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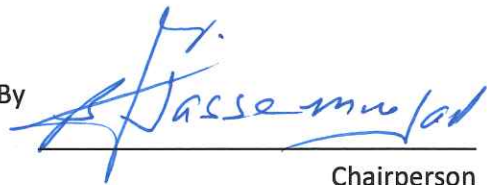
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EMBEDDED TEST AND MEASUREMENT

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Acknowledgements

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Abstract

A virtual test set was designed to perform audio range analog transmission impairment testing in accordance with IEEE Std 743-1995. This virtual test set was developed in the Laboratory Virtual Instrument Engineering Workbench (LabVIEW) development suite, a programing language primarily focused on automation and control, and was designed with flexibility in mind. Through the use of queues and parallel processing the virtual test set is capable of functioning as a transmission impairment measurement test set, a function generator, an oscilloscope, a spectrum analyzer, and a digital multimeter. The impairment measurements the virtual test set can perform include: 1004Hz Net Loss, C-Notched Noise Ratio, Impulse Noise, Phase Jitter, 3-Tone Slope, Attenuation Distortion, Intermodulation Distortion, Envelope Delay Distortion, and Echo Loss. The virtual test set is capable of performing all of the impairment measurement tests automatically and can even send and receive selective signaling and frequency shift keying signals. The virtual test set demonstrates that software based test equipment, with minimal hardware for interfacing purposes, can produce test sets that are smaller, lighter, more user friendly, and more cost effective analog test sets than hardware based test sets.

Acronyms and Abbreviations

0TLP	Zero Transmission Level Point
2nd IMD	Second Harmonic Intermodulation Distortion
3rd IMD	Third Harmonic Intermodulation Distortion
A	Amps
AC	Alternating Current
AGND	Analog Ground Reference
AI	Analog Input
AM	Amplitude Modulation
AO	Analog Output
Arms	Root Mean Square Amps
DAQ	Data Acquisition Device
dB	Decibel
dBm	Decibel referenced to one milliwatt
dBrn	Decibel referenced to one picowatt
dBrnc0	dBrn after a C-Message Filter at a zero transmission level point
DC	Direct Current
DGND	Digital Ground Reference
DIO	Digital Input and Output
DMM	Digital Multi-Meter
EDD	Envelope Delay Distortion
FM	Frequency Modulation
Hz	Hertz
I	Current
IEEE	Institute of Electrical and Electronics Engineers
IMD	Intermodulation Distortion
kHz	kilohertz
kS/s	kilo-Samples per second
LabVIEW	Laboratory Virtual Instrument Engineering Workbench
mA	Milliamp
MDT	Mobile Data Terminal
MOhms	Mega-Ohms
mV	Millivolt
NA	Not Applicable
NI	National Instruments
Op-amp	Operational Amplifier
OS	Operating System
P	Power
PM	Phase Modulation
R	Resistance

RAM	Random Access Memory
RMS	Root Mean Square
Rx	Receive
SPDT	Single Pull Double Throw
S/s	Samples per second
TIMS	Transmission Impairment Measurement Set
TLP	Transmission Level Point
Tx	Transmit
TxRx	Transmit and Receive
USB	Universal Serial Bus
V	Volts
VI	Virtual Instrument
Vrms	Root Mean Square Volts
VMDT	Virtual Mobile Data Terminal
Z	Impedance

1   Introduction

The telecommunications industry uses portable test sets in the field during preventative and corrective maintenance of analog voice switch systems to determine if a communication channel meets the voice and data quality standards detailed in the technical instruction handbook for the system. These test sets use physical components to perform the filtering, measurements, and provide the user interface necessary for the tests. The dependence on hardware means test sets are heavy, bulky, expensive, and not user friendly. Furthermore, these hardware based test sets cannot be updated to comply with new standards. New technology has provided solutions for performing preventative and corrective maintenance tests for telecommunication systems using virtual instruments. The filtering, measurements, and user interface can now be accomplished with software that can be embedded into a computer test set, only requiring hardware to interface with a communication system.

A virtual test set was developed as a proof of concept for software based test sets that can be embedded into any computer. Through the use of queues and parallel processing the virtual test set is capable of functioning as a transmission impairment measurement test set, a function generator, an oscilloscope, a spectrum analyzer, and a digital multi-meter. The impairment measurements the virtual test set can perform include: 1004Hz Net Loss, C-Notched Noise Ratio, Impulse Noise, Phase Jitter, 3-Tone Slope, Attenuation Distortion, Intermodulation Distortion, Envelope Delay Distortion, and Echo Loss. The virtual test set's application was placed on laptops used by field technicians who routinely perform maintenance on audio range analog transmission lines. This virtual test set demonstrates that only a minimum amount of hardware is needed to collect data from a transmission line and software can be used to calculate the impairments detected on the line in accordance with IEEE standard 743-1995 **[1]**. Utilizing software allows for a flexible, customizable, and user friendly interface while fewer hardware components allows for a smaller, lighter, and more cost effective test set. In addition, since the measurements are performed using software, functions can be added, removed, or updated.

2   Objective

The objective of this project was to develop a virtual test set capable of performing audio range analog transmission impairment tests in accordance with IEEE Standard 743-1995 **[1]**. This virtual test set serves as a demonstration of the ability to develop software based test sets that can be embedded into existing computer platforms. Specifically, the virtual test set is being developed to perform the functions of a Transmission Impairment Measurement Set (TIMS). In addition to performing the functions of a hardware TIMS the virtual test set has been developed to perform the functions of other hardware based test equipment including a DMM, Oscilloscope, Function Generator, and Spectrum Analyzer to create one all in one test set that is smaller, lighter, more cost effective, and more user friendly than the hardware based equipment.

3   Hardware

The virtual test set is an executable file that is embedded in a computer platform to allow users to generate signals and perform audio range analog tests. However, the signals generated and processed inside the computer are virtual signals, an array of data points used to represent a signal, while the signals in telecommunications are analog signals. To convert an analog signal to a virtual signal, or vice versa, it is necessary to use a Data Acquisition Device (DAQ) which serves as a digital-to-analog and analog-to-digital converter. In addition, a connection box is used to provide connections to measurement circuits, perform impedance matching, as well as function as a hybrid circuit.

A DAQ serves the function of converting the virtual signals used by the virtual test set application to and from the analog signals used in telecommunications. In this project the NI-MyDAQ, **Figure 3-1**, was used because it has a sufficient sample rate for audio range analog input and output signals, a built in Digital MultiMeter (DMM), and built in +/- 15 volt power pins used for external circuits **[2]**. The MyDAQ was originally developed for academic purposes, designed to be used in class rooms. Since it was designed to be used in various lessons it includes many functions not included with other NI DAQs while possessing sufficient power to meet all of the virtual test set's requirements. The MyDAQ communicates with the computer through a standard USB port and obtains all of its power directly from the USB port. If the virtual test set application is installed on a laptop, the virtual test set can be transported and used anywhere without the need for an external power supply. More information on the MyDAQ is provided in **Appendix A**.

Figure 3-1: NI MyDAQ

In addition to converting between virtual signals and analog signals, the virtual test set must be connected to the circuit undergoing testing. The line interface contains a circuit board along with the MyDAQ and is used as a bridge between the computer and the system being tested, **Figure 3-2**. The line interface provides screw terminal, phone jack, RJ-45 jack, and bantam jack connections in both 2 and 4 wire configurations. In addition to the physical connections, the line interface provides: signal amplification, impedance matching, and a hybrid circuit. The process of setting up the Line Interface will be discussed in detail in **Section 4.2.1**.
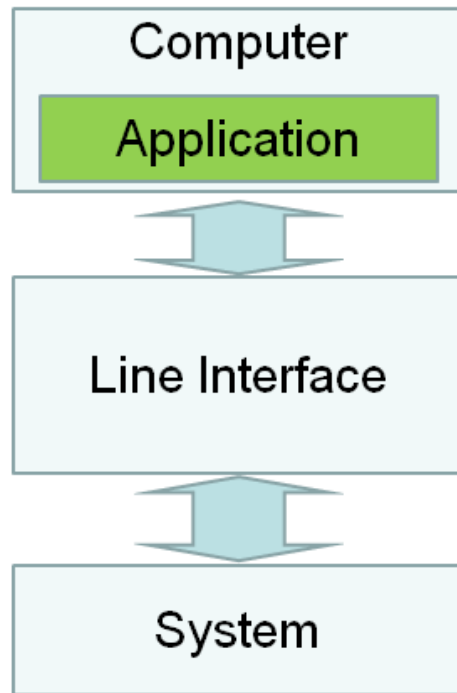
Figure 3-2: Connection Block Diagram

