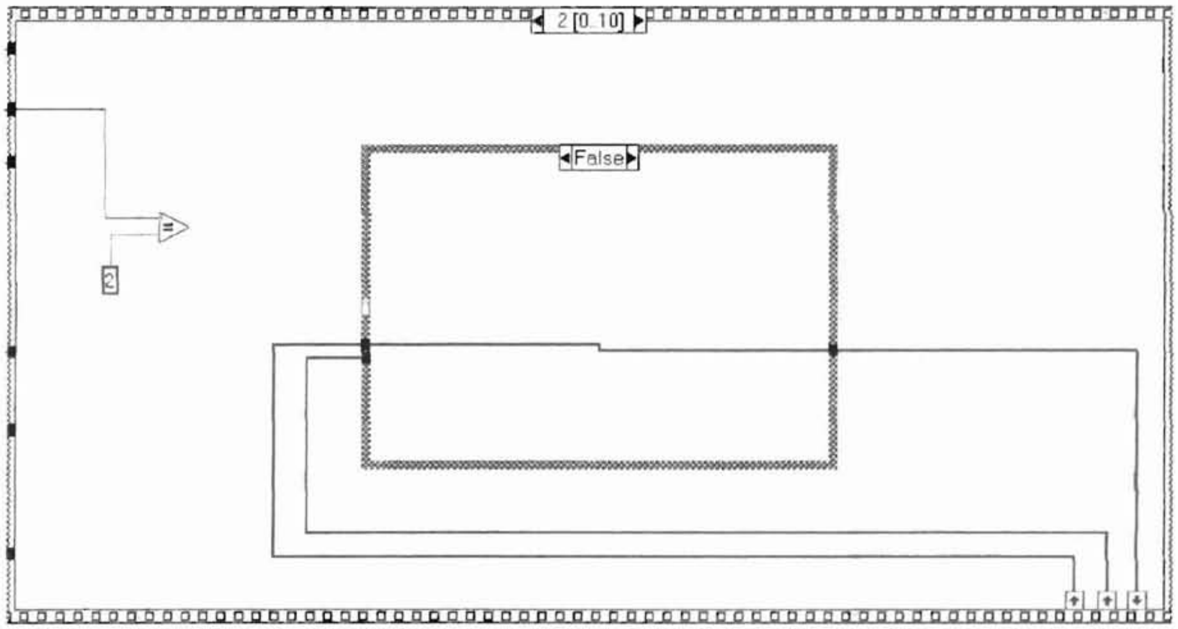


Figure A.13.7: Save Processed File VI Sequence 2, Subsequence False Diagram

The process shown in figure A.13.8 is used when the file contains 16-bit data. The convert VI described in A.2 is applied to the 16-bit data to convert the data into 8-bit format. If the wave file is already in 8-bit format, the process shown in figure A.13.9 is executed and simply ensures that the data is provided to the next stages of this VI in the unsigned 8-bit format.

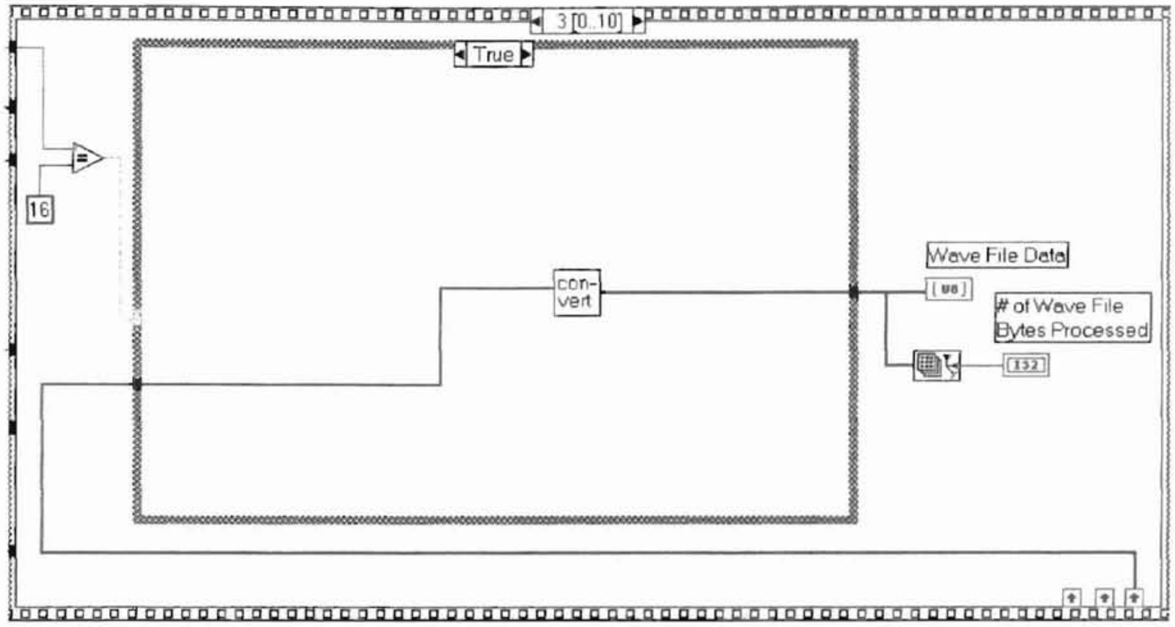


Figure A.13.8: Save Processed File VI Sequence 3, Subsequence True Diagram

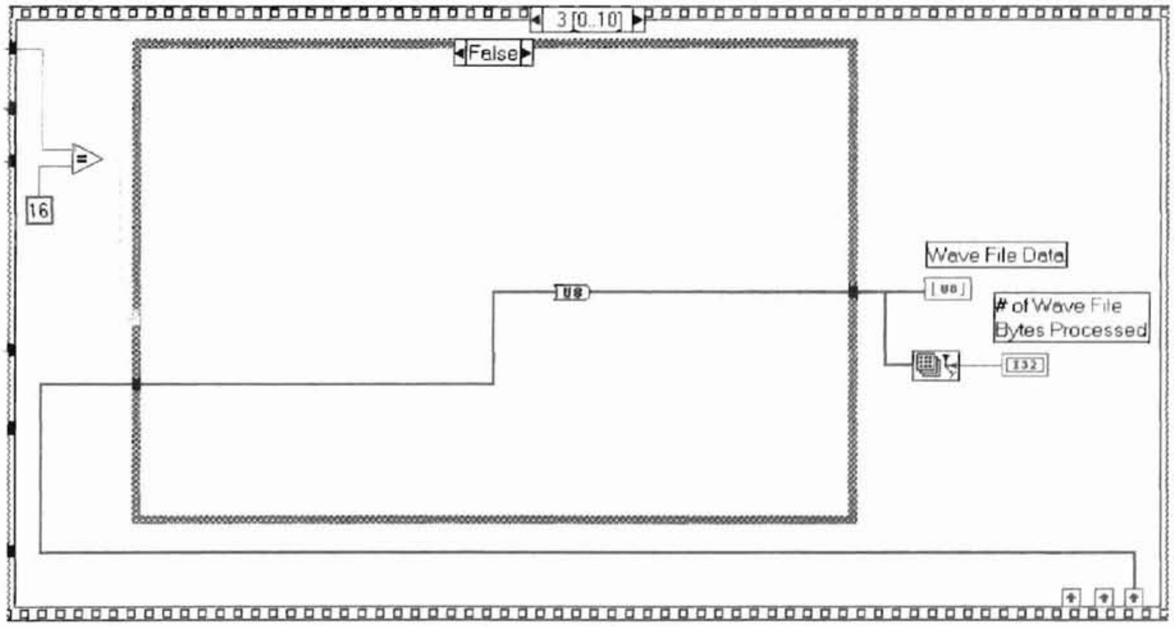


Figure A.13.9: Save Processed File VI Sequence 3, Subsequence False Diagram

Sequence 4 of the VI starts the header file creation process. The data is placed into an array that will become the header file and later appended to the wave file data. Sequence

4 shows the insertion of the ASCII characters "RIFF", the size of the wave file, and the ASCII characters "WAVEfmt" to the header. Subsequence 0 through 3 convert the decimal value of the wave file size to four 8-bit words. Subsequences 1 through 3 are shown in figures A.13.11 through A.13.13.

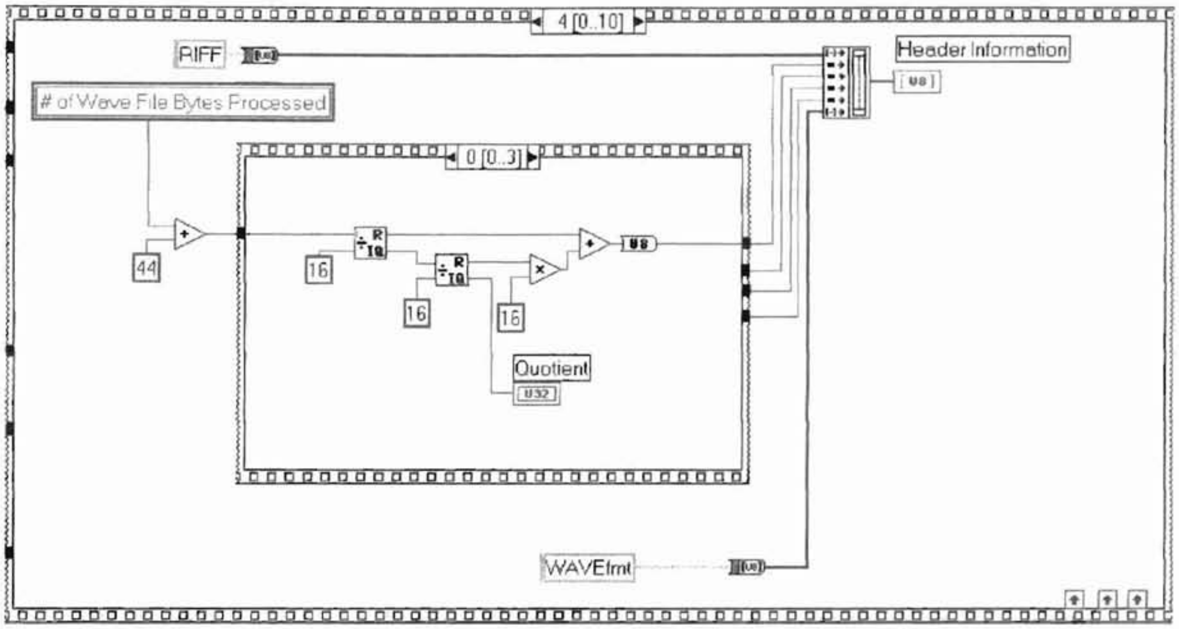


Figure A.13.10: Save Processed File VI Sequence 4, Subsequence 0 Diagram

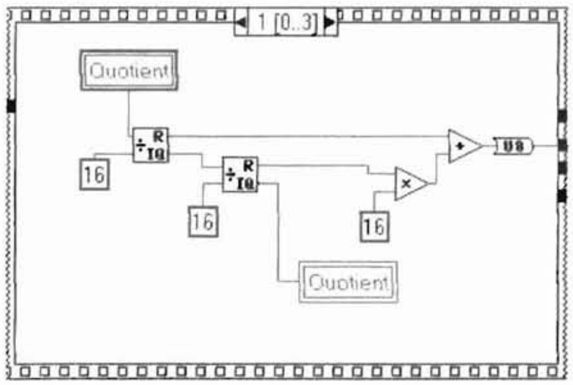


Figure A.13.11: Save Processed File VI Sequence 4, Subsequence 1 Diagram

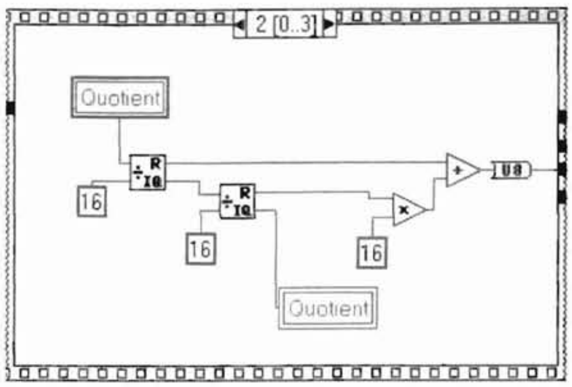


Figure A.13.12: Save Processed File VI Sequence 4, Subsequence 2 Diagram

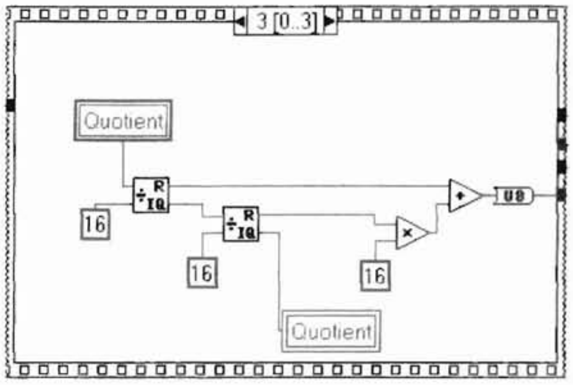


Figure A.13.13: Save Processed File VI Sequence 4, Subsequence 3 Diagram

Sequence 5, Figure A.13.14, shows the encoding of the wave file format and the number of channels into the wave file header. This program saves the data in the Microsoft PCM format.

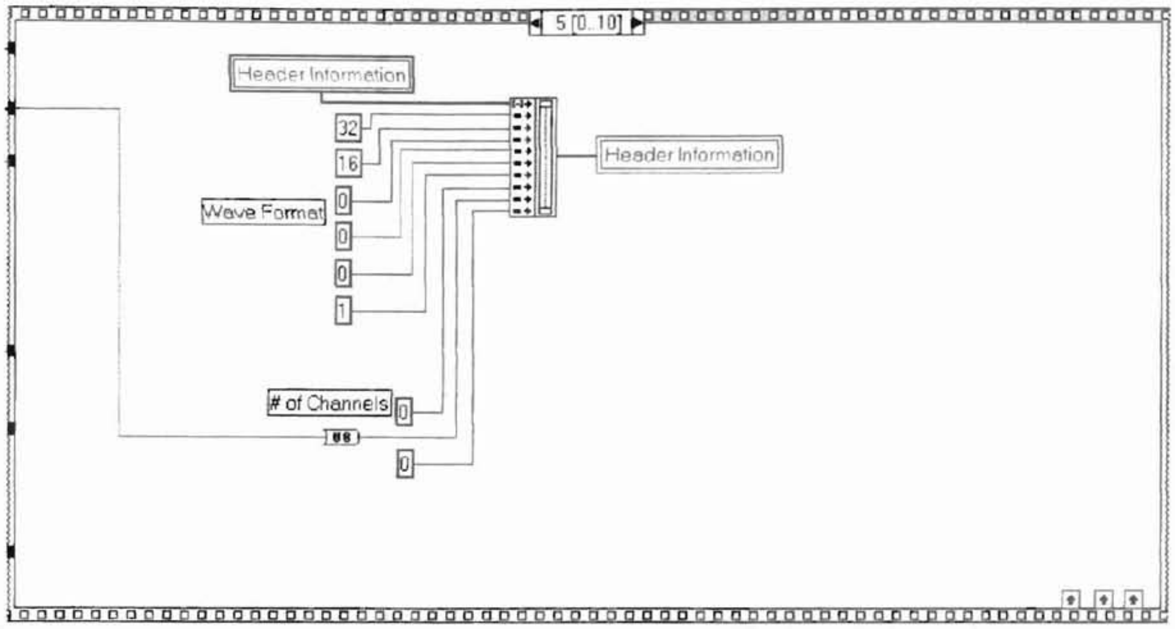


Figure A.13.14: Save Processed File VI Sequence 5 Diagram

Sequence 6, shown in figures A.13.15 through A.13.20, converts the file sample rate from a decimal value to four 8-bit words and encodes them into the header file.

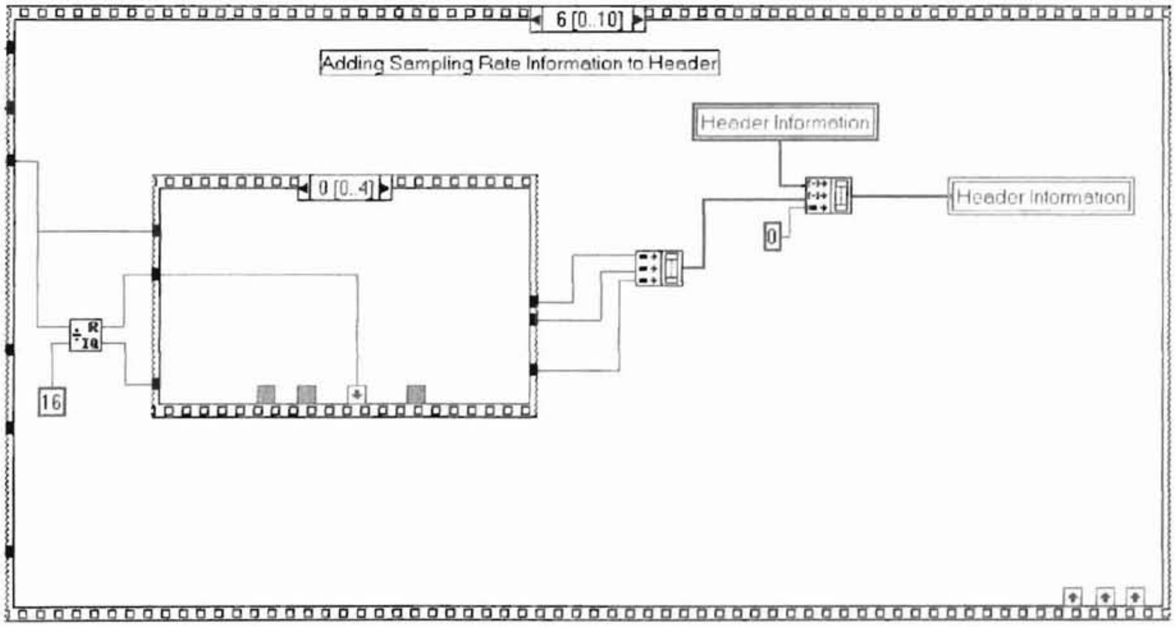


Figure A.13.15: Save Processed File VI Sequence 6, Subsequence 0 Diagram

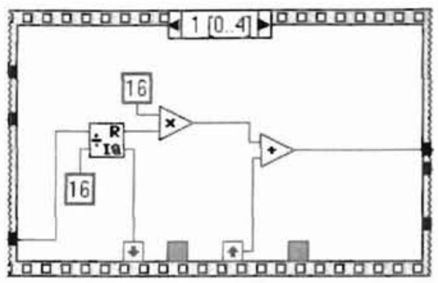


Figure A.13.16: Save Processed File VI Sequence 6, Subsequence 1 Diagram

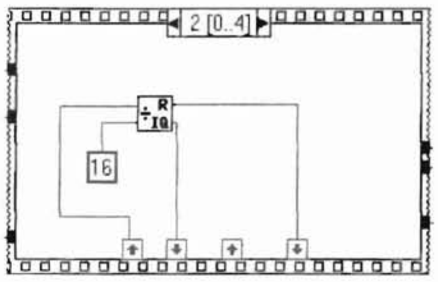


Figure A.13.17: Save Processed File VI Sequence 6, Subsequence 2 Diagram

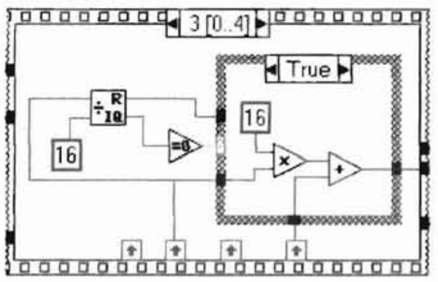


Figure A.13.18: Save Processed File VI Sequence 6, Subsequence 3-True Diagram

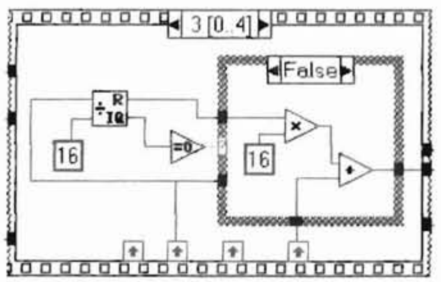


Figure A.13.19: Save Processed File VI Sequence 6, Subsequence 3-False Diagram

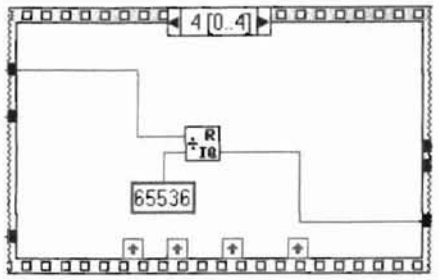


Figure A.13.20: Save Processed File VI Sequence 6, Subsequence 4Diagram

The number of bytes per second playback rate is calculated in VI sequence 7 and encoded into the wave file header. This process is shown in figure A.13.21. Only subsequence 0 is shown since it is identical in format to that of figures A.13.16 through A.13.20 for subsequences 1 through 4.

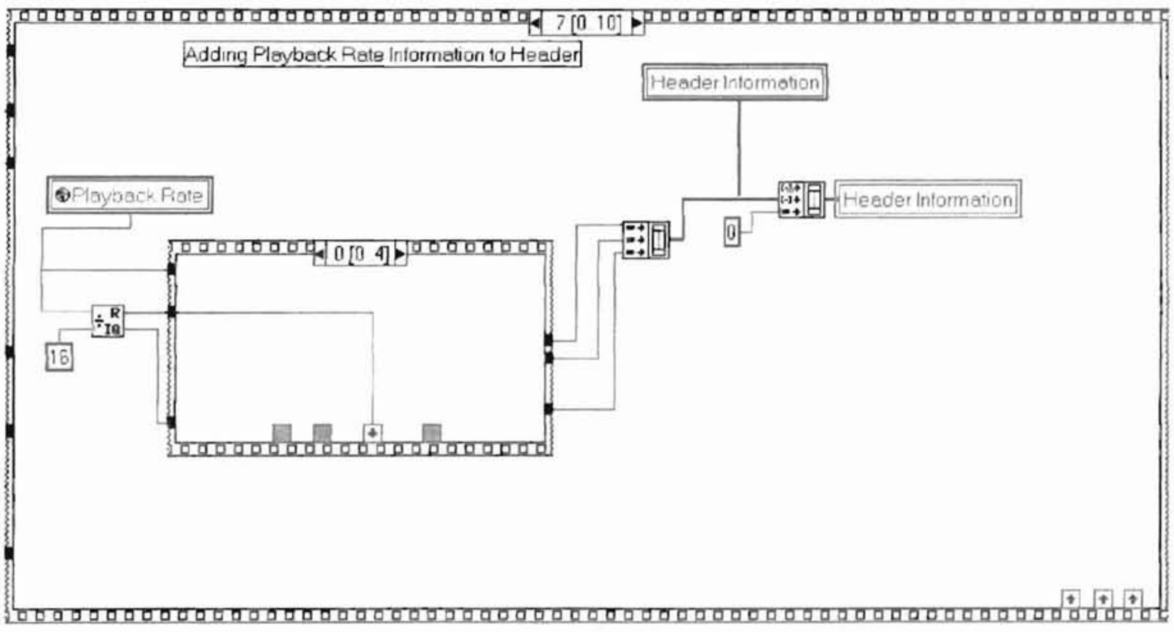


Figure A.13.21: Save Processed File VI Sequence 7, Subsequence 0 Diagram

Sequence 8 shown in figure A.13.22 adds the number of bytes to process at a given time, the number of bits per sample and the ASCII letters "data" to indicate the start of the wave file audio data.

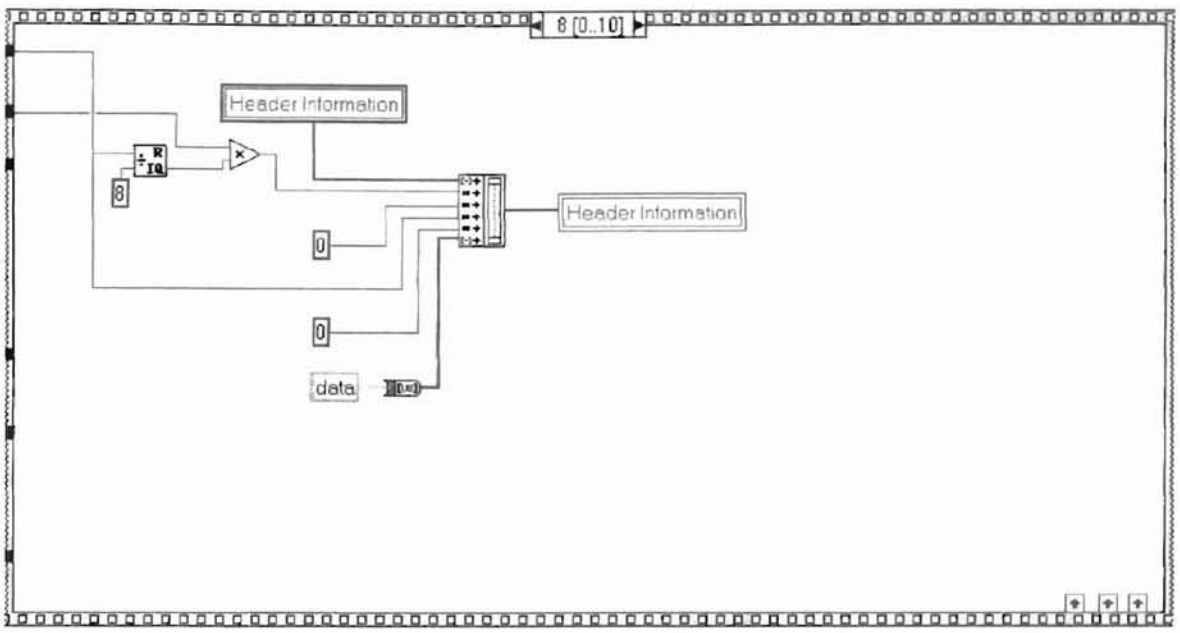


Figure A.13.22: Save Processed File VI Sequence 8 Diagram

Sequence 9, shown in figure A.13.23, records the number of sample bytes found in the audio format of the wave file. It is converted from decimal to 8-bit words using a similar conversion process as that shown in earlier sequences.

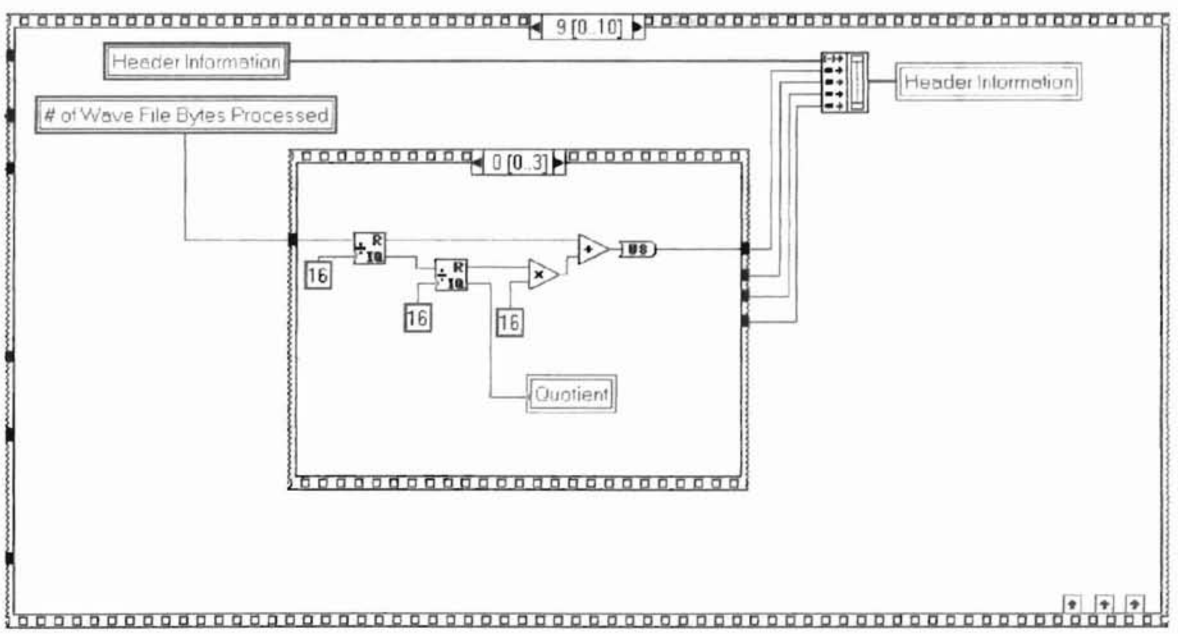


Figure A.13.23: Save Processed File VI Sequence 9, Subsequence 0 Diagram

Sequence 10 is where the wave file header and wave file audio data is combined into one file and saved to disk. The default name of the wave file is shifted.wav, but the operator can select any valid file name. If a file of the same name already exists in the directory of where the file is to be saved, the operator is prompted to confirm the file name. If the operator decides to write over the existing file, the VI deletes the old file first and then saves the new file onto disk. This is shown in figure A.13.24. If a file does not exist with the same file name, the file is created and saved to disk as shown in figure A.13.25.