The MyDAQ and circuit board inside the Line Interface, **Figure 3-3**, are both powered through the USB cable when it is connected to a computer, eliminating the need for an additional power source. As part of the setup process the user must: connect the Line Interface to the computer using the USB cable, select two or four wire mode using the toggle switch, select the desired impedances for the transmit and receive circuits using the three rotary switches, and connect to the system under test using one of the available connection types. The Line Interface is capable of connecting to circuits using bantam jacks, standard phone cords, Ethernet cables, or direct wire connections, **Figure 3-4**. In addition, the line interface provides direct access to the DMM and Power pins through Banana Jack Plugs. More hardware details are provided in **Appendix A**.
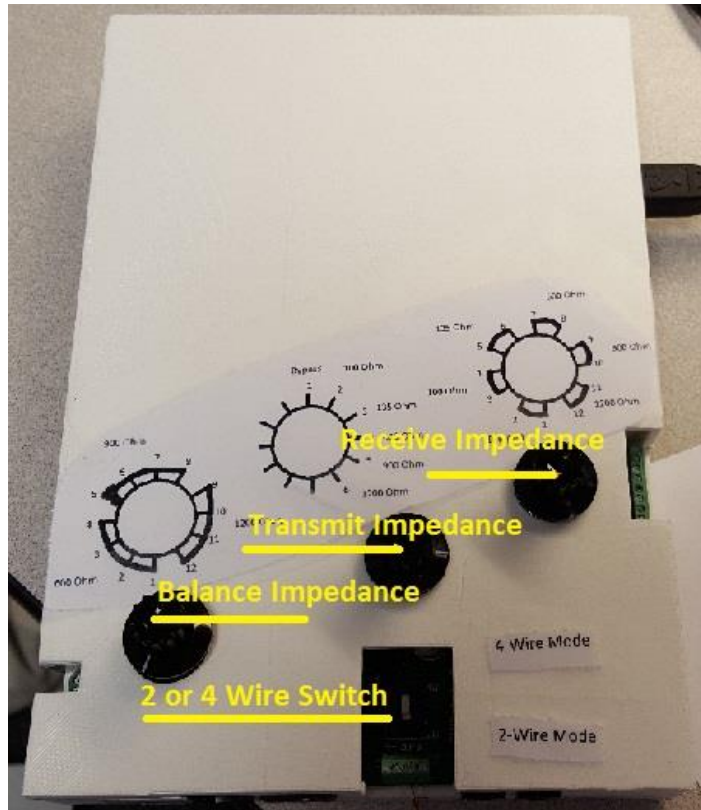
Figure 3-3: VIRTUAL TEST SET Line Interface
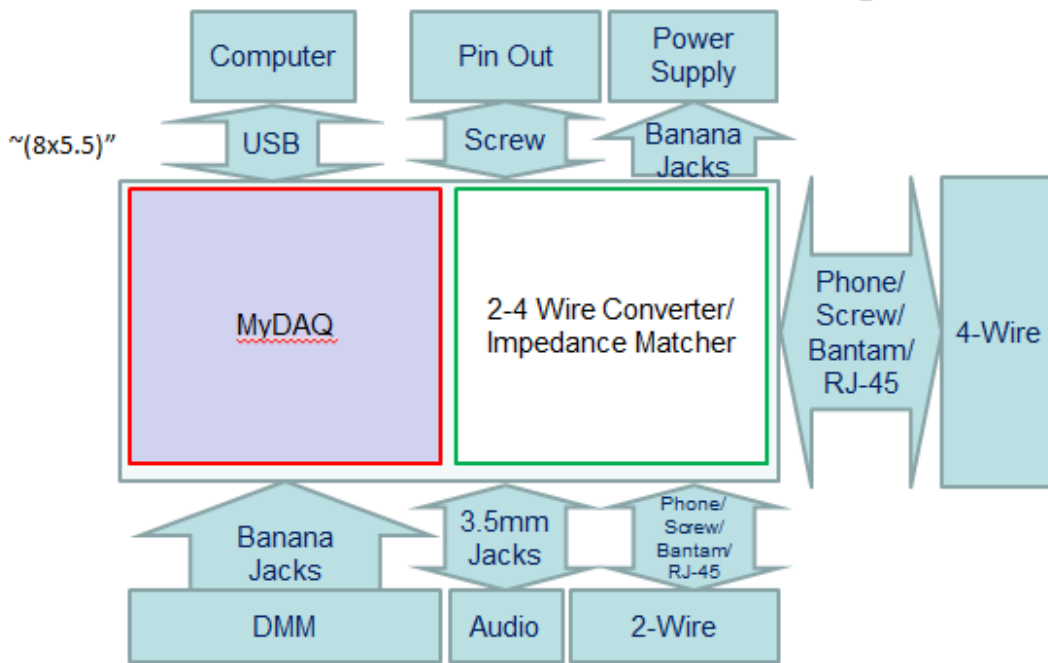
Figure 3-4: Line Interface Block Diagram.

4 Software

The purpose of the virtual test set is to perform the functions of hardware based signal generation and measurement devices with a single software based device. The virtual test set only uses hardware to transmit and receive signals, the waveforms are generated and processed using software. The software takes the form of an executable application that can be installed onto a computer. In addition to the virtual test set's application, drivers for the DAQ are required to communicate with the hardware and the LabVIEW Runtime Engine is required to run the application. Additional software requirements are listed in **Table 4-1** while the system requirements are listed in **Table 4-2**.

Table 4-1: Virtual Test Set Software Requirements

Software	Function
Application and Folders	The Virtual Test Set
LabVIEW Runtime Engine	Runs the VIRTUAL TEST SET Application
DAQmx Drivers	Operates the MyDAQ
Microsoft Word	Used for Reports
Adobe Reader	Used to View PDFs

Table 4-2: Virtual Test Set System Requirements

Specification	Requirement
Processor	Pentium 4 G1 or better.
RAM	256 MB
OS	Windows 7 or later.
Disk Space	2.4 GB

4.1 Approach

Many different programs were used during the development of the virtual test set. Multisim was utilized to simulate the hardware circuit, Altium was utilized to design the circuit board, and AutoCAD was used to design the case for Line Interface. However, the virtual test set's application was developed in LabVIEW. The application was developed as a queued state machine, a type of parallel processing that uses queues to control the flow of a program.

4.1.1 LabVIEW

LabVIEW was chosen for the virtual test set because it is an industry standard graphical programming language developed for automation and control. LabVIEW's icon based graphical programing makes programs easy to learn and use while also providing powerful tools for automation and control. Each LabVIEW program or application, referred to as Virtual Instrument (VI), contains two basic components: the front panel and the block diagram.

4.1.1.1 Front Panel

In LabVIEW the front panel, **Figure 4-1**, is a user interface which is opened automatically whenever the users open a VI file. Controls send information to the block diagram to be used in programs while indicators display the information gathered from the block diagram. Controls can range

from numeric and text, entries which can contain a wide range of entries, to buttons and slide bars that are limited to a select range of values. Indicators can be allowed to display any range of numeric or text values or be limited to specific ranges. It is possible to add, remove, move, and change front panel objects programmatically allowing the appearance of any application to be user friendly and fully customizable.
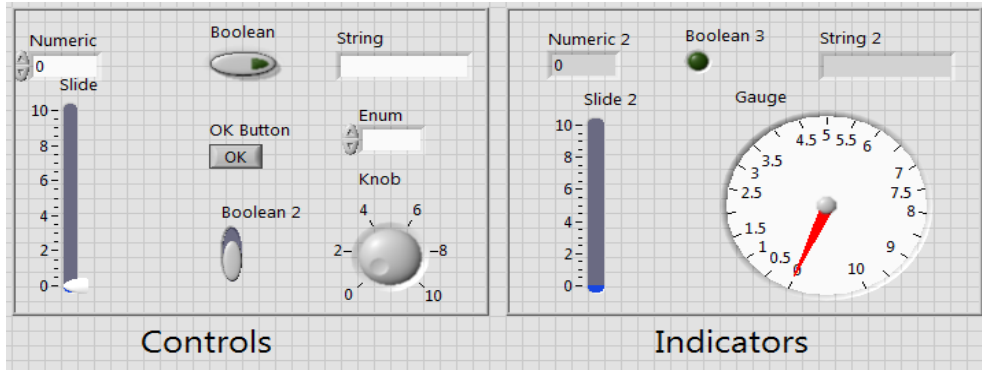
Figure 4-1: LabVIEW Front Panel