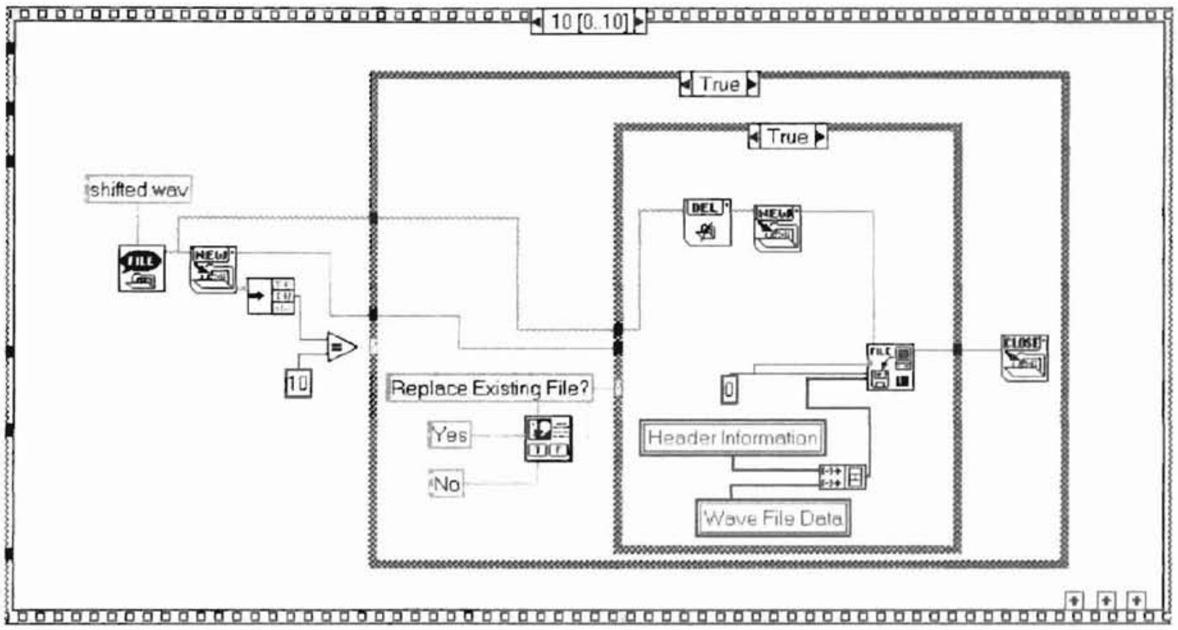


Figure A.13.24: Save Processed File VI Sequence 10, Subsequence True-True Diagram

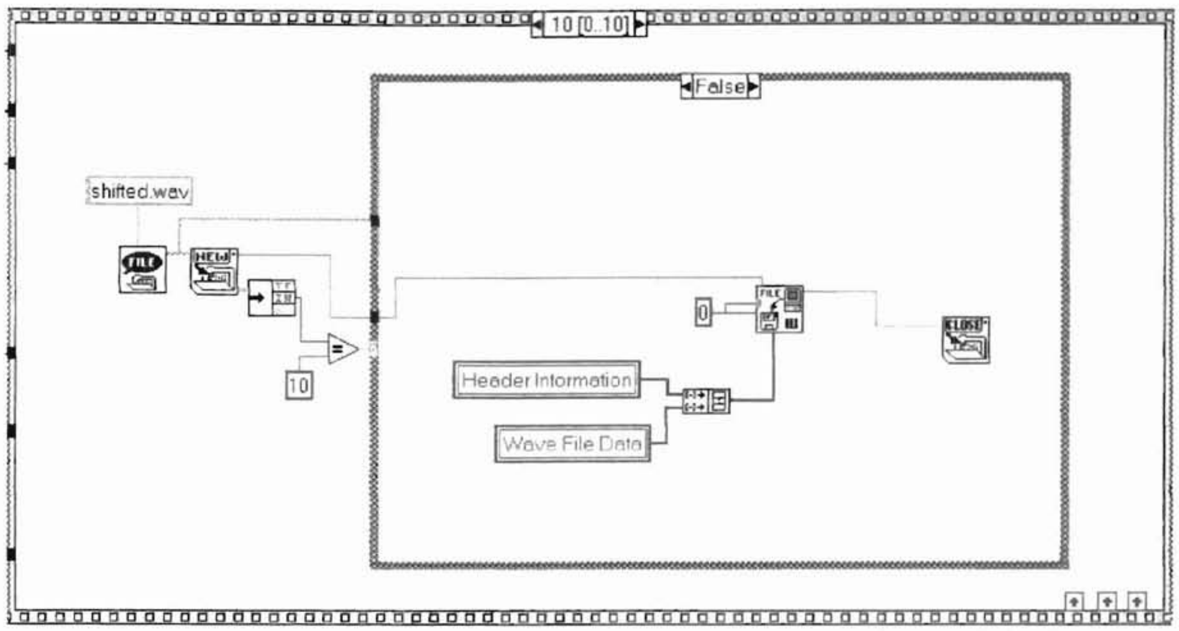


Figure A.13.25: Save Processed File VI Sequence 10, Subsequence True-False Diagram

A.14 Shift Data VI: Used to take the output of the peak finder VI and the input array data from the FFT output for the left and right channel and shift the data in the frequency domain in order to generate the desired harmonics once the IFFT is performed. Sequence 0, shown in figure A.14.2, clears the shifted data array for use.

Figure A.14.1: Shift Data VI Icon

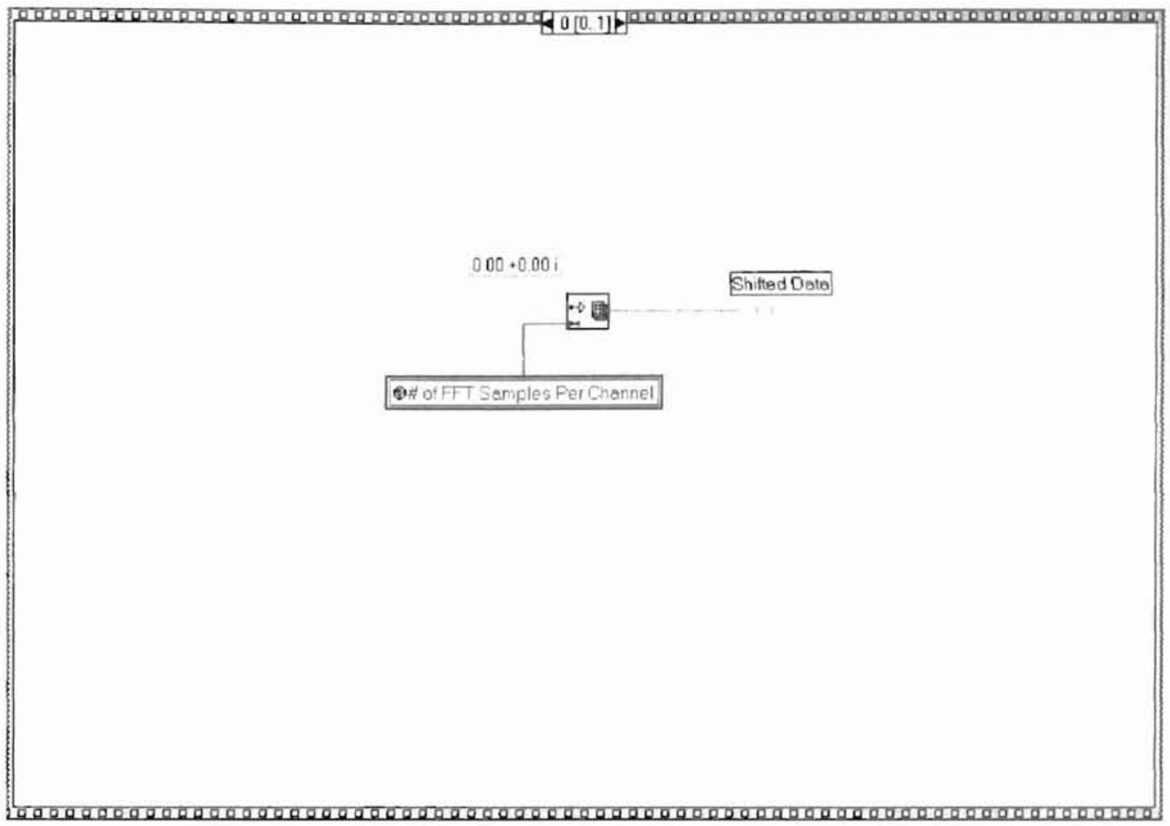


Figure A.14.2: Shift Data VI Sequence 0 Diagram

Sequence 1 is executed N number of times, where N is the number of different harmonic peaks that the operator selected for shifting. This sequence is shown in Figure A.14.3 through A.14.6. Subsequence 0 pads all undesired data points below the desired shifted data starting location with all zeros.

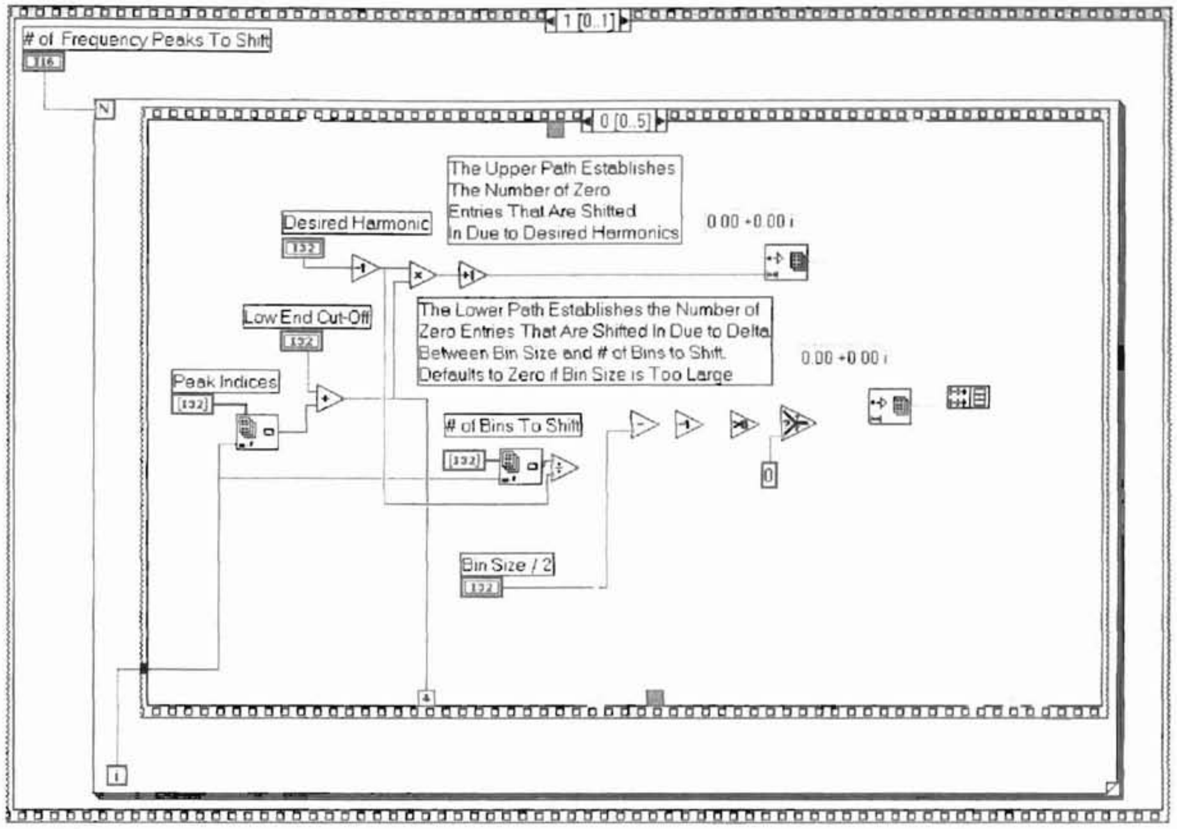

1 [0..1]
# of Frequency Peaks To Shift
I16
N
0 [0..5]
The Upper Path Establishes The Number of Zero Entries That Are Shifted In Due to Desired Harmonics
Desired Harmonic
I32
0.00 +0.00 i
Low End Cut-Off
I32
The Lower Path Establishes the Number of Zero Entries That Are Shifted In Due to Delta Between Bin Size and # of Bins to Shift. Defaults to Zero if Bin Size is Too Large
0.00 +0.00 i
Peak Indices
I32
# of Bins To Shift
I32
0
Bin Size / 2
I32
i

Figure A.14.3: Shift Data VI Sequence 1, Subsequence 0 Diagram

Subsequence 1, figure A.14.4, ensures that the data in array index value 0 is not shifted with the data. This is because the data in bin 0 of the FFT algorithm is the DC component of the signal. Subsequence 2, figure A.14.5 prevents data from being shifted past the Nyquist frequency range.

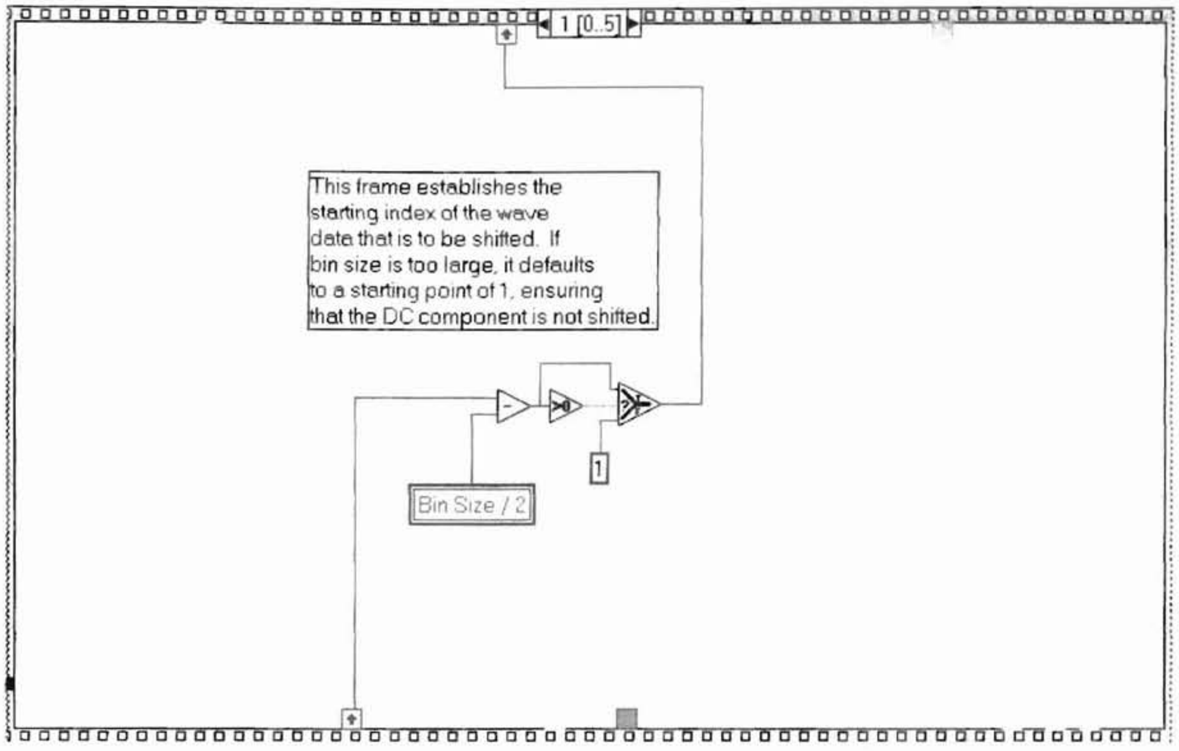

1 [0..5]
This frame establishes the starting index of the wave data that is to be shifted. If bin size is too large, it defaults to a starting point of 1, ensuring that the DC component is not shifted.
1
Bin Size / 2