4.1.1.2   Block Diagram

The block diagram, **Figure 4-2**, represents the program's code as visual icons including blocks, loops, and colored wires. The block diagram also contains the sub-VIs, sub-routines with their own front panel and block diagram. Sub-VIs can also contain their own sub-VIs to make an extensive hierarchy. All the tools in LabVIEW are simple to learn but through the use of loops, structures, and sub-VIs there is almost no limit to what LabVIEW can accomplish. In-depth details about the virtual test's block diagrams set are given in **Appendix B**.
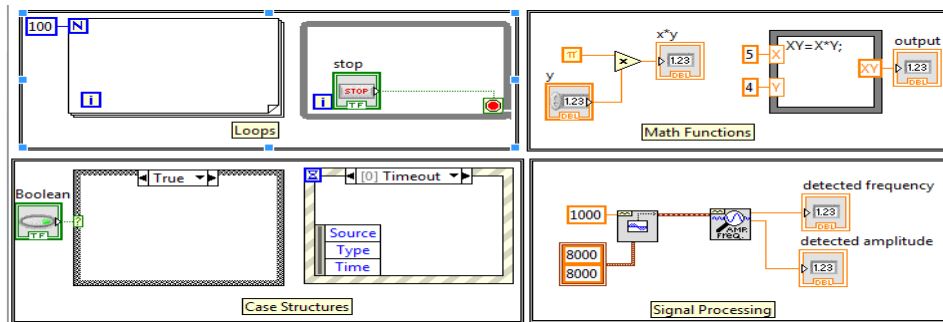
Figure 4-2: LabVIEW Block Diagram Example.

4.1.2   Parallel Processing

Parallel processing is the process of performing multiple functions simultaneously. The major benefit of using parallel processing is that different functions can work at different speeds. One loop can be handling a complicated process while another waits for a new command and the two loops can run independently without waiting for the other to complete any calculation. **Figure 4-3** displays two while loops running in parallel.

Figure 4-3: Parallel Processing

4.1.3   Queued State Machine

A Queued State Machine is a type of parallel processing. Like all parallel processes, a queued state machine uses multiple loops to perform functions simultaneously. However, a queued state machine breaks its parallel processes into specific categories. These categories include the command loop, the queue loop, and the execution loops as shown in **Figure 4-4**. The command loop accepts commands from the user and passes the information to the queue and execution loops as needed. The queue loop ensures commands are executed in the proper order. Finally, the execution loop performs the functions associated with the commands.
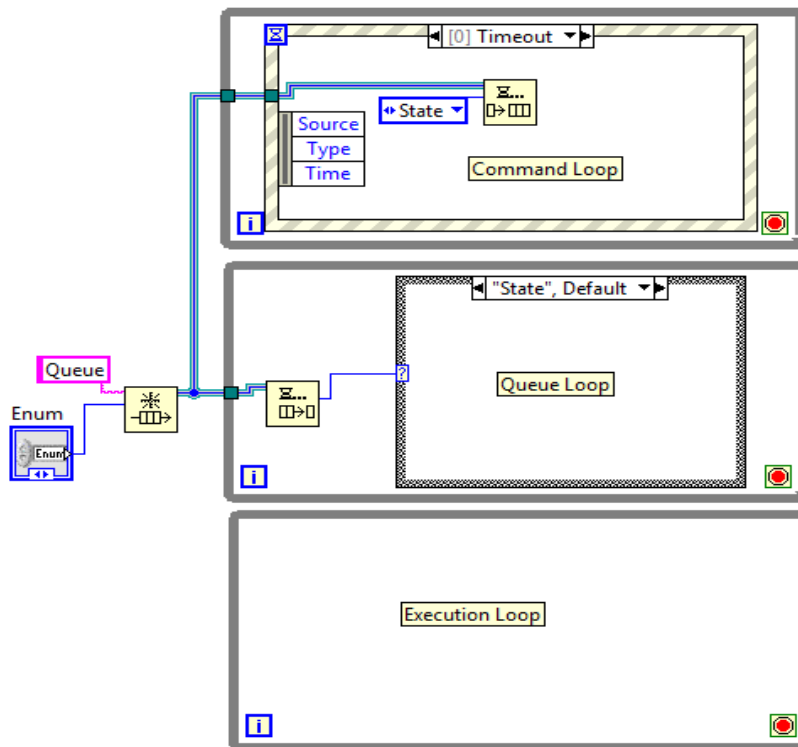
Figure 4-4: Queued State Machine

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4.1.4 Queue

A queue allows for multiple commands to be executed in a specific order. Typically all commands are executed in the order they are entered but some commands can be executed before others and the queue can be emptied causing all commands to be canceled. **Figure 4-5** illustrates the steps required to use a queue:

1) **Create Queue:** The 'Obtain Queue' function allows a queue to be created with a given name and value type. Obtain Queue outputs a queue reference that can be passed to different loops and even different VIs. Any queue function using the same queue reference will act on the same queue.

2) **Add Queue Elements:** Functions such as 'Enqueue Element' will add elements to the queue. These elements can be any data or command as long as it is of the same type used for 'Obtain Queue.' These elements can be entered at the back of the queue or at the front of the queue but will always be executed going from the front of the queue to the back.

3) **Receive Elements:** The 'Dequeue Element' function will remove the element at the front of the queue so the values contained within can be used to perform specific actions.

4) **Destroy Queue:** The 'Release Queue' function will remove any remaining elements from the queue and destroy the queue reference. This is typically only done when the application needs to close.

Queues allow for carefully controlled flow of data but function much slower than other forms of data transportation. Queues are best used when controlling the flow of data is very important and the speed of the data flow is not as important.
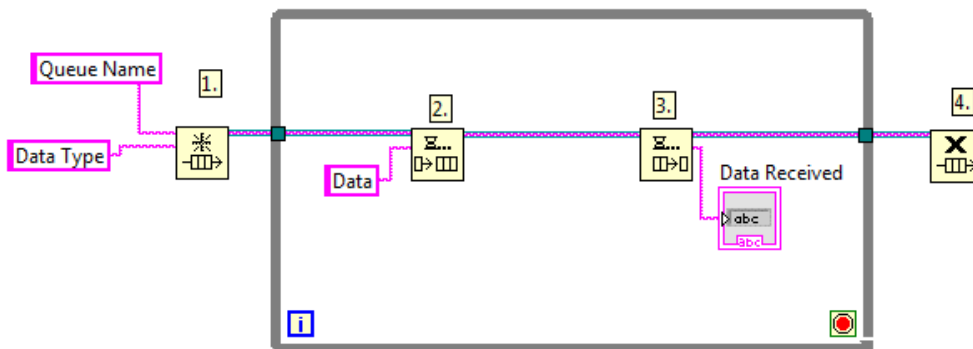
Figure 4-5: Queue Flow Control

4.2 Initial Setup and Flow Control

The virtual test set has many functions and settings, all of which must be activated and changed without interfering with each other. Special VIs are used to setup and control these functions and settings.