Figure A.14.4: Shift Data VI Sequence 1, Subsequence 1 Diagram

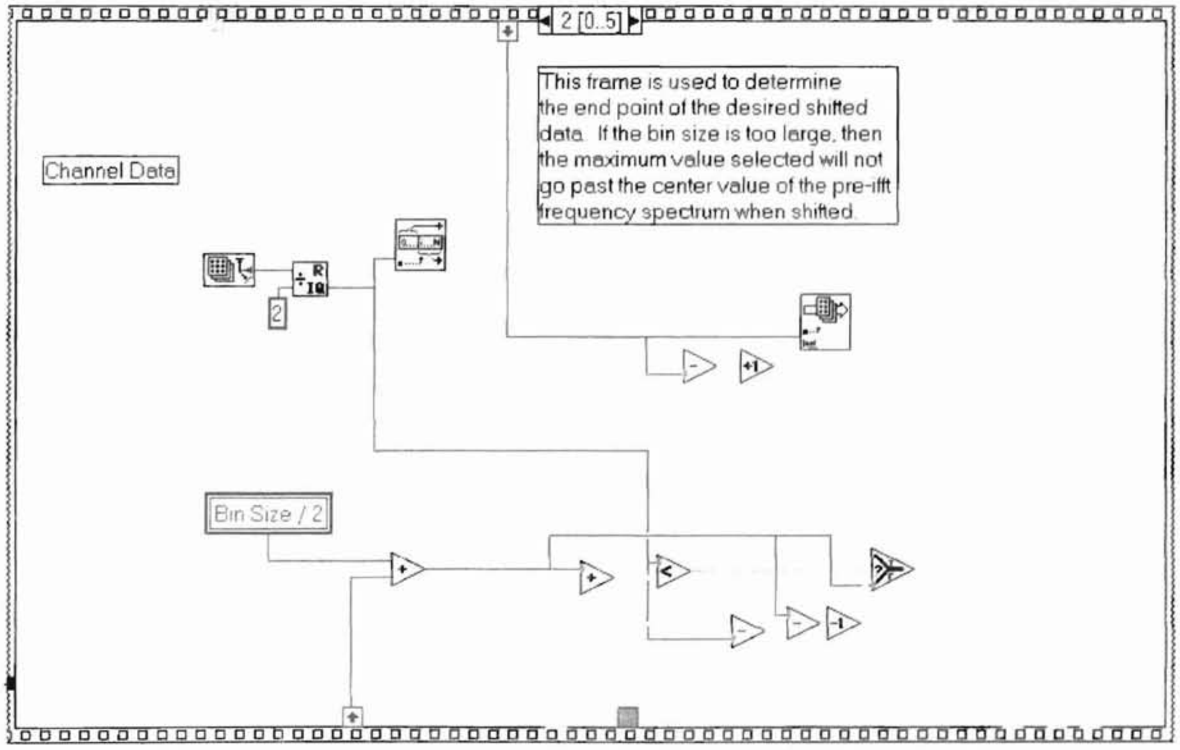

2 [0..5]
This frame is used to determine the end point of the desired shifted data. If the bin size is too large, then the maximum value selected will not go past the center value of the pre-ifft frequency spectrum when shifted.
Channel Data
2
Bin Size / 2

Figure A.14.5: Shift Data VI Sequence 1, Subsequence 2 Diagram

Subsequence 3, figure A.14.6, takes the desired data to be shifted and rotates the phase of each point by (N-1)*the phase of the peak bin, where N is the desired harmonic. Subsequence 4, figure A.14.7, fills the remaining half of the frequency spectrum with all zeros. It then calculates the complex conjugate of the first half of the data, reverses it and combines all data together to create a complete frequency spectra over the full sample rate for correct processing by the IFFT algorithm. Subsequence 5, figure A.14.8, creates the array that is used by the IFFT algorithm and allows each subsequent process of shifting data to be added together into one array.

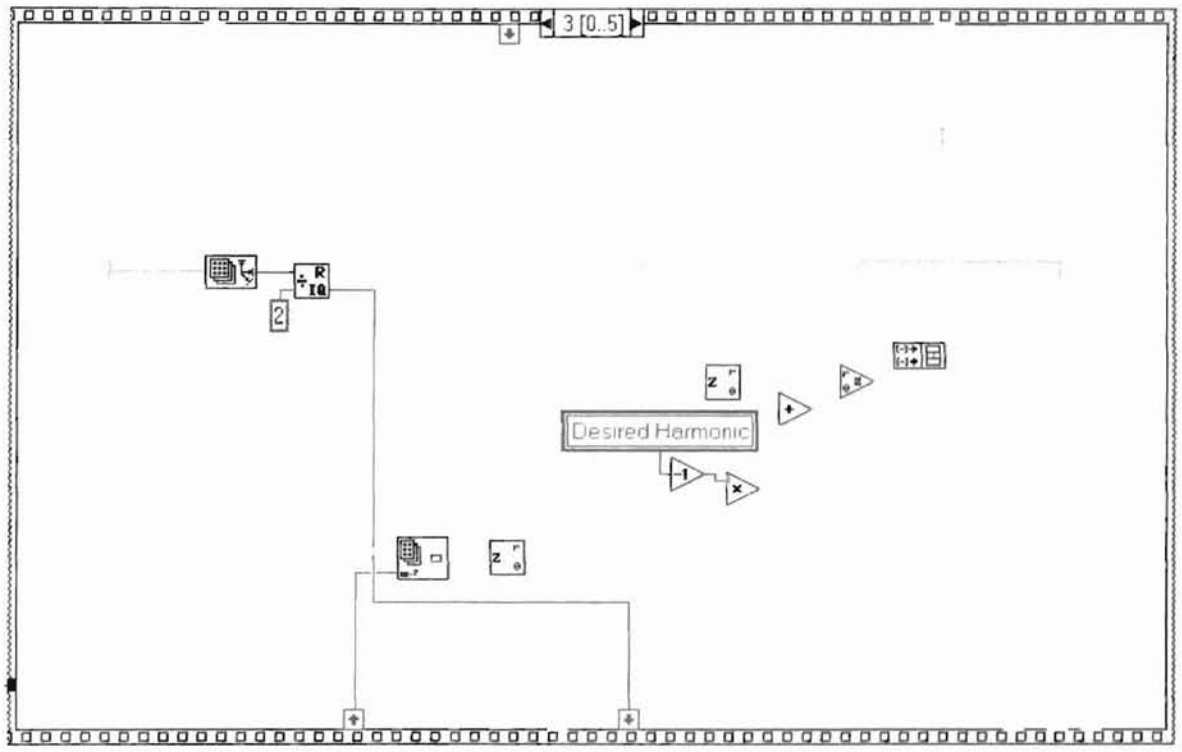


Figure A.14.6: Shift Data VI Sequence 1, Subsequence 3 Diagram

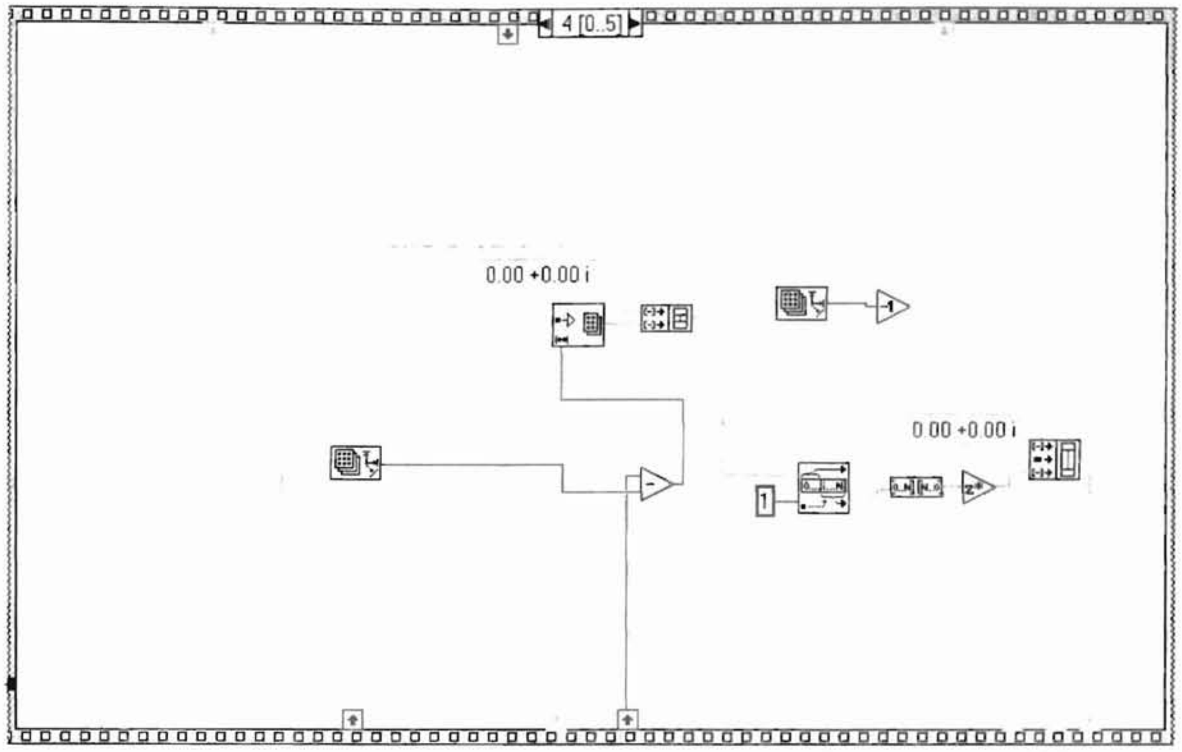


Figure A.14.7: Shift Data VI Sequence 1, Subsequence 4 Diagram

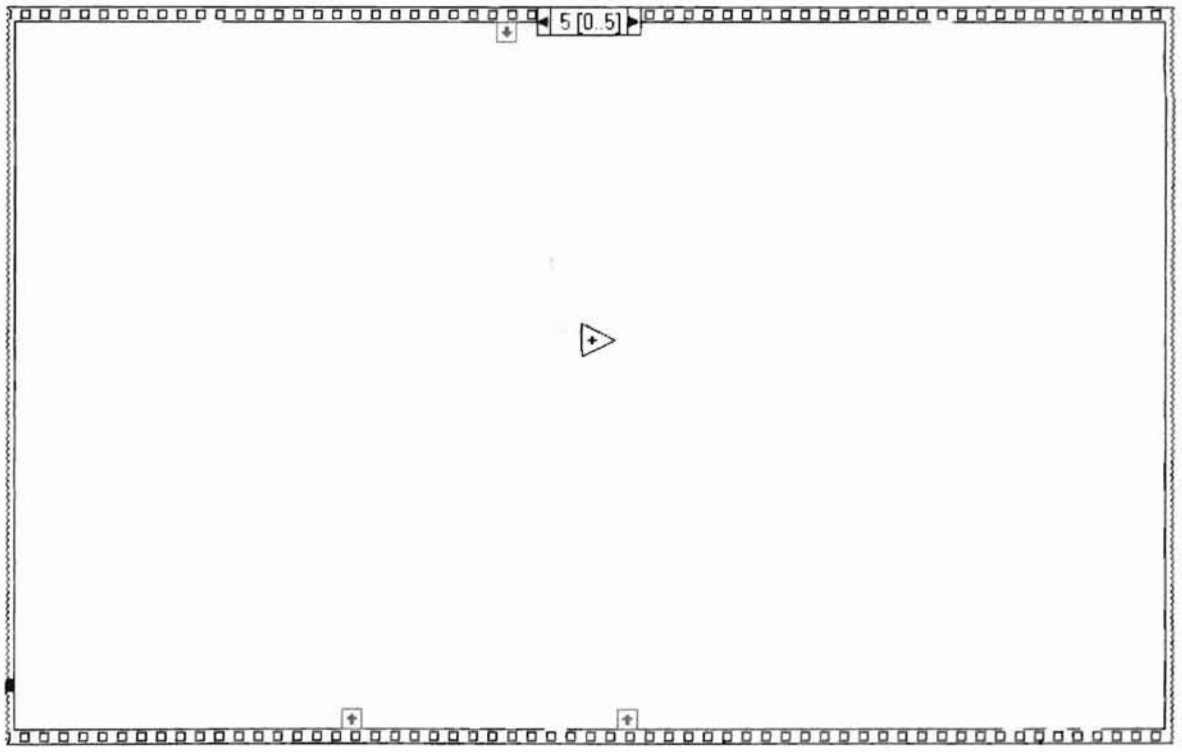


Figure A.14.8: Shift Data VI Sequence 1, Subsequence 5 Diagram

A.15 Split Data VI: Used to split the data into left and right channel if stereo for processing. It also splits the data in the needed order if the file is 8-bit or 16-bit. Sequence 0 initializes the left and right channel arrays and sequence 1 splits the data as required. See figures A.15.2 and A.15.3.

Figure A.15.1: Split Data VI Icon

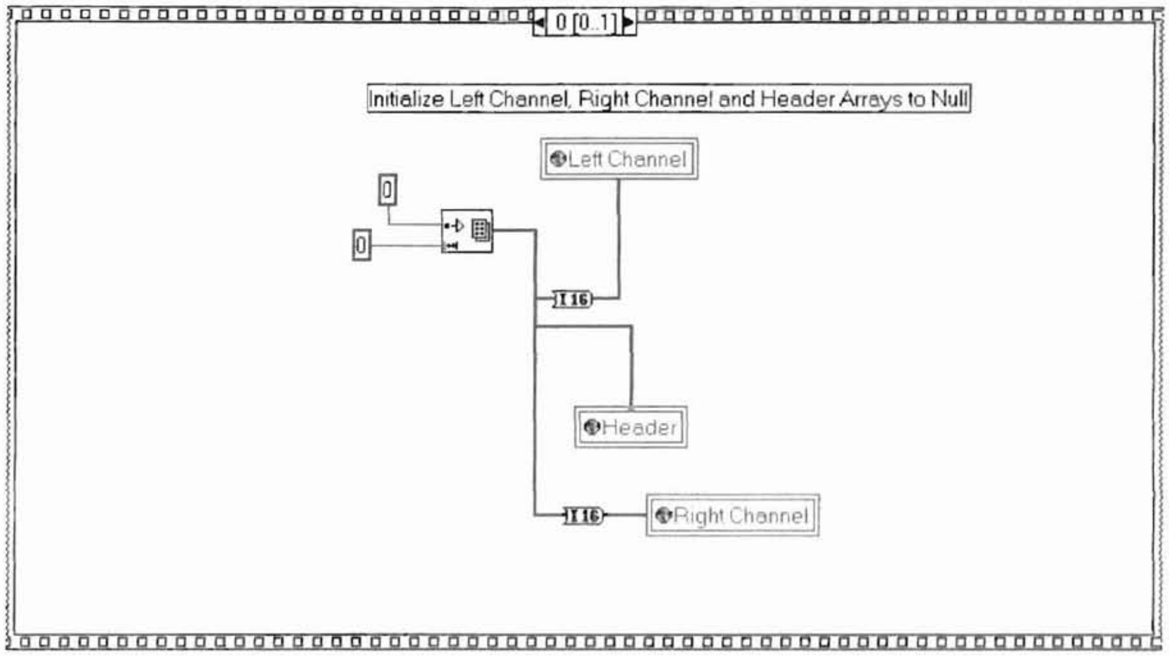

0 [0..1]
Initialize Left Channel, Right Channel and Header Arrays to Null
Left Channel
0
0
I16
Header
I16
Right Channel

Figure A.15.2: Split Data VI Sequence 0 Diagram

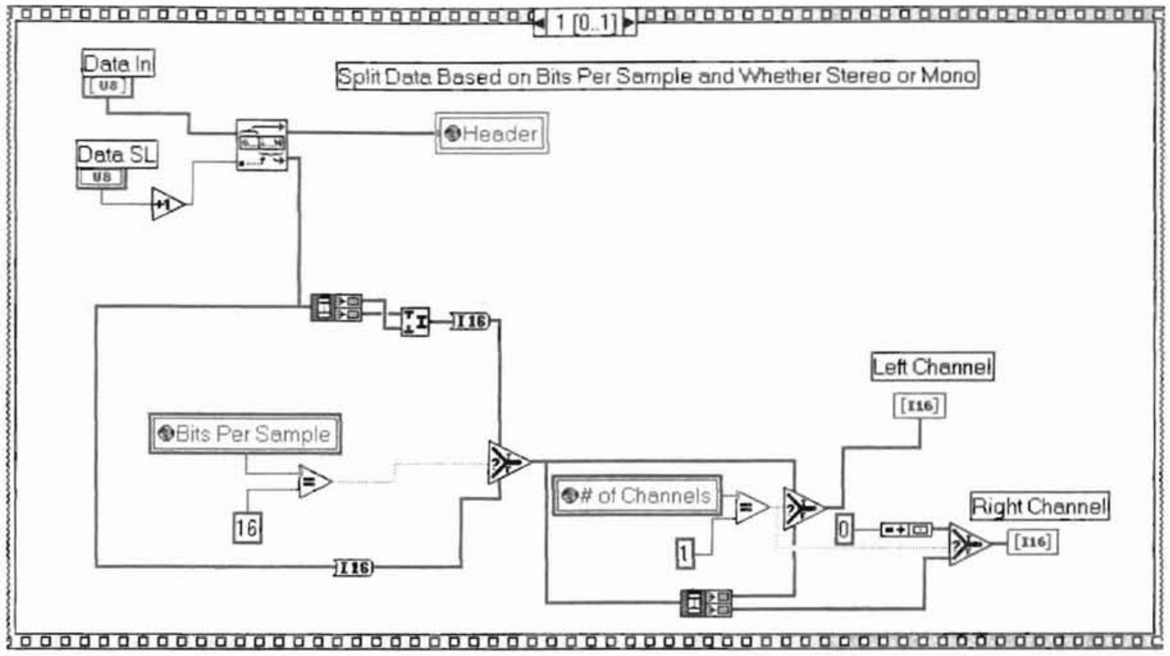
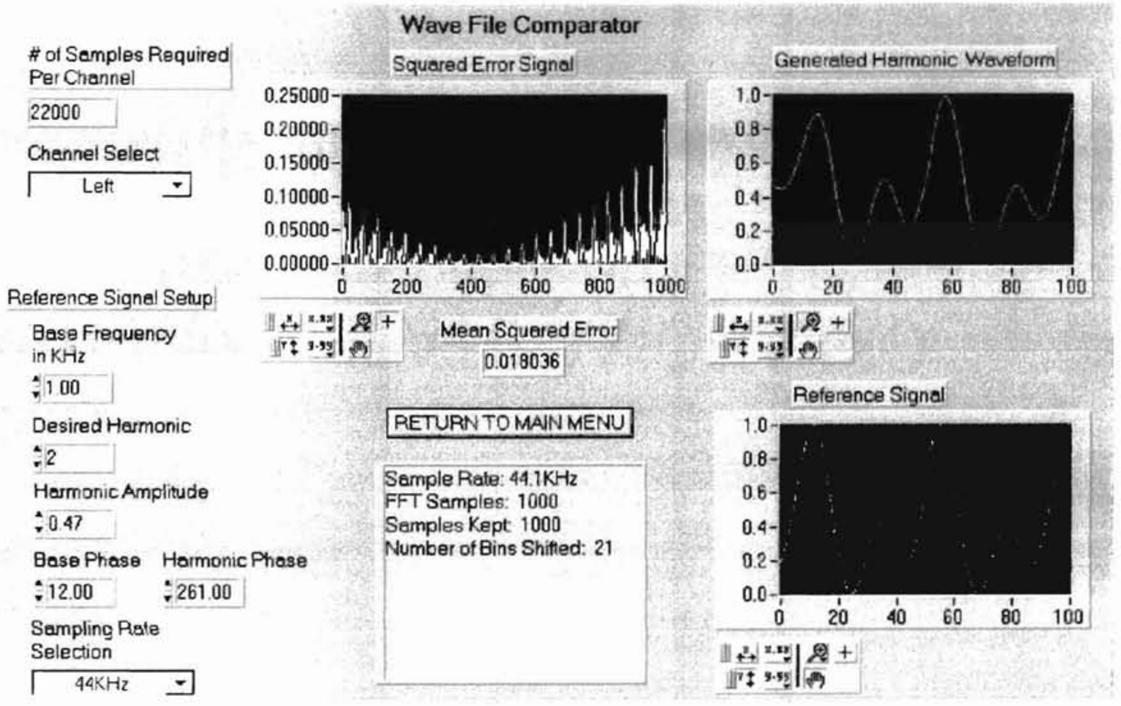


Figure A.15.3: Split Data VI Sequence 1 Diagram

# APPENDIX B

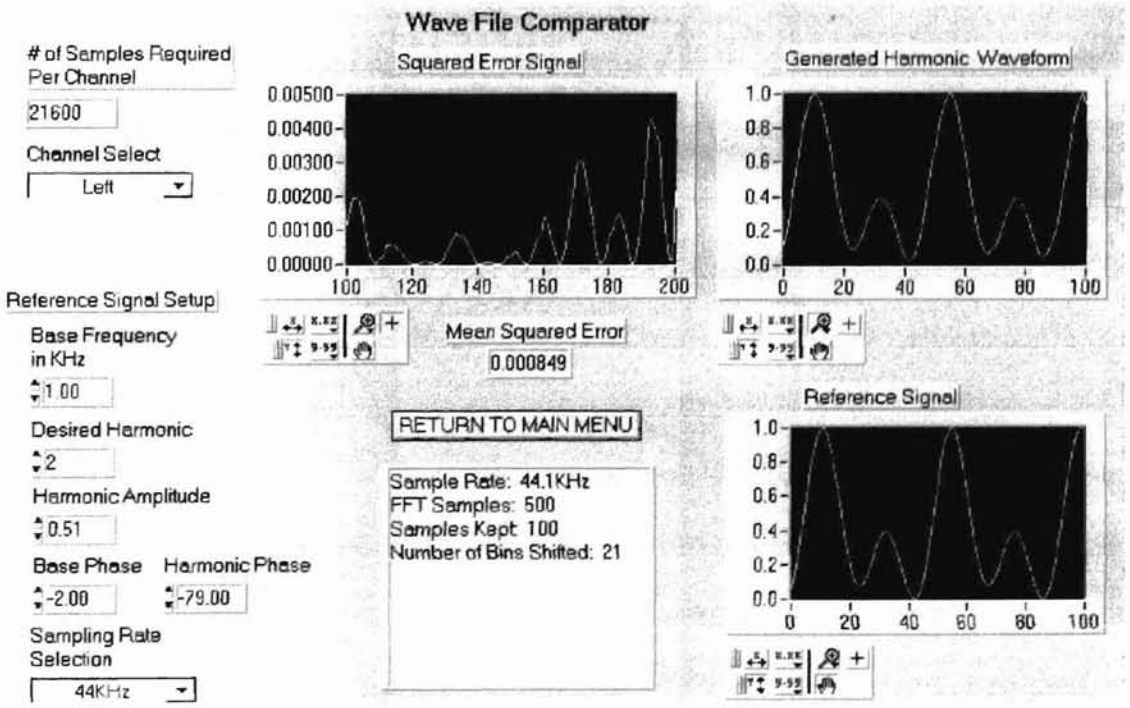
## TEST RUN GRAPHICAL RESULTS



# of Samples Required Per Channel: 22000
Channel Select: Left
Base Frequency in KHz: 1.00
Desired Harmonic: 2
Harmonic Amplitude: 0.47
Base Phase: 12.00
Harmonic Phase: 261.00
Sampling Rate Selection: 44KHz
Mean Squared Error: 0.018036

Sample Rate: 44.1KHz
FFT Samples: 1000
Samples Kept: 1000
Number of Bins Shifted: 21

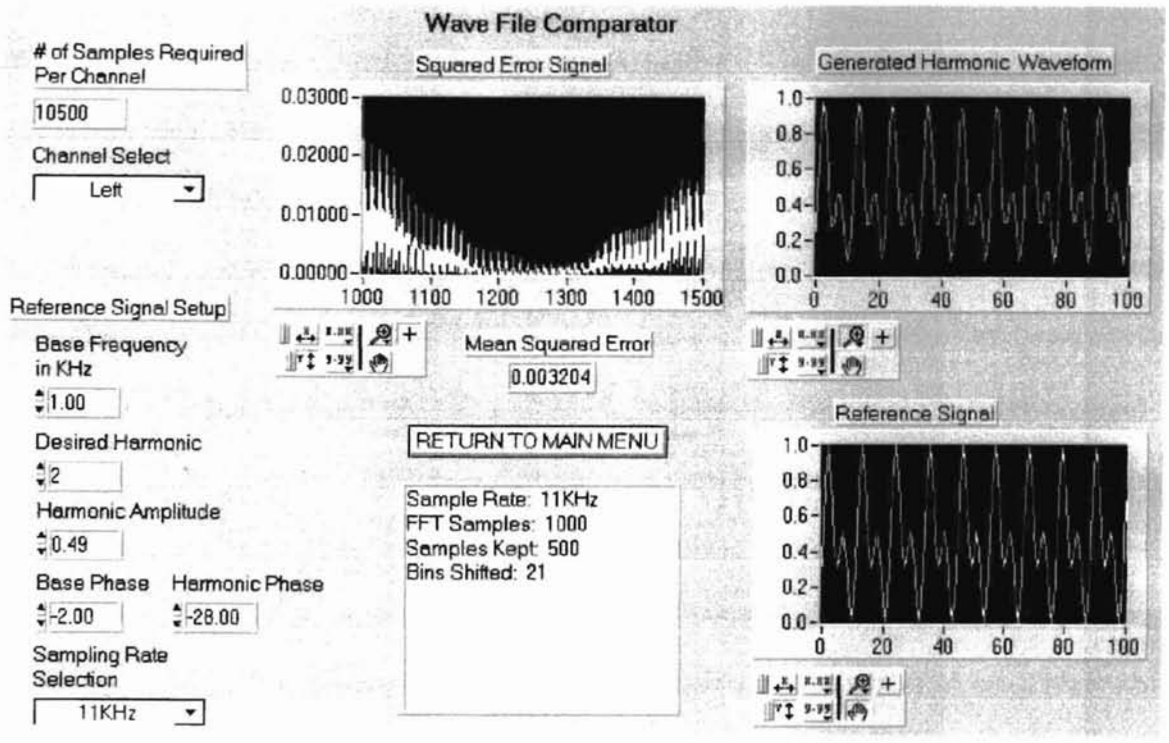
Figure B1: Test Run 1 Results



# of Samples Required Per Channel: 21600
Channel Select: Left
Base Frequency in KHz: 1.00
Desired Harmonic: 2
Harmonic Amplitude: 0.51
Base Phase: -2.00
Harmonic Phase: -79.00
Sampling Rate Selection: 44KHz
Mean Squared Error: 0.000849

Sample Rate: 44.1KHz
FFT Samples: 500
Samples Kept: 100
Number of Bins Shifted: 21