4.2.1 Open First

This VI is the first VI that is opened when the virtual test set is activated. The virtual test set requires the connector box to be connected and all hardware settings must be configured correctly to obtain accurate results. This VI gives detailed step by step instructions for setting up the hardware.
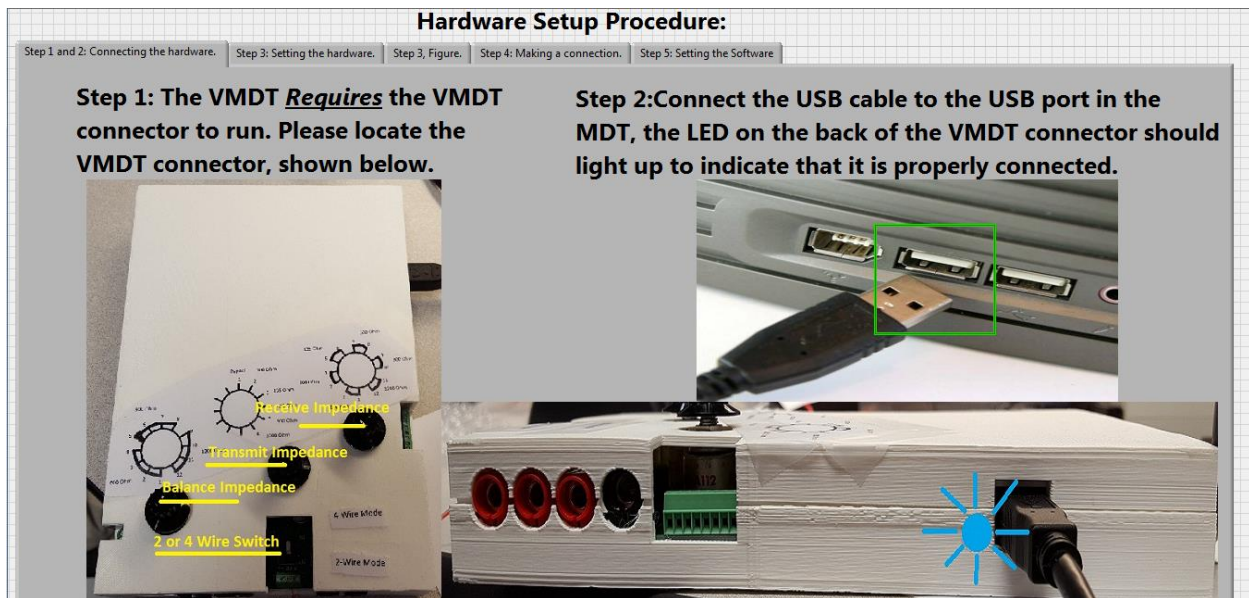
Figure 4-6: Open First Front Panel

The front panel of the Open First VI is comprised of a series of tabs, each dedicated to providing instructions for setting up the hardware and software before using the virtual test set, **Figure 4-6**. These tabs provide the following steps of instructions:

1. Locating the Line Interface.
2. Connecting the Line Interface to the computer using the USB cable.
3. Adjusting the hardware settings to match the circuit being tested.
4. Connecting to the test circuit.
5. Configuring the software.

Locating the Line Interface: The virtual test set requires the Line Interface to run. This step includes a picture of the Line Interface, **Figure 4-7**, to help the user locate the Line Interface.
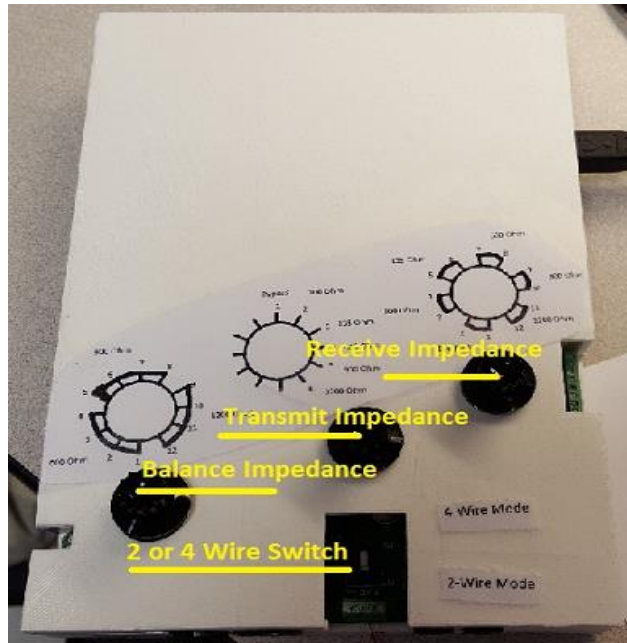
Figure 4-7: The Line Interface

      Connecting the Line Interface to the computer using the USB cable: This step tells the user how to connect the Line Interface to a laptop, **Figure 4-8**, and informs them that the LED on the side of the Line Interface will light up once it is properly connected, **Figure 4-9**.

Figure 4-8: Connecting the Line Interface's USB cable.

Figure 4-9: The Line Interface's power indicator LED.

Adjusting the hardware settings to match the circuit that is being tested: The Line Interface's hardware settings are changed using three rotary switches and a toggle switch:

Rotary switch 1 changes the receive circuit impedance: 100 Ohm, 135 Ohm, 600 Ohm, 900 Ohm, 1200 Ohm, or Bridged.

Rotary switch 2 changes the transmit circuit impedance: 100 Ohm, 135 Ohm, 600 Ohm, 900 Ohm, 1200 Ohm, or Bypass.

Rotary switch 3 changes the hybrid circuit balancing impedance: 600 Ohm, 900 Ohm, or 1200 Ohm.

The toggle switch toggles between 2-Wire mode and 4-Wire mode. In 2-Wire mode each of the rotary switches should be set to the same impedance value.

An Illustration is provided to indicate the locations and positions of the switches on the Line Interface, **Figure 4-10**.
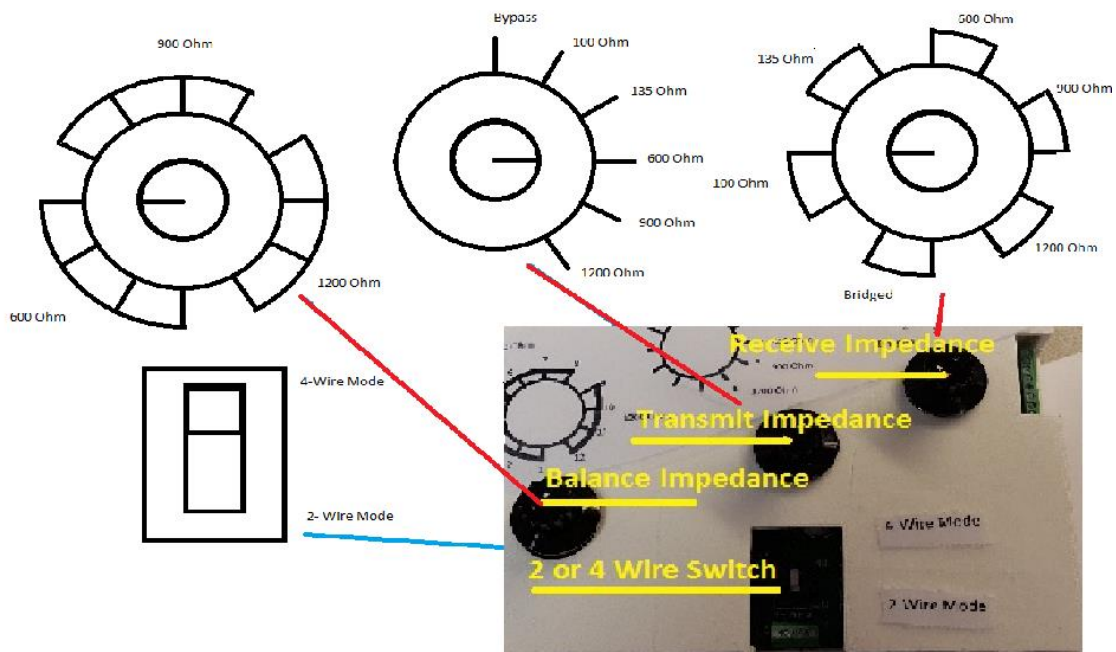
Figure 4-10: Line Interface Switch Locations and Positions.

Connecting to the test circuit: The Line Interface allows for connections to be made using straight wires, phone jacks, RJ-45 jacks, and bantam jacks. However, only one type of connection can be made at a time for each of the circuits: transmit, receive, and hybrid. This means that an RJ-45 jack can be used for the transmit circuit and a Bantam jack can be used for the receive circuit but neither circuit can use a RJ-45 and a Bantam Jack at the same time. Illustrations are provided to demonstrate what each connection looks like and notes are provided on how to properly connect to them, **Figure 4-11**.

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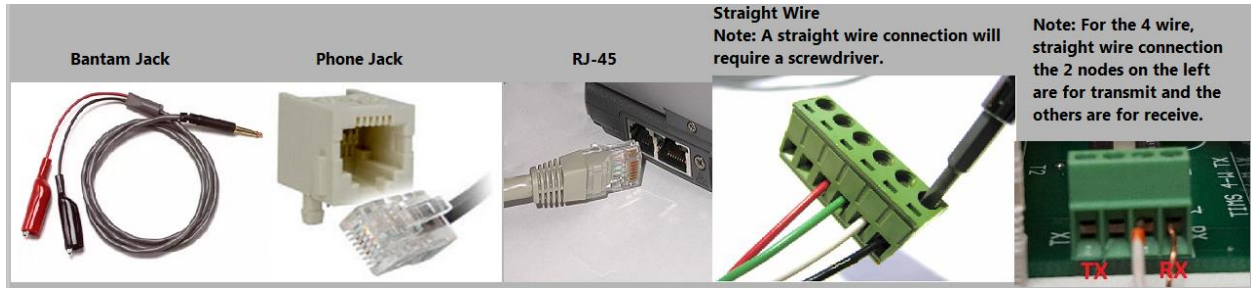
Figure 4-11: Line Interface Connections.

Configuring the software: For accurate results, the software settings must match the hardware settings. These setting include:

- Selecting the DAQ, specifically the MyDAQ contained within the connector box.
- Selecting the line impedance for the transmit and receive circuits.
- Selecting the transmit/receive mode. Standard 4-Wire is used for 4-wire impedance matching. 2-Wire is used for 2-wire mode with the hybrid circuit. Bypass ignores the output impedance. Bridged mode uses an open circuit for the impedance and can measure signals from a line with minimal effect on the system.
- Setting the Transmission Level Point (TLP). The TLP is how much the signal will be attenuated before reaching the reference point of the system, known as the 0TLP. Setting the TLP will scale signals so that the power measured at the 0TLP will equal the power indicated by the virtual test set when transmitting or receiving.

Once the initial hardware and software settings are complete the user clicks Setup Complete to begin using the virtual test set, **Figure 4-12**.
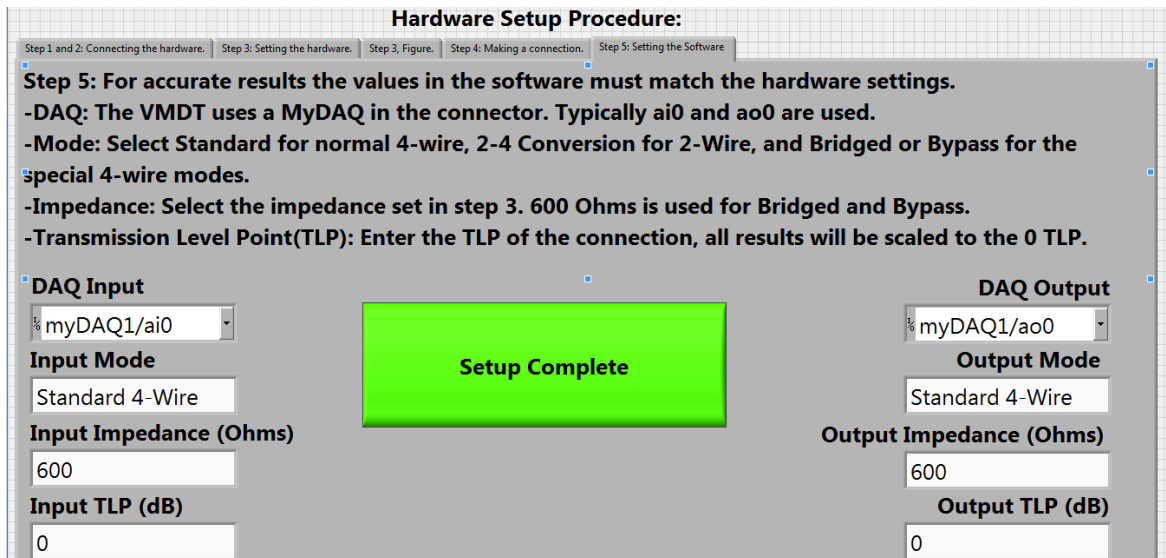
Figure 4-12: Setting the software to match the hardware.

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4.2.2 VMDT VI

The Virtual Mobile Data Terminal (VMDT) VI, **Figure 4-13**, is the master controller of the virtual test set, containing the Main Queue and all the other virtual test set functions. The VMDT VI is responsible for monitoring and controlling the states of the each of the virtual test set functions to ensure that the virtual test set runs smoothly. The primary function of the VMDT VI for the user is to provide access to the virtual test set's functions, **Figure 4-14**.
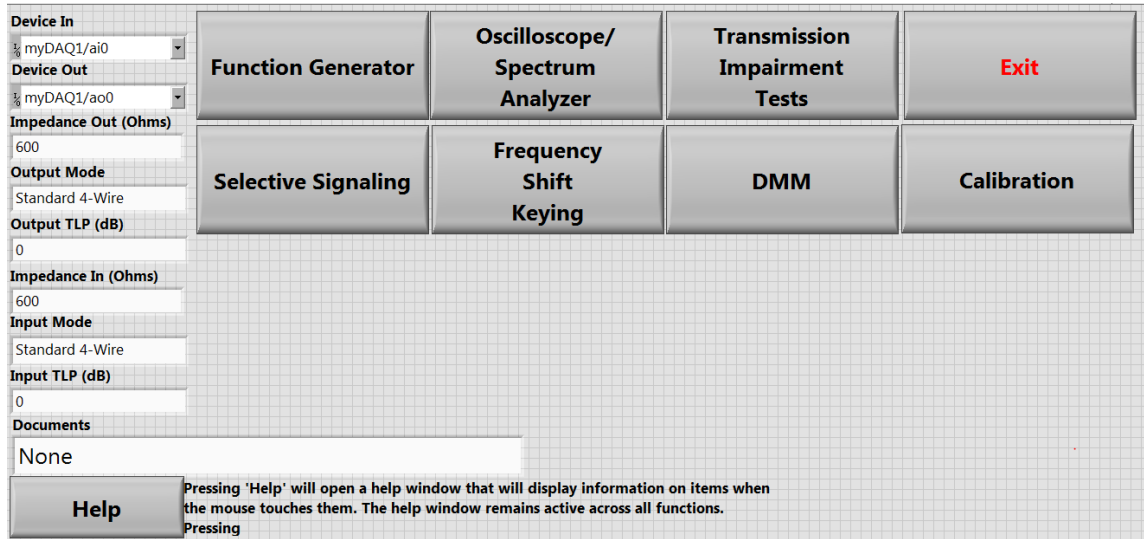
Figure 4-13: The VMDT VI.
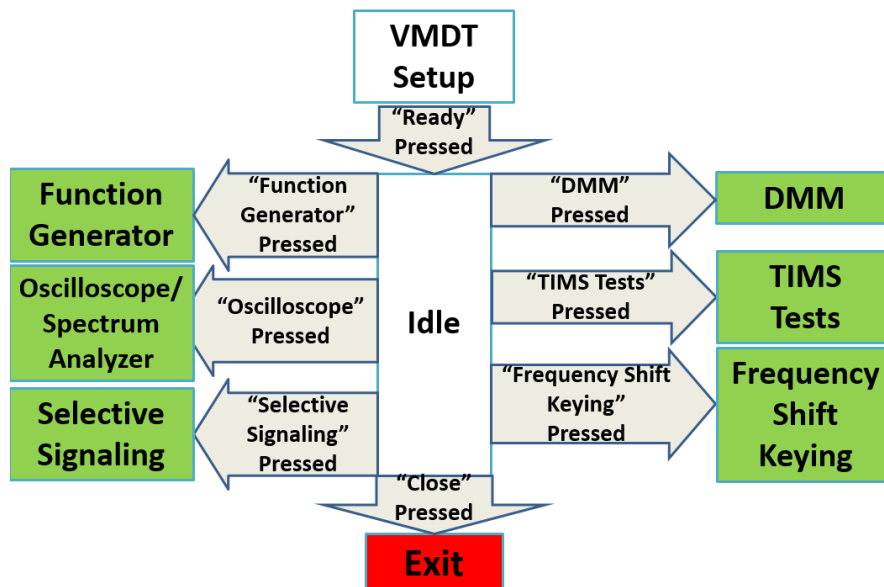
Figure 4-14: VMDT VI Flow Chart

The VMDT VI has five primary functions:

1. Provide access to all virtual test set functions.
2. Allow the change of hardware settings.
3. Provide access to reference material.
4. Toggle the help window.
5. Control communication between functions.