Figure B2: Test Run 7 Results



# of Samples Required Per Channel: 10500

Channel Select: Left

Reference Signal Setup

Base Frequency in KHz: 1.00

Desired Harmonic: 2

Harmonic Amplitude: 0.49

Base Phase: -2.00

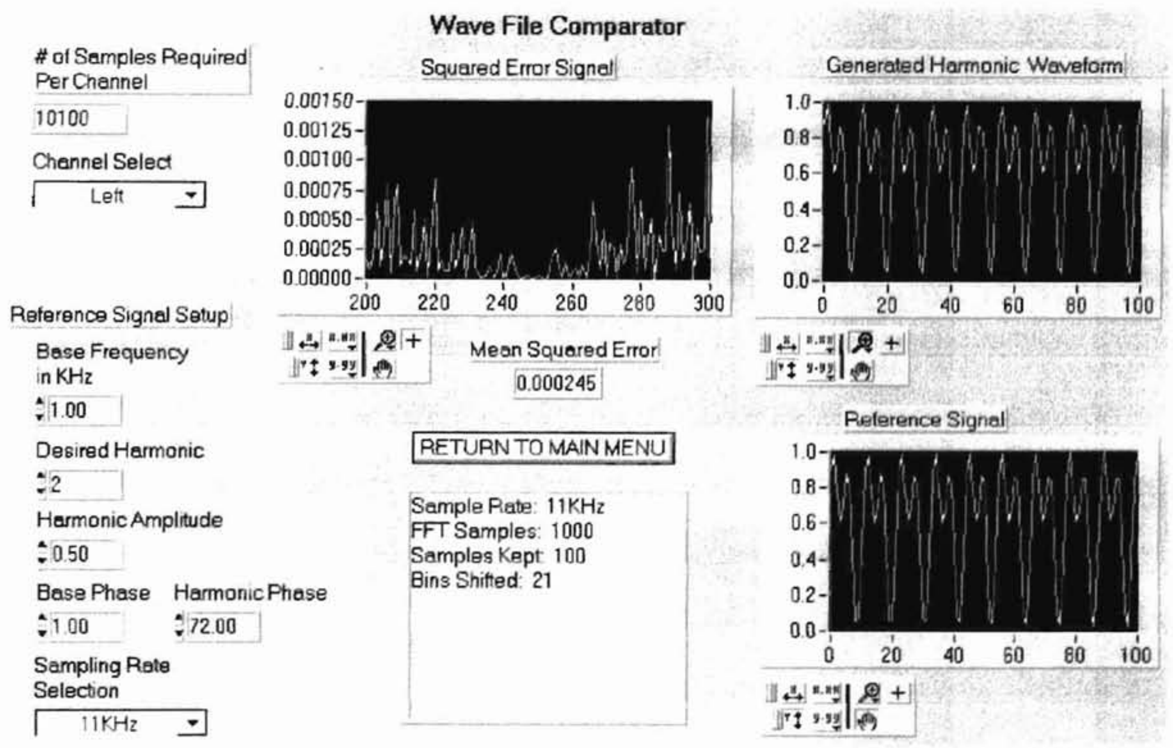
Harmonic Phase: -28.00

Sampling Rate Selection: 11KHz

Mean Squared Error: 0.003204

Sample Rate: 11KHz
FFT Samples: 1000
Samples Kept: 500
Bins Shifted: 21

Figure B3: Test Run 12 Results



# of Samples Required Per Channel: 10100

Channel Select: Left

Reference Signal Setup

Base Frequency in KHz: 1.00

Desired Harmonic: 2

Harmonic Amplitude: 0.50

Base Phase: 1.00

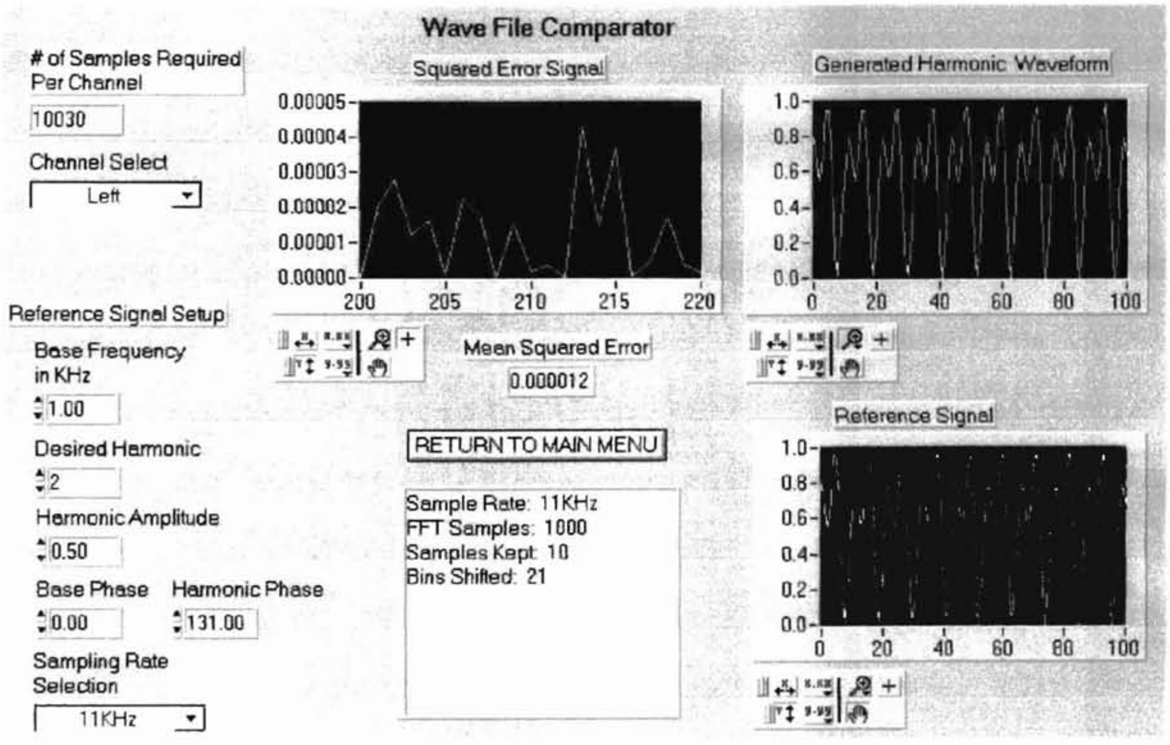
Harmonic Phase: 72.00

Sampling Rate Selection: 11KHz

Mean Squared Error: 0.000245

Sample Rate: 11KHz
FFT Samples: 1000
Samples Kept: 100
Bins Shifted: 21

Figure B4: Test Run 13 Results



# of Samples Required Per Channel: 10030

Channel Select: Left

Reference Signal Setup

Base Frequency in KHz: 1.00

Desired Harmonic: 2

Harmonic Amplitude: 0.50

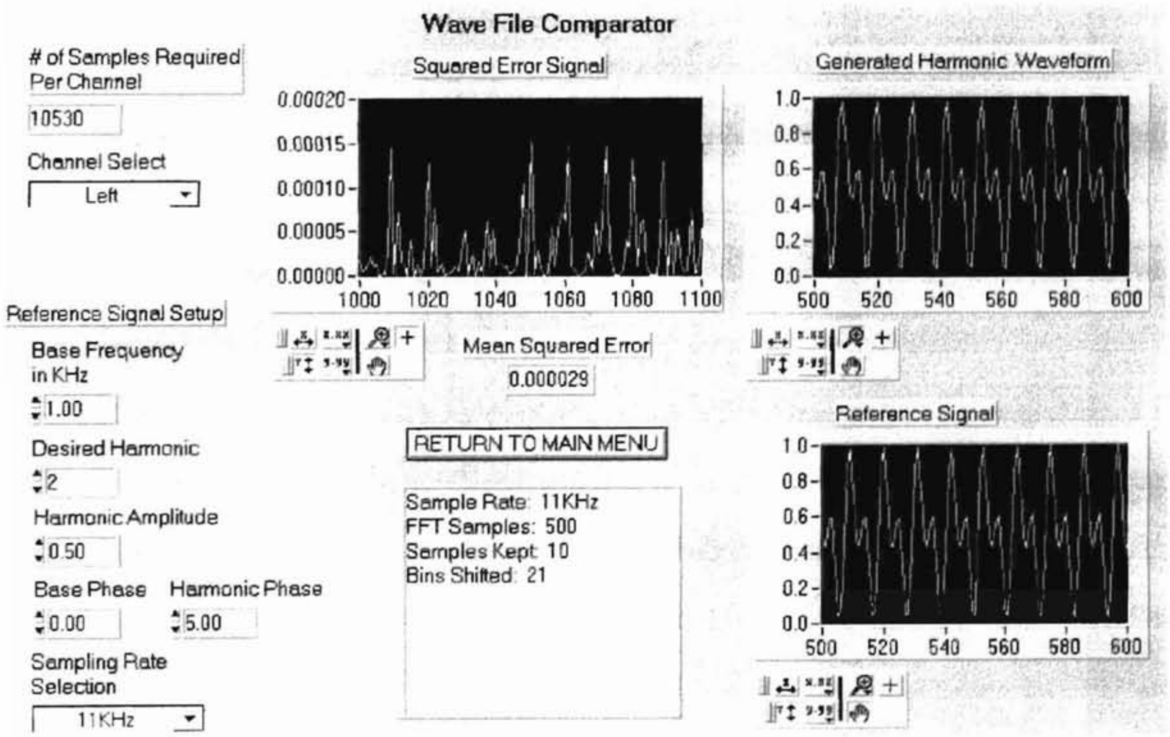
Base Phase: 0.00 Harmonic Phase: 131.00

Sampling Rate Selection: 11KHz

Mean Squared Error: 0.000012

Sample Rate: 11KHz
FFT Samples: 1000
Samples Kept: 10
Bins Shifted: 21

Figure B5: Test Run 14 Results



# of Samples Required Per Channel: 10530

Channel Select: Left

Reference Signal Setup

Base Frequency in KHz: 1.00

Desired Harmonic: 2

Harmonic Amplitude: 0.50

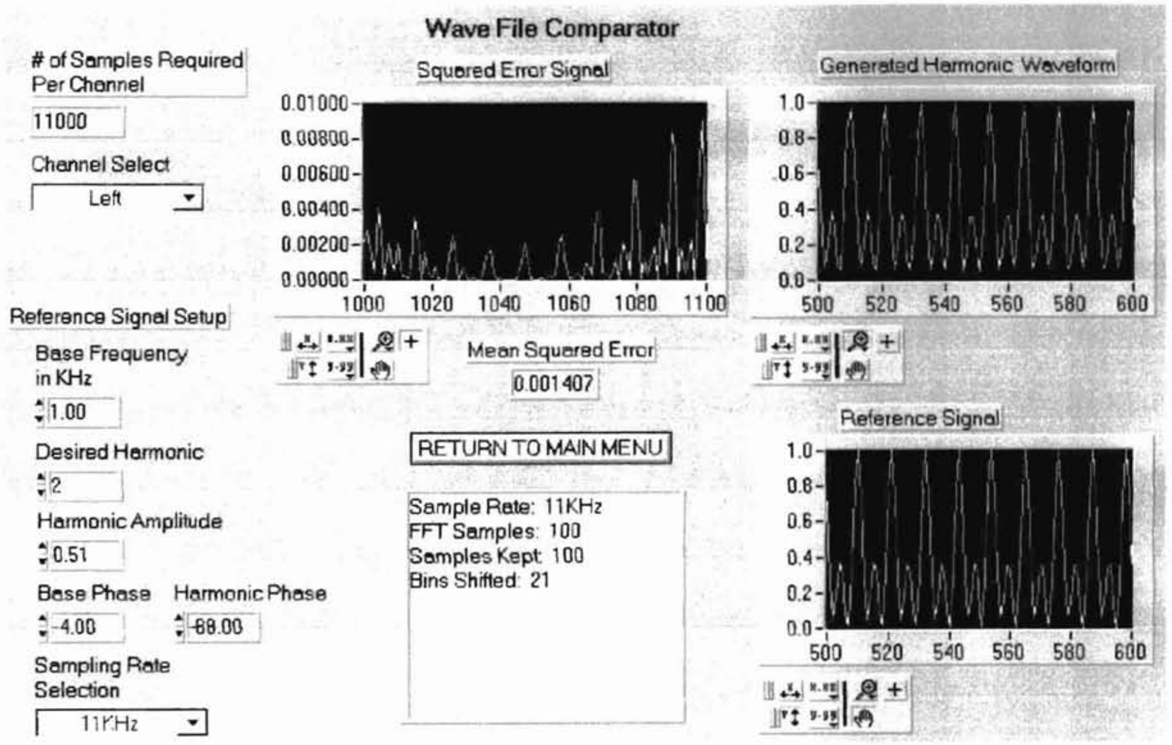
Base Phase: 0.00 Harmonic Phase: 5.00

Sampling Rate Selection: 11KHz

Mean Squared Error: 0.000029

Sample Rate: 11KHz
FFT Samples: 500
Samples Kept: 10
Bins Shifted: 21

Figure B6: Test Run 17 Results



Wave File Comparator

# of Samples Required Per Channel: 11000

Channel Select: Left

Reference Signal Setup

Base Frequency in KHz: 1.00

Desired Harmonic: 2

Harmonic Amplitude: 0.51

Base Phase: -4.00

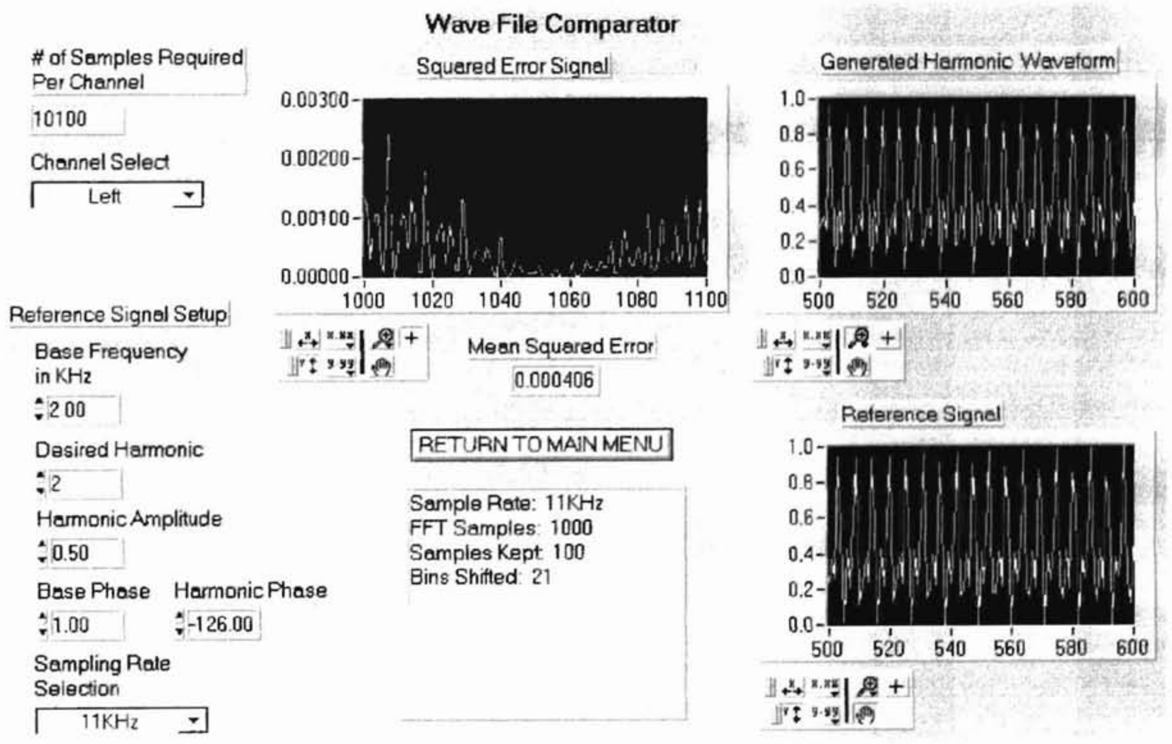
Harmonic Phase: -88.00

Sampling Rate Selection: 11KHz

Mean Squared Error: 0.001407

Sample Rate: 11KHz
FFT Samples: 100
Samples Kept: 100
Bins Shifted: 21

Figure B7: Test Run 18 Results



Wave File Comparator

# of Samples Required Per Channel: 10100

Channel Select: Left

Reference Signal Setup

Base Frequency in KHz: 2.00

Desired Harmonic: 2

Harmonic Amplitude: 0.50

Base Phase: 1.00

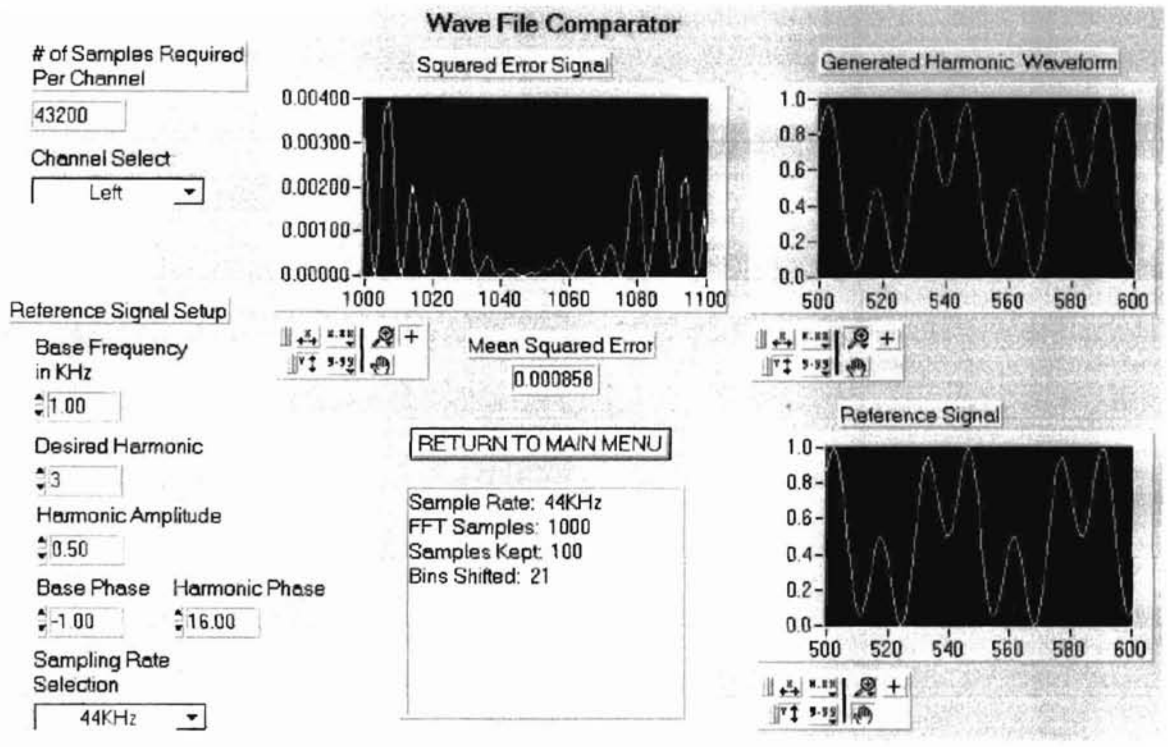
Harmonic Phase: -126.00

Sampling Rate Selection: 11KHz

Mean Squared Error: 0.000406

Sample Rate: 11KHz
FFT Samples: 1000
Samples Kept: 100
Bins Shifted: 21

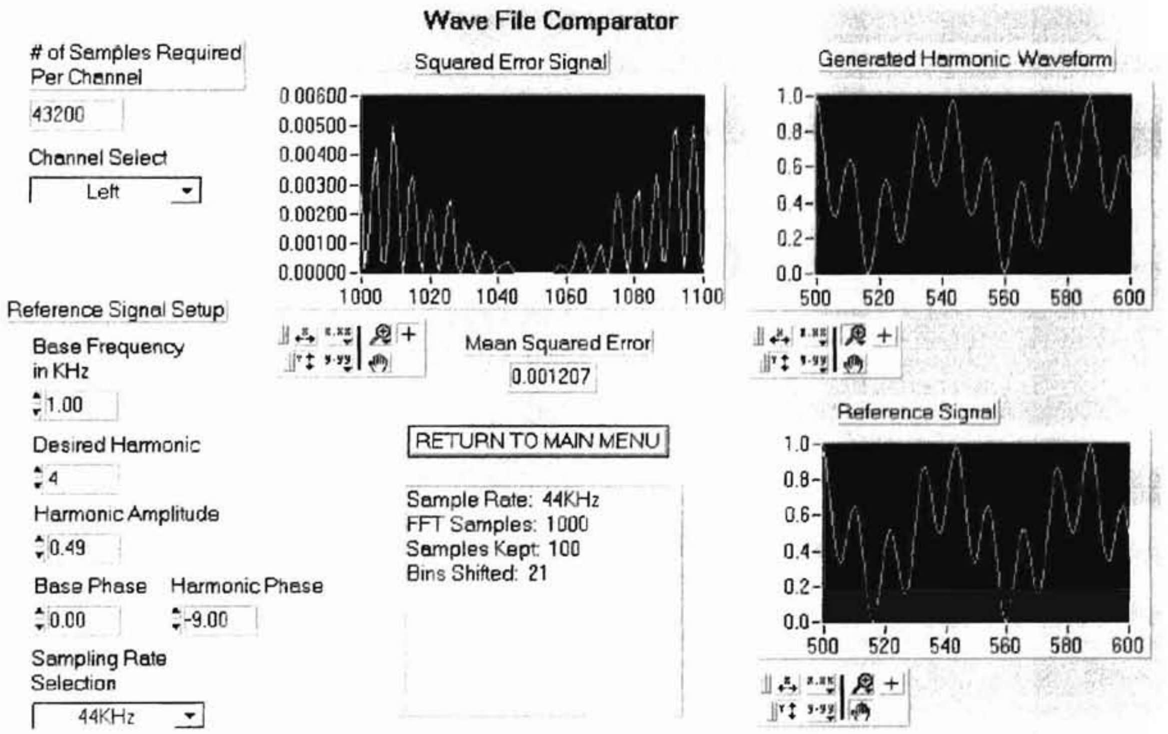
Figure B8: Test Run 32 Results



# of Samples Required Per Channel: 43200
Channel Select: Left
Reference Signal Setup
Base Frequency in KHz: 1.00
Desired Harmonic: 3
Harmonic Amplitude: 0.50
Base Phase: -1.00
Harmonic Phase: 16.00
Sampling Rate Selection: 44KHz
Mean Squared Error: 0.000858

```
Sample Rate: 44KHz
FFT Samples: 1000
Samples Kept: 100
Bins Shifted: 21
```