Provide access to all of the virtual test set functions: Whenever a function is selected the VMDT activates the associated sub-VI and deactivates any conflicting VIs. For example, if Selective Signaling is selected the virtual test set will stop transmitting and receiving signals and open the Selective Signaling sub-VI, **Figure 4-15**.
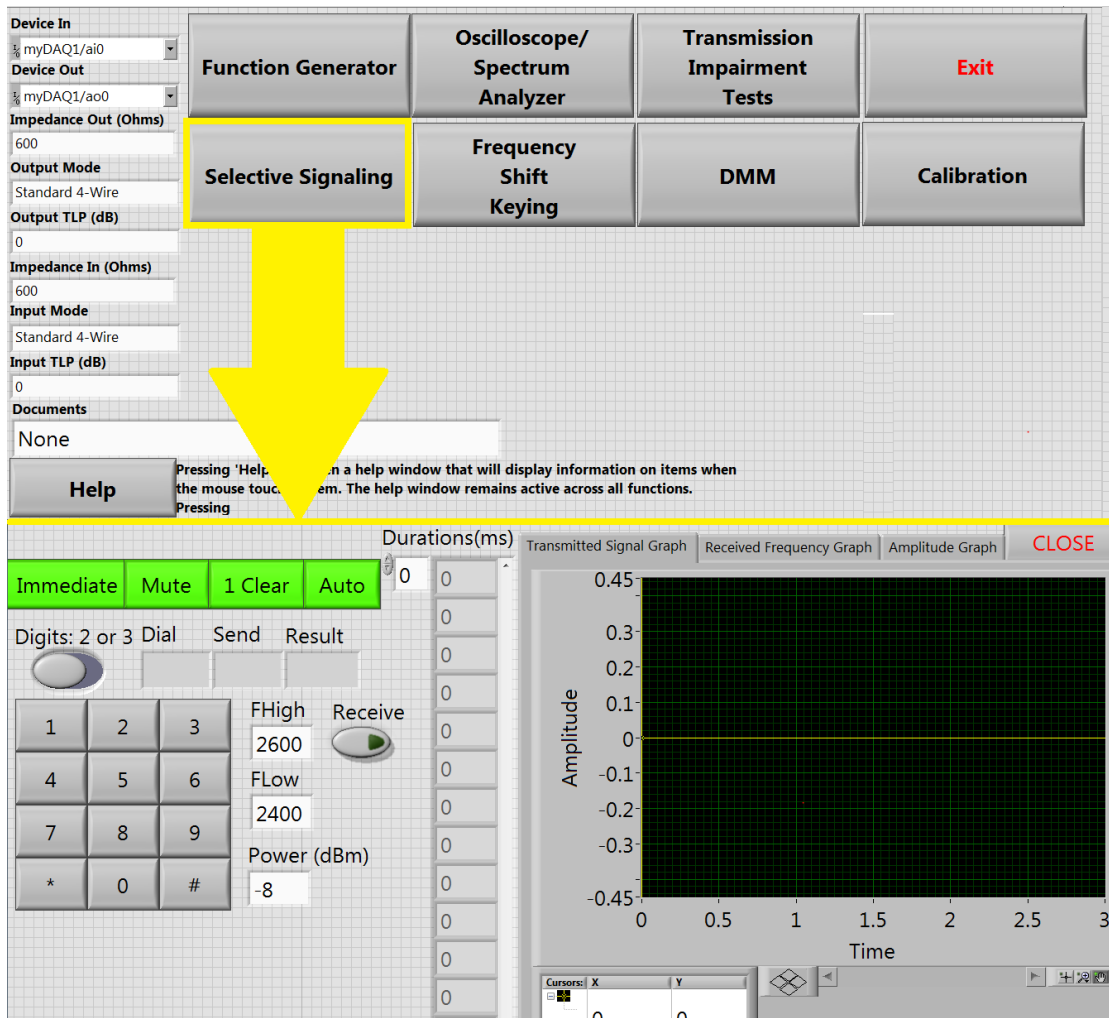
Figure 4-15: Selecting Selective Signaling.

Allow the change of hardware settings: If the settings of the physical hardware are changed the hardware settings in the software need to be changed as well. The VMDT VI provides access to all of the hardware settings so they can be changed without restarting the application. Whenever the hardware settings are changed, all functions that communicate with the hardware are reset and the hardware settings are passed to all the functions.

Provide accesses to reference material: Additional material, such as technical manuals and IEEE specifications, can be accessed through the Documents pull down menu, **Figure 4-16**.
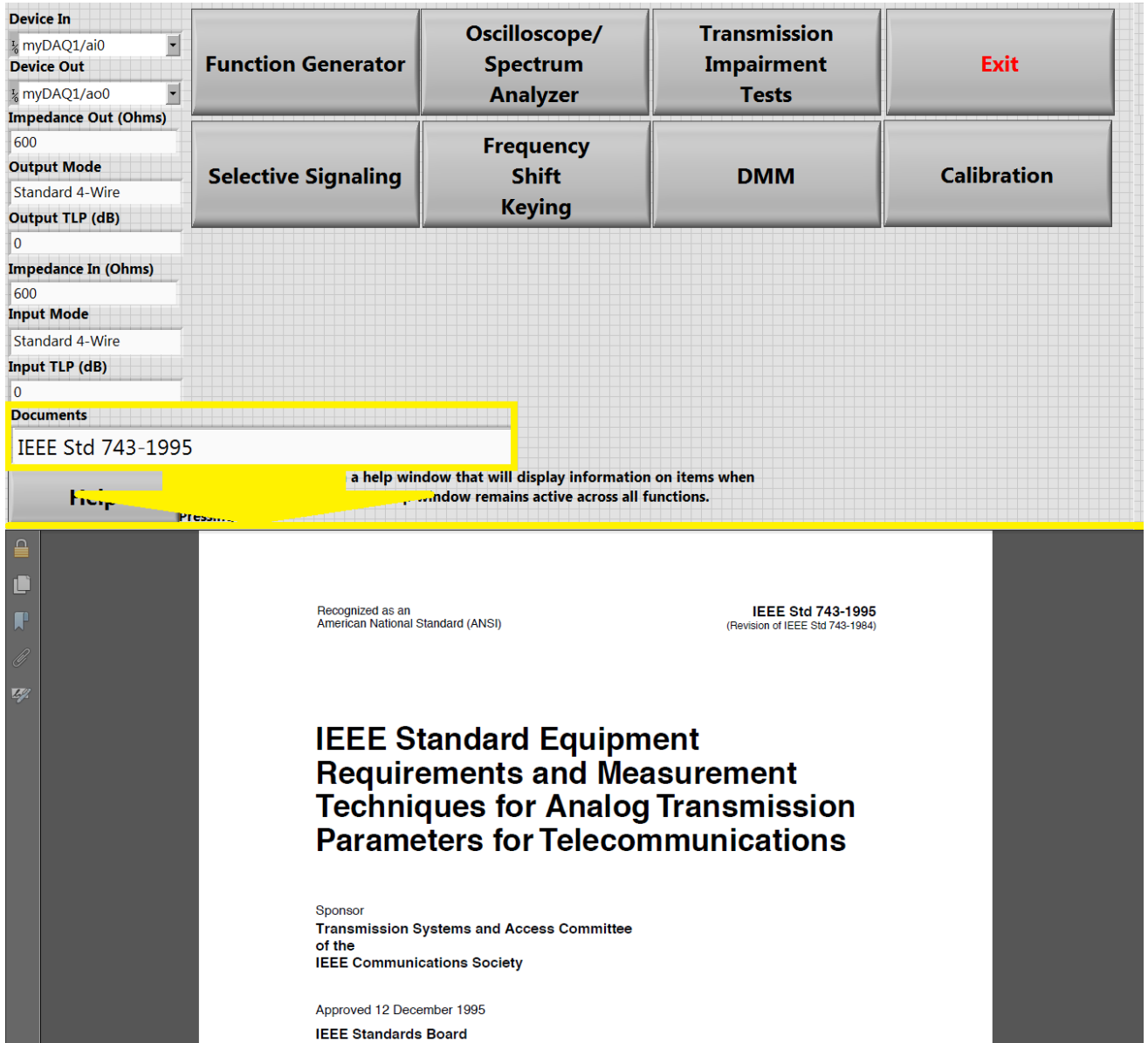
Figure 4-16: Selecting a Document.

Toggle the help window: The help window will display information about any front panel control or indicator the mouse touches, **Figure 4-17**.
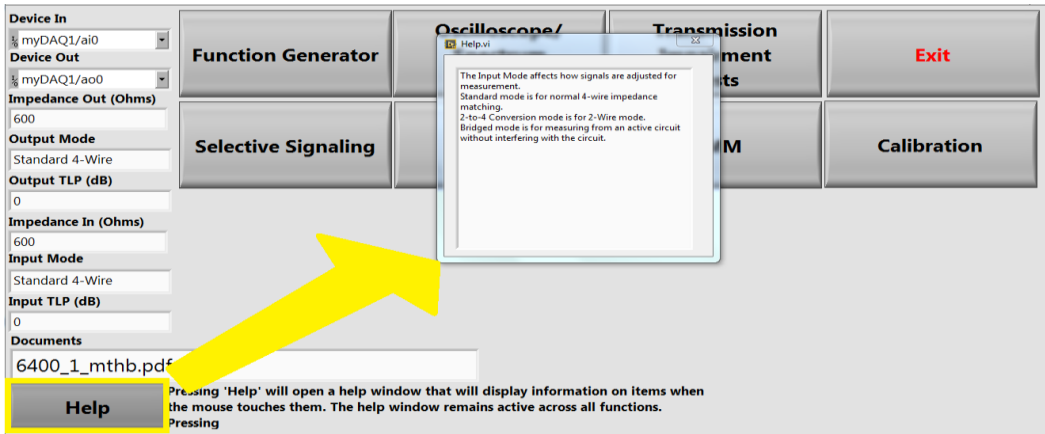
Figure 4-17: Opening the help window.

Control communication between functions: Some of the virtual test set's functions communicate with each other. However, all the communication between the functions are passed to the VMDT VI before being passed to the destination functions. Information that must reach its destination as soon as possible is sent through direct control references, all other information is handled through the queued state machine.

4.3 Function Generator

This VI is responsible for generating the continuous signals that will be transmitted. The signal that is configured will be displayed in both time and frequency domain graphs. Once transmit is selected the configured signal will be transmitted. The Front Panel of the function generator is split in half, **Figure 4-18**. The left half allows the user to configure signals to be transmitted by the virtual test set while the right half displays the configured signal. The Function Generator will continuously update the signal graph as the user changes settings, **Figure 4-19**.
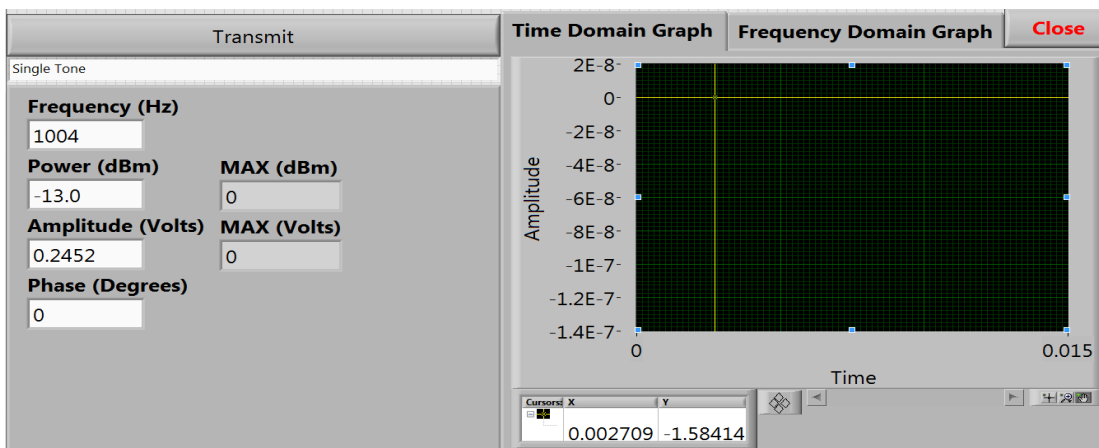
Figure 4-18: Function Generator Front Panel
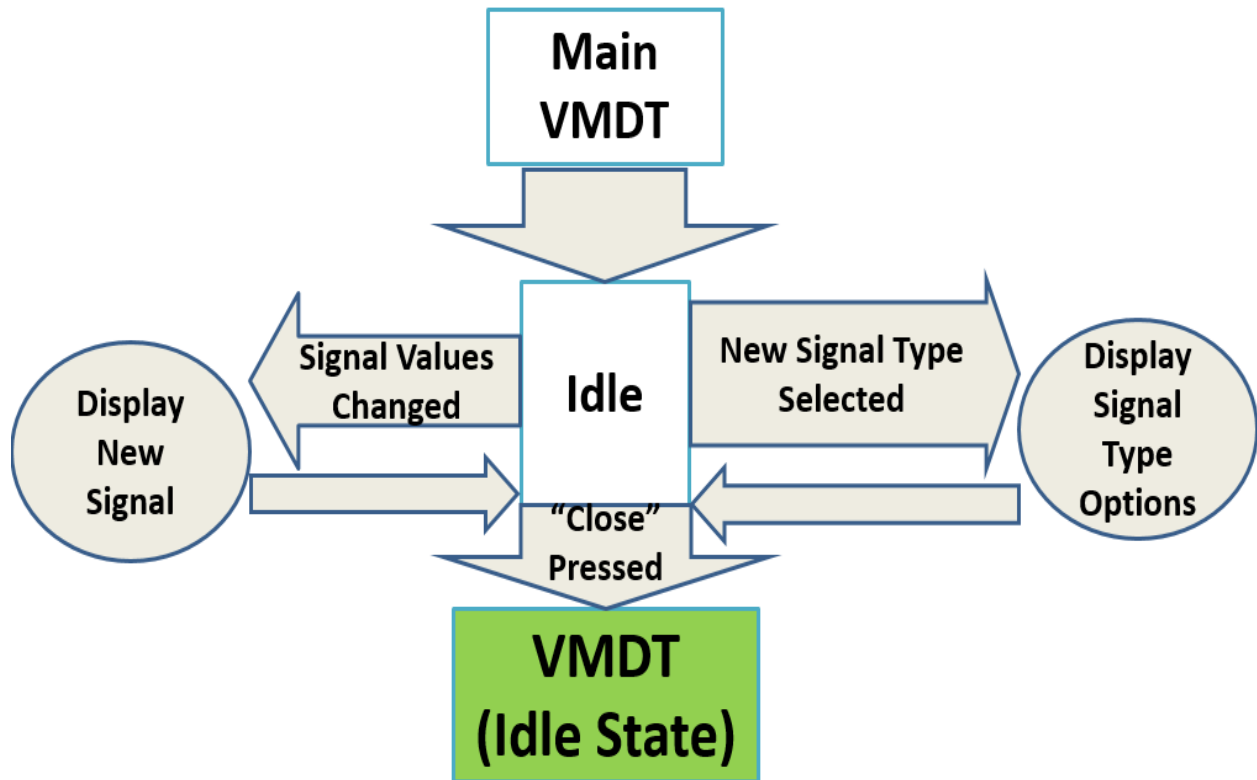
Figure 4-19: Function Generator Flow Chart

The signals the function generator can produce are broken into the following categories:

- Single Tone.
- Multi-tone
- General Function Generator
- Noise
- Amplitude Modulation
- Frequency Modulation
- Impulse Noise.