Figure B9: Test Run 37 Results



# of Samples Required Per Channel: 43200
Channel Select: Left
Reference Signal Setup
Base Frequency in KHz: 1.00
Desired Harmonic: 4
Harmonic Amplitude: 0.49
Base Phase: 0.00
Harmonic Phase: -9.00
Sampling Rate Selection: 44KHz
Mean Squared Error: 0.001207

```
Sample Rate: 44KHz
FFT Samples: 1000
Samples Kept: 100
Bins Shifted: 21
```

Figure B10: Test Run 38 Results

# APPENDIX C

## NUMERICAL TEST RESULTS DATA

The data of appendix c, verifies that the algorithm implemented meets the criteria of the harmonic generation algorithm by showing that the number of bins shifted and the phase rotation occurs as specified. Table 1 specifics are as follows:

# of Bins Shifted per peak = 5

Harmonic = 2

Low End Cut-off = 5

# of Peaks to Shift = 1

The results of Table 1 in appendix C show that the peak was determined to occur at bin 5 (the bins below, although larger, were ignored due to the low-end cut-off value set). Since the peak occurred at bin 5 and the desired harmonic was 2, the bins where shifted 5 bins to bin. The phase of each shifted bin was also rotated by 2.142 radians. Similar results can be found for Tables 2 through 4. Specifics about each table setup is found below.

Table 2 Data:

# of Bins Shifted per peak = 5

Harmonic = 3

Low End Cut-off = 5

# of Peaks to Shift = 1

Table 3 Data:

# of Bins Shifted per peak = 5

Harmonic = 4

Low End Cut-off = 5

# of Peaks to Shift = 1

Table 4 Data:

| | |
|---|---|
| # of Bins Shifted per peak | = 5 |
| Harmonic | = 2 |
| Low End Cut-off | = 5 |
| # of Peaks to Shift | = 2 |
| Peak Separation | = 2 |

Table 1:

| *Bin #* | *Original Magnitude* | *Original Phase (Rads)* | *Shifted Magnitude* | *Shifted Phase (Rads)* |
|---|---|---|---|---|
| 0 | 13626 | 0 | 0 | 0 |
| 1 | 1260.619 | -0.398 | 0 | 0 |
| 2 | 5630.856 | -0.676 | 0 | 0 |
| 3 | 2078.73 | 2.296 | 0 | 0 |
| 4 | 908.392 | 2.193 | 0 | 0 |
| 5 | 595.539 | 2.142 | 0 | 0 |
| 6 | 449.267 | 2.103 | 0 | 0 |
| 7 | 366.139 | 2.091 | 0 | 0 |
| 8 | 305.932 | 2.078 | 2078.73 | -1.845 |
| 9 | 270.836 | 2.091 | 908.392 | -1.948 |
| 10 | 235.49 | 2.088 | 595.539 | -1.999 |
| 11 | 213.798 | 2.13 | 449.267 | -2.039 |
| 12 | 193.516 | 2.13 | 366.139 | -2.05 |
| 13 | 181.456 | 2.144 | 0 | 0 |
| 14 | 166.429 | 2.158 | 0 | 0 |
| 15 | 152.237 | 2.189 | 0 | 0 |
| 16 | 148.185 | 2.208 | 0 | 0 |
| 17 | 137.283 | 2.228 | 0 | 0 |
| 18 | 128.231 | 2.242 | 0 | 0 |
| 19 | 124.854 | 2.281 | 0 | 0 |
| 20 | 113.89 | 2.301 | 0 | 0 |
| 21 | 113.633 | 2.316 | 0 | 0 |
| 22 | 106.668 | 2.341 | 0 | 0 |
| 23 | 104.017 | 2.341 | 0 | 0 |
| 24 | 101.642 | 2.412 | 0 | 0 |
| 25 | 99.247 | 2.428 | 0 | 0 |
| 26 | 93.092 | 2.467 | 0 | 0 |
| 27 | 91.751 | 2.483 | 0 | 0 |
| 28 | 91.951 | 2.469 | 0 | 0 |
| 29 | 82.923 | 2.503 | 0 | 0 |
| 30 | 84.661 | 2.534 | 0 | 0 |
| 31 | 81.965 | 2.614 | 0 | 0 |
| 32 | 81.853 | 2.636 | 0 | 0 |
| 33 | 79.134 | 2.649 | 0 | 0 |
| 34 | 76.401 | 2.674 | 0 | 0 |
| 35 | 76.477 | 2.699 | 0 | 0 |
| 36 | 74.661 | 2.749 | 0 | 0 |
| 37 | 75.537 | 2.818 | 0 | 0 |
| 38 | 77.467 | 2.771 | 0 | 0 |
| 39 | 72.549 | 2.82 | 0 | 0 |
| 40 | 70.612 | 2.822 | 0 | 0 |

| | | | | |
|---|---|---|---|---|
| 41 | 70.259 | 2.85 | 0 | 0 |
| 42 | 68.603 | 2.902 | 0 | 0 |
| 43 | 69.586 | 2.921 | 0 | 0 |
| 44 | 70.586 | 2.951 | 0 | 0 |
| 45 | 66.551 | 2.979 | 0 | 0 |
| 46 | 68.92 | 3.019 | 0 | 0 |
| 47 | 71.701 | 3.034 | 0 | 0 |
| 48 | 68.442 | 3.071 | 0 | 0 |
| 49 | 68.39 | -3.134 | 0 | 0 |
| 50 | 68 | -3.142 | 0 | 0 |
| 51 | 68.39 | 3.134 | 0 | 0 |
| 52 | 68.442 | -3.071 | 0 | 0 |
| 53 | 71.701 | -3.034 | 0 | 0 |
| 54 | 68.92 | -3.019 | 0 | 0 |
| 55 | 66.551 | -2.979 | 0 | 0 |
| 56 | 70.586 | -2.951 | 0 | 0 |
| 57 | 69.586 | -2.921 | 0 | 0 |
| 58 | 68.603 | -2.902 | 0 | 0 |
| 59 | 70.259 | -2.85 | 0 | 0 |
| 60 | 70.612 | -2.822 | 0 | 0 |
| 61 | 72.549 | -2.82 | 0 | 0 |
| 62 | 77.467 | -2.771 | 0 | 0 |
| 63 | 75.537 | -2.818 | 0 | 0 |
| 64 | 74.661 | -2.749 | 0 | 0 |
| 65 | 76.477 | -2.699 | 0 | 0 |
| 66 | 76.401 | -2.674 | 0 | 0 |
| 67 | 79.134 | -2.649 | 0 | 0 |
| 68 | 81.853 | -2.636 | 0 | 0 |
| 69 | 81.965 | -2.614 | 0 | 0 |
| 70 | 84.661 | -2.534 | 0 | 0 |
| 71 | 82.923 | -2.503 | 0 | 0 |
| 72 | 91.951 | -2.469 | 0 | 0 |
| 73 | 91.751 | -2.483 | 0 | 0 |
| 74 | 93.092 | -2.467 | 0 | 0 |
| 75 | 99.247 | -2.428 | 0 | 0 |
| 76 | 101.642 | -2.412 | 0 | 0 |
| 77 | 104.017 | -2.341 | 0 | 0 |
| 78 | 106.668 | -2.341 | 0 | 0 |
| 79 | 113.633 | -2.316 | 0 | 0 |
| 80 | 113.89 | -2.301 | 0 | 0 |
| 81 | 124.854 | -2.281 | 0 | 0 |
| 82 | 128.231 | -2.242 | 0 | 0 |
| 83 | 137.283 | -2.228 | 0 | 0 |
| 84 | 148.185 | -2.208 | 0 | 0 |
| 85 | 152.237 | -2.189 | 0 | 0 |
| 86 | 166.429 | -2.158 | 0 | 0 |

| | | | | |
|---|---|---|---|---|
| 87 | 181.456 | -2.144 | 0 | 0 |
| 88 | 193.516 | -2.13 | 366.139 | 2.05 |
| 89 | 213.798 | -2.13 | 449.267 | 2.039 |
| 90 | 235.49 | -2.088 | 595.539 | 1.999 |
| 91 | 270.836 | -2.091 | 908.392 | 1.948 |
| 92 | 305.932 | -2.078 | 2078.73 | 1.845 |
| 93 | 366.139 | -2.091 | 0 | 0 |
| 94 | 449.267 | -2.103 | 0 | 0 |
| 95 | 595.539 | -2.142 | 0 | 0 |
| 96 | 908.392 | -2.193 | 0 | 0 |
| 97 | 2078.73 | -2.296 | 0 | 0 |
| 98 | 5630.856 | 0.676 | 0 | 0 |
| 99 | 1260.619 | 0.398 | 0 | 0 |

Table 2

| Bin # | Original Magnitude | Original Phase (Rads) | Shifted Magnitude | Shifted Phase (Rads) |
|---|---|---|---|---|
| 0 | 13626 | 0 | 0 | 0 |
| 1 | 1260.619 | -0.398 | 0 | 0 |
| 2 | 5630.856 | -0.676 | 0 | 0 |
| 3 | 2078.73 | 2.296 | 0 | 0 |
| 4 | 908.392 | 2.193 | 0 | 0 |
| 5 | 595.539 | 2.142 | 0 | 0 |
| 6 | 449.267 | 2.103 | 0 | 0 |
| 7 | 366.139 | 2.091 | 0 | 0 |
| 8 | 305.932 | 2.078 | 0 | 0 |
| 9 | 270.836 | 2.091 | 0 | 0 |
| 10 | 235.49 | 2.088 | 0 | 0 |
| 11 | 213.798 | 2.13 | 0 | 0 |
| 12 | 193.516 | 2.13 | 0 | 0 |
| 13 | 181.456 | 2.144 | 2078.73 | 0.297 |
| 14 | 166.429 | 2.158 | 908.392 | 0.194 |
| 15 | 152.237 | 2.189 | 595.539 | 0.143 |
| 16 | 148.185 | 2.208 | 449.267 | 0.104 |
| 17 | 137.283 | 2.228 | 366.139 | 0.092 |
| 18 | 128.231 | 2.242 | 0 | 0 |
| 19 | 124.854 | 2.281 | 0 | 0 |
| 20 | 113.89 | 2.301 | 0 | 0 |
| 21 | 113.633 | 2.316 | 0 | 0 |
| 22 | 106.668 | 2.341 | 0 | 0 |
| 23 | 104.017 | 2.341 | 0 | 0 |
| 24 | 101.642 | 2.412 | 0 | 0 |
| 25 | 99.247 | 2.428 | 0 | 0 |
| 26 | 93.092 | 2.467 | 0 | 0 |
| 27 | 91.751 | 2.483 | 0 | 0 |
| 28 | 91.951 | 2.469 | 0 | 0 |
| 29 | 82.923 | 2.503 | 0 | 0 |
| 30 | 84.661 | 2.534 | 0 | 0 |
| 31 | 81.965 | 2.614 | 0 | 0 |
| 32 | 81.853 | 2.636 | 0 | 0 |
| 33 | 79.134 | 2.649 | 0 | 0 |
| 34 | 76.401 | 2.674 | 0 | 0 |
| 35 | 76.477 | 2.699 | 0 | 0 |
| 36 | 74.661 | 2.749 | 0 | 0 |
| 37 | 75.537 | 2.818 | 0 | 0 |
| 38 | 77.467 | 2.771 | 0 | 0 |
| 39 | 72.549 | 2.82 | 0 | 0 |
| 40 | 70.612 | 2.822 | 0 | 0 |

| | | | | |
|---|---|---|---|---|
| 41 | 70.259 | 2.85 | 0 | 0 |
| 42 | 68.603 | 2.902 | 0 | 0 |
| 43 | 69.586 | 2.921 | 0 | 0 |
| 44 | 70.586 | 2.951 | 0 | 0 |
| 45 | 66.551 | 2.979 | 0 | 0 |
| 46 | 68.92 | 3.019 | 0 | 0 |
| 47 | 71.701 | 3.034 | 0 | 0 |
| 48 | 68.442 | 3.071 | 0 | 0 |
| 49 | 68.39 | -3.134 | 0 | 0 |
| 50 | 68 | -3.142 | 0 | 0 |
| 51 | 68.39 | 3.134 | 0 | 0 |
| 52 | 68.442 | -3.071 | 0 | 0 |
| 53 | 71.701 | -3.034 | 0 | 0 |
| 54 | 68.92 | -3.019 | 0 | 0 |
| 55 | 66.551 | -2.979 | 0 | 0 |
| 56 | 70.586 | -2.951 | 0 | 0 |
| 57 | 69.586 | -2.921 | 0 | 0 |
| 58 | 68.603 | -2.902 | 0 | 0 |
| 59 | 70.259 | -2.85 | 0 | 0 |
| 60 | 70.612 | -2.822 | 0 | 0 |
| 61 | 72.549 | -2.82 | 0 | 0 |
| 62 | 77.467 | -2.771 | 0 | 0 |
| 63 | 75.537 | -2.818 | 0 | 0 |
| 64 | 74.661 | -2.749 | 0 | 0 |
| 65 | 76.477 | -2.699 | 0 | 0 |
| 66 | 76.401 | -2.674 | 0 | 0 |
| 67 | 79.134 | -2.649 | 0 | 0 |
| 68 | 81.853 | -2.636 | 0 | 0 |
| 69 | 81.965 | -2.614 | 0 | 0 |
| 70 | 84.661 | -2.534 | 0 | 0 |
| 71 | 82.923 | -2.503 | 0 | 0 |
| 72 | 91.951 | -2.469 | 0 | 0 |
| 73 | 91.751 | -2.483 | 0 | 0 |
| 74 | 93.092 | -2.467 | 0 | 0 |
| 75 | 99.247 | -2.428 | 0 | 0 |
| 76 | 101.642 | -2.412 | 0 | 0 |
| 77 | 104.017 | -2.341 | 0 | 0 |
| 78 | 106.668 | -2.341 | 0 | 0 |
| 79 | 113.633 | -2.316 | 0 | 0 |
| 80 | 113.89 | -2.301 | 0 | 0 |
| 81 | 124.854 | -2.281 | 0 | 0 |
| 82 | 128.231 | -2.242 | 0 | 0 |
| 83 | 137.283 | -2.228 | 366.139 | -0.092 |
| 84 | 148.185 | -2.208 | 449.267 | -0.104 |
| 85 | 152.237 | -2.189 | 595.539 | -0.143 |
| 86 | 166.429 | -2.158 | 908.392 | -0.194 |

|  |  |  |  |  |
|---|---|---|---|---|
| 87 | 181.456 | -2.144 | 2078.73 | -0.297 |
| 88 | 193.516 | -2.13 | 0 | 0 |
| 89 | 213.798 | -2.13 | 0 | 0 |
| 90 | 235.49 | -2.088 | 0 | 0 |
| 91 | 270.836 | -2.091 | 0 | 0 |
| 92 | 305.932 | -2.078 | 0 | 0 |
| 93 | 366.139 | -2.091 | 0 | 0 |
| 94 | 449.267 | -2.103 | 0 | 0 |
| 95 | 595.539 | -2.142 | 0 | 0 |
| 96 | 908.392 | -2.193 | 0 | 0 |
| 97 | 2078.73 | -2.296 | 0 | 0 |
| 98 | 5630.856 | 0.676 | 0 | 0 |
| 99 | 1260.619 | 0.398 | 0 | 0 |

Table 3

| *Bin #* | *Original Magnitude* | *Original Phase (Rads)* | *Shifted Magnitude* | *Shifted Phase (Rads)* |
|---|---|---|---|---|
| 0 | 13626 | 0 | 0 | 0 |
| 1 | 1260.619 | -0.398 | 0 | 0 |
| 2 | 5630.856 | -0.676 | 0 | 0 |
| 3 | 2078.73 | 2.296 | 0 | 0 |
| 4 | 908.392 | 2.193 | 0 | 0 |
| 5 | 595.539 | 2.142 | 0 | 0 |
| 6 | 449.267 | 2.103 | 0 | 0 |
| 7 | 366.139 | 2.091 | 0 | 0 |
| 8 | 305.932 | 2.078 | 0 | 0 |
| 9 | 270.836 | 2.091 | 0 | 0 |
| 10 | 235.49 | 2.088 | 0 | 0 |
| 11 | 213.798 | 2.13 | 0 | 0 |
| 12 | 193.516 | 2.13 | 0 | 0 |
| 13 | 181.456 | 2.144 | 0 | 0 |
| 14 | 166.429 | 2.158 | 0 | 0 |
| 15 | 152.237 | 2.189 | 0 | 0 |
| 16 | 148.185 | 2.208 | 0 | 0 |
| 17 | 137.283 | 2.228 | 0 | 0 |
| 18 | 128.231 | 2.242 | 2078.73 | 2.439 |
| 19 | 124.854 | 2.281 | 908.392 | 2.336 |
| 20 | 113.89 | 2.301 | 595.539 | 2.285 |
| 21 | 113.633 | 2.316 | 449.267 | 2.246 |
| 22 | 106.668 | 2.341 | 366.139 | 2.234 |
| 23 | 104.017 | 2.341 | 0 | 0 |
| 24 | 101.642 | 2.412 | 0 | 0 |
| 25 | 99.247 | 2.428 | 0 | 0 |
| 26 | 93.092 | 2.467 | 0 | 0 |
| 27 | 91.751 | 2.483 | 0 | 0 |
| 28 | 91.951 | 2.469 | 0 | 0 |
| 29 | 82.923 | 2.503 | 0 | 0 |
| 30 | 84.661 | 2.534 | 0 | 0 |
| 31 | 81.965 | 2.614 | 0 | 0 |
| 32 | 81.853 | 2.636 | 0 | 0 |
| 33 | 79.134 | 2.649 | 0 | 0 |
| 34 | 76.401 | 2.674 | 0 | 0 |
| 35 | 76.477 | 2.699 | 0 | 0 |
| 36 | 74.661 | 2.749 | 0 | 0 |
| 37 | 75.537 | 2.818 | 0 | 0 |
| 38 | 77.467 | 2.771 | 0 | 0 |
| 39 | 72.549 | 2.82 | 0 | 0 |
| 40 | 70.612 | 2.822 | 0 | 0 |

| | | | | |
|---|---|---|---|---|
| 41 | 70.259 | 2.85 | 0 | 0 |
| 42 | 68.603 | 2.902 | 0 | 0 |
| 43 | 69.586 | 2.921 | 0 | 0 |
| 44 | 70.586 | 2.951 | 0 | 0 |
| 45 | 66.551 | 2.979 | 0 | 0 |
| 46 | 68.92 | 3.019 | 0 | 0 |
| 47 | 71.701 | 3.034 | 0 | 0 |
| 48 | 68.442 | 3.071 | 0 | 0 |
| 49 | 68.39 | -3.134 | 0 | 0 |
| 50 | 68 | -3.142 | 0 | 0 |
| 51 | 68.39 | 3.134 | 0 | 0 |
| 52 | 68.442 | -3.071 | 0 | 0 |
| 53 | 71.701 | -3.034 | 0 | 0 |
| 54 | 68.92 | -3.019 | 0 | 0 |
| 55 | 66.551 | -2.979 | 0 | 0 |
| 56 | 70.586 | -2.951 | 0 | 0 |
| 57 | 69.586 | -2.921 | 0 | 0 |
| 58 | 68.603 | -2.902 | 0 | 0 |
| 59 | 70.259 | -2.85 | 0 | 0 |
| 60 | 70.612 | -2.822 | 0 | 0 |
| 61 | 72.549 | -2.82 | 0 | 0 |
| 62 | 77.467 | -2.771 | 0 | 0 |
| 63 | 75.537 | -2.818 | 0 | 0 |
| 64 | 74.661 | -2.749 | 0 | 0 |
| 65 | 76.477 | -2.699 | 0 | 0 |
| 66 | 76.401 | -2.674 | 0 | 0 |
| 67 | 79.134 | -2.649 | 0 | 0 |
| 68 | 81.853 | -2.636 | 0 | 0 |
| 69 | 81.965 | -2.614 | 0 | 0 |
| 70 | 84.661 | -2.534 | 0 | 0 |
| 71 | 82.923 | -2.503 | 0 | 0 |
| 72 | 91.951 | -2.469 | 0 | 0 |
| 73 | 91.751 | -2.483 | 0 | 0 |
| 74 | 93.092 | -2.467 | 0 | 0 |
| 75 | 99.247 | -2.428 | 0 | 0 |
| 76 | 101.642 | -2.412 | 0 | 0 |
| 77 | 104.017 | -2.341 | 0 | 0 |
| 78 | 106.668 | -2.341 | 366.139 | -2.234 |
| 79 | 113.633 | -2.316 | 449.267 | -2.246 |
| 80 | 113.89 | -2.301 | 595.539 | -2.285 |
| 81 | 124.854 | -2.281 | 908.392 | -2.336 |
| 82 | 128.231 | -2.242 | 2078.73 | -2.439 |
| 83 | 137.283 | -2.228 | 0 | 0 |
| 84 | 148.185 | -2.208 | 0 | 0 |
| 85 | 152.237 | -2.189 | 0 | 0 |
| 86 | 166.429 | -2.158 | 0 | 0 |

| | | | | |
|---|---|---|---|---|
| 87 | 181.456 | -2.144 | 0 | 0 |
| 88 | 193.516 | -2.13 | 0 | 0 |
| 89 | 213.798 | -2.13 | 0 | 0 |
| 90 | 235.49 | -2.088 | 0 | 0 |
| 91 | 270.836 | -2.091 | 0 | 0 |
| 92 | 305.932 | -2.078 | 0 | 0 |
| 93 | 366.139 | -2.091 | 0 | 0 |
| 94 | 449.267 | -2.103 | 0 | 0 |
| 95 | 595.539 | -2.142 | 0 | 0 |
| 96 | 908.392 | -2.193 | 0 | 0 |
| 97 | 2078.73 | -2.296 | 0 | 0 |
| 98 | 5630.856 | 0.676 | 0 | 0 |
| 99 | 1260.619 | 0.398 | 0 | 0 |

Table 4

| *Bin #* | *Original Magnitude* | *Original Phase (Rads)* | *Shifted Magnitude* | *Shifted Phase (Rads)* |
|---|---|---|---|---|
| 0 | 12360 | 0 | 0 | 0 |
| 1 | 1020.241 | -0.228 | 0 | 0 |
| 2 | 3353.315 | -0.542 | 0 | 0 |
| 3 | 570.451 | 1.136 | 0 | 0 |
| 4 | 2101.392 | -0.417 | 0 | 0 |
| 5 | 2134.573 | 2.396 | 0 | 0 |
| 6 | 783.431 | 2.122 | 0 | 0 |
| 7 | 291.142 | 2.194 | 0 | 0 |
| 8 | 242.901 | 2.281 | 570.451 | -2.751 |
| 9 | 202.596 | 2.289 | 2101.392 | 1.979 |
| 10 | 176.028 | 2.301 | 2134.573 | -1.491 |
| 11 | 151.751 | 2.325 | 783.431 | -1.765 |
| 12 | 131.545 | 2.333 | 291.142 | -1.693 |
| 13 | 124.636 | 2.323 | 0 | 0 |
| 14 | 115.743 | 2.32 | 0 | 0 |
| 15 | 105.368 | 2.352 | 0 | 0 |
| 16 | 102.362 | 2.321 | 0 | 0 |
| 17 | 93.445 | 2.354 | 0 | 0 |
| 18 | 84.667 | 2.351 | 242.901 | -1.702 |
| 19 | 80.726 | 2.371 | 202.596 | -1.693 |
| 20 | 76.436 | 2.407 | 176.028 | -1.681 |
| 21 | 72.967 | 2.444 | 151.751 | -1.658 |
| 22 | 69.335 | 2.453 | 131.545 | -1.649 |
| 23 | 66.091 | 2.454 | 0 | 0 |
| 24 | 65.203 | 2.4 | 0 | 0 |
| 25 | 66.573 | 2.57 | 0 | 0 |
| 26 | 59.457 | 2.553 | 0 | 0 |
| 27 | 60.363 | 2.565 | 0 | 0 |
| 28 | 57.802 | 2.521 | 0 | 0 |
| 29 | 55.742 | 2.578 | 0 | 0 |
| 30 | 52.556 | 2.549 | 0 | 0 |
| 31 | 50.701 | 2.681 | 0 | 0 |
| 32 | 54.711 | 2.646 | 0 | 0 |
| 33 | 50.779 | 2.704 | 0 | 0 |
| 34 | 48.258 | 2.773 | 0 | 0 |
| 35 | 50.238 | 2.77 | 0 | 0 |
| 36 | 44.988 | 2.814 | 0 | 0 |
| 37 | 47.232 | 2.879 | 0 | 0 |
| 38 | 47.976 | 2.838 | 0 | 0 |
| 39 | 45.892 | 2.868 | 0 | 0 |
| 40 | 44.581 | 2.9 | 0 | 0 |

| | | | | |
|---|---|---|---|---|
| 41 | 43.906 | 2.866 | 0 | 0 |
| 42 | 45.677 | 2.866 | 0 | 0 |
| 43 | 42.934 | 3.001 | 0 | 0 |
| 44 | 41.212 | 2.975 | 0 | 0 |
| 45 | 39.847 | 3.039 | 0 | 0 |
| 46 | 39.91 | 2.925 | 0 | 0 |
| 47 | 45.347 | 2.965 | 0 | 0 |
| 48 | 43.376 | 3.055 | 0 | 0 |
| 49 | 42.647 | -3.091 | 0 | 0 |
| 50 | 48 | -3.142 | 0 | 0 |
| 51 | 42.647 | 3.091 | 0 | 0 |
| 52 | 43.376 | -3.055 | 0 | 0 |
| 53 | 45.347 | -2.965 | 0 | 0 |
| 54 | 39.91 | -2.925 | 0 | 0 |
| 55 | 39.847 | -3.039 | 0 | 0 |
| 56 | 41.212 | -2.975 | 0 | 0 |
| 57 | 42.934 | -3.001 | 0 | 0 |
| 58 | 45.677 | -2.866 | 0 | 0 |
| 59 | 43.906 | -2.866 | 0 | 0 |
| 60 | 44.581 | -2.9 | 0 | 0 |
| 61 | 45.892 | -2.868 | 0 | 0 |
| 62 | 47.976 | -2.838 | 0 | 0 |
| 63 | 47.232 | -2.879 | 0 | 0 |
| 64 | 44.988 | -2.814 | 0 | 0 |
| 65 | 50.238 | -2.77 | 0 | 0 |
| 66 | 48.258 | -2.773 | 0 | 0 |
| 67 | 50.779 | -2.704 | 0 | 0 |
| 68 | 54.711 | -2.646 | 0 | 0 |
| 69 | 50.701 | -2.681 | 0 | 0 |
| 70 | 52.556 | -2.549 | 0 | 0 |
| 71 | 55.742 | -2.578 | 0 | 0 |
| 72 | 57.802 | -2.521 | 0 | 0 |
| 73 | 60.363 | -2.565 | 0 | 0 |
| 74 | 59.457 | -2.553 | 0 | 0 |
| 75 | 66.573 | -2.57 | 0 | 0 |
| 76 | 65.203 | -2.4 | 0 | 0 |
| 77 | 66.091 | -2.454 | 0 | 0 |
| 78 | 69.335 | -2.453 | 131.545 | 1.649 |
| 79 | 72.967 | -2.444 | 151.751 | 1.658 |
| 80 | 76.436 | -2.407 | 176.028 | 1.681 |
| 81 | 80.726 | -2.371 | 202.596 | 1.693 |
| 82 | 84.667 | -2.351 | 242.901 | 1.702 |
| 83 | 93.445 | -2.354 | 0 | 0 |
| 84 | 102.362 | -2.321 | 0 | 0 |
| 85 | 105.368 | -2.352 | 0 | 0 |
| 86 | 115.743 | -2.32 | 0 | 0 |

| | | | | |
|---|---|---|---|---|
| 87 | 124.636 | -2.323 | 0 | 0 |
| 88 | 131.545 | -2.333 | 291.142 | 1.693 |
| 89 | 151.751 | -2.325 | 783.431 | 1.765 |
| 90 | 176.028 | -2.301 | 2134.573 | 1.491 |
| 91 | 202.596 | -2.289 | 2101.392 | -1.979 |
| 92 | 242.901 | -2.281 | 570.451 | 2.751 |
| 93 | 291.142 | -2.194 | 0 | 0 |
| 94 | 783.431 | -2.122 | 0 | 0 |
| 95 | 2134.573 | -2.396 | 0 | 0 |
| 96 | 2101.392 | 0.417 | 0 | 0 |
| 97 | 570.451 | -1.136 | 0 | 0 |
| 98 | 3353.315 | 0.542 | 0 | 0 |
| 99 | 1020.241 | 0.228 | 0 | 0 |

VITA

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Master of Science

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