4.3.1 Single Tone Generation

Single Tone Generation generates a single sinusoidal signal and displays the maximum power and amplitude that is allowed, **Figure 4-20**. The maximum power and amplitude is based on the output limitations of the Line Interface and is dependent on the hardware settings, **Table 4-3**. The frequency, amplitude, phase, and power can all be set individually. Whenever the amplitude or power is changed the other is updated to reflect the change. The final single tone signal is generated according to, **Equation 4-1**.
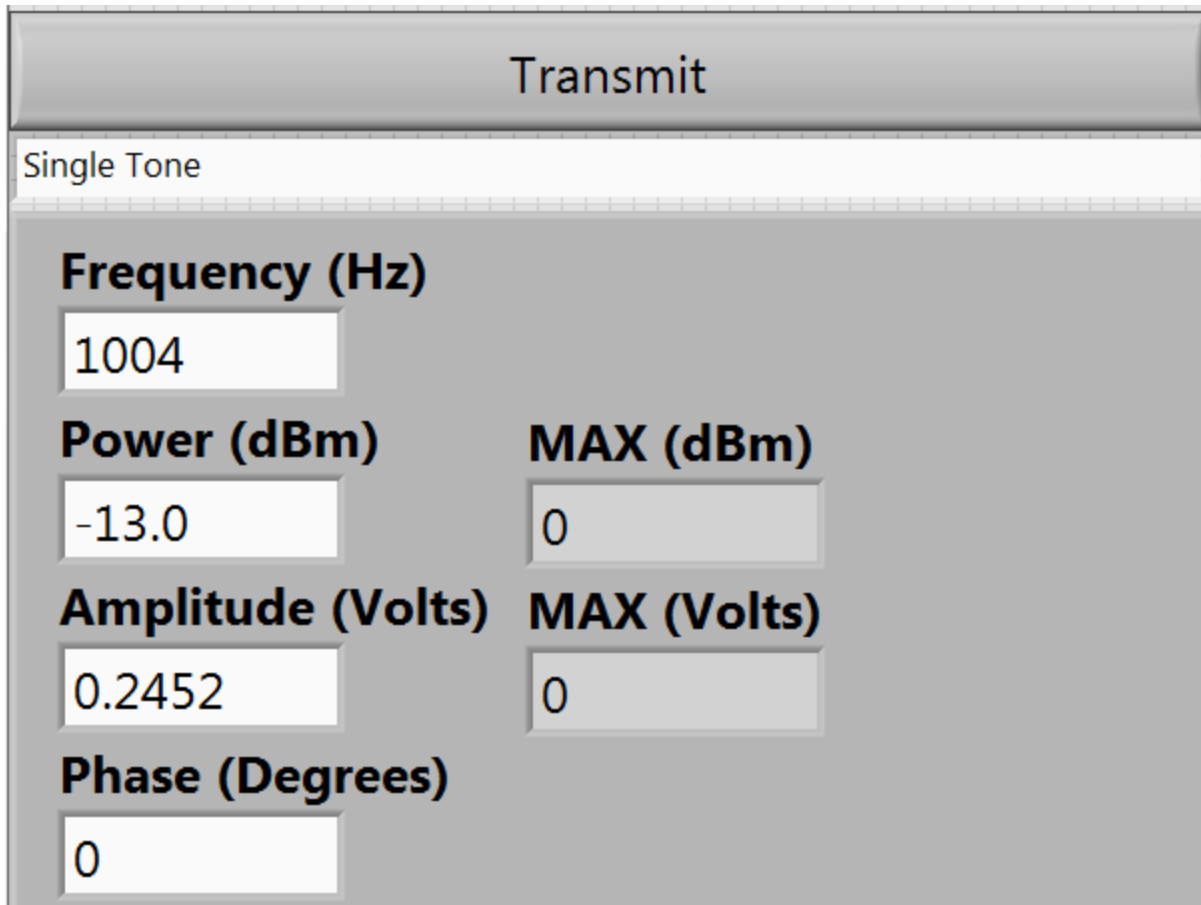
Figure 4-20: Function Generator, Single Tone Front Panel

Table 4-3: Function Generator, Maximum Single Tone Output.

Output Setting	Max Volts	Max Power
Standard, 1200Ohms	6.25	12.1
Standard, 900Ohms	6.25	13.4
Standard, 600Ohms	6.25	15.1
Standard, 135Ohms	4.32	18.4
Standard, 100Ohms	3.2	17.1
2 Wire, 1200Ohms	4.17	8.6
2 Wire, 900Ohms	4.17	9.9
2 Wire, 600Ohms	4.17	11.6
Bypass	12.5	21.1

$$signal = A \sin(2\pi f t + \phi) \tag{4-1}$$

Where A is the amplitude, f is the frequency, φ is the initial phase, and t is time.

4.3.2   Multi-Tone Generation

Multi-Tone Generation is used whenever two or more sinusoidal tones need to be transmitted at the same time, most commonly used for the standard 23-tone signal. Each individual tone can have the frequency, amplitude, and initial phase can be configured independently, **Figure 4-21**. The final signal is generated according to **Equation 4-2**. The displayed power of the signal is calculated using **Equation 4-3.** In addition to being able to customize each tone individually the user has the option of setting the entire signal to a preset, such as the 23 tone signal, using the Standard Signals control.
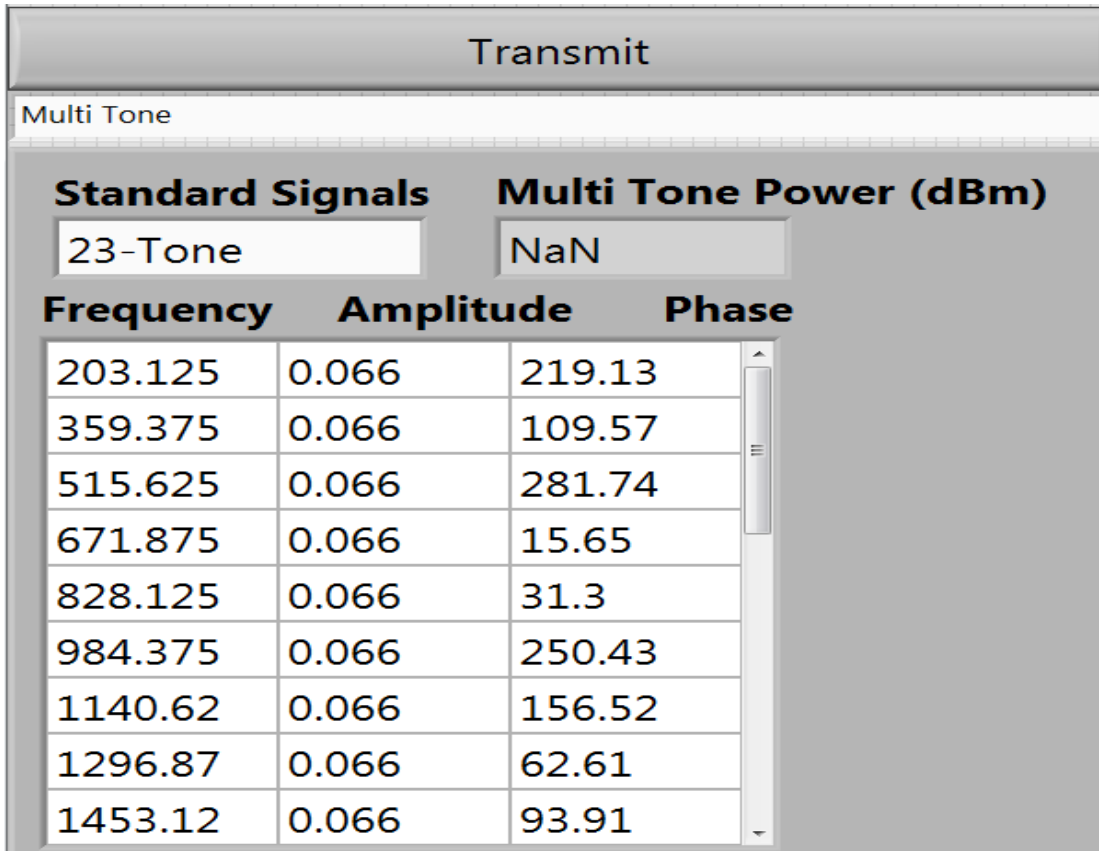
Figure 4-21: Function Generator, MultiTone Front Panel

$$\text{signal} = \sum_{n=1}^{N} A_n \sin(2\pi f_n t + \phi_n) \qquad (4\text{-}2)$$

Where A_n is the component amplitude, f_n is the component frequency, t is time, and φ_n is the component initial phase.

$$T_{pow(dBm)} = 10 * \log \frac{\sum_{n=1}^{N} \left(\frac{A_n{}^2}{2}\right)}{0.001*Z} \qquad (4\text{-}3)$$

Where An is the amplitude of the component tone, Z is the impedance, and Tpow(dBm) is the total power in dBm.

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4.3.3 General Function Generation

The general function generator is used to generate non-sinusoidal waveforms, signals with DC components, and perform frequency sweeps. The controls provided allow the user to choose the signal type, set all of the signal components, and set a frequency sweep, **Figure 4-22**. During a frequency sweep additional settings take effect including the range of frequencies, the number of steps in the sweep, and how long each tone will be held. The general function generator is able to generate sine, square, triangle, and sawtooth waves.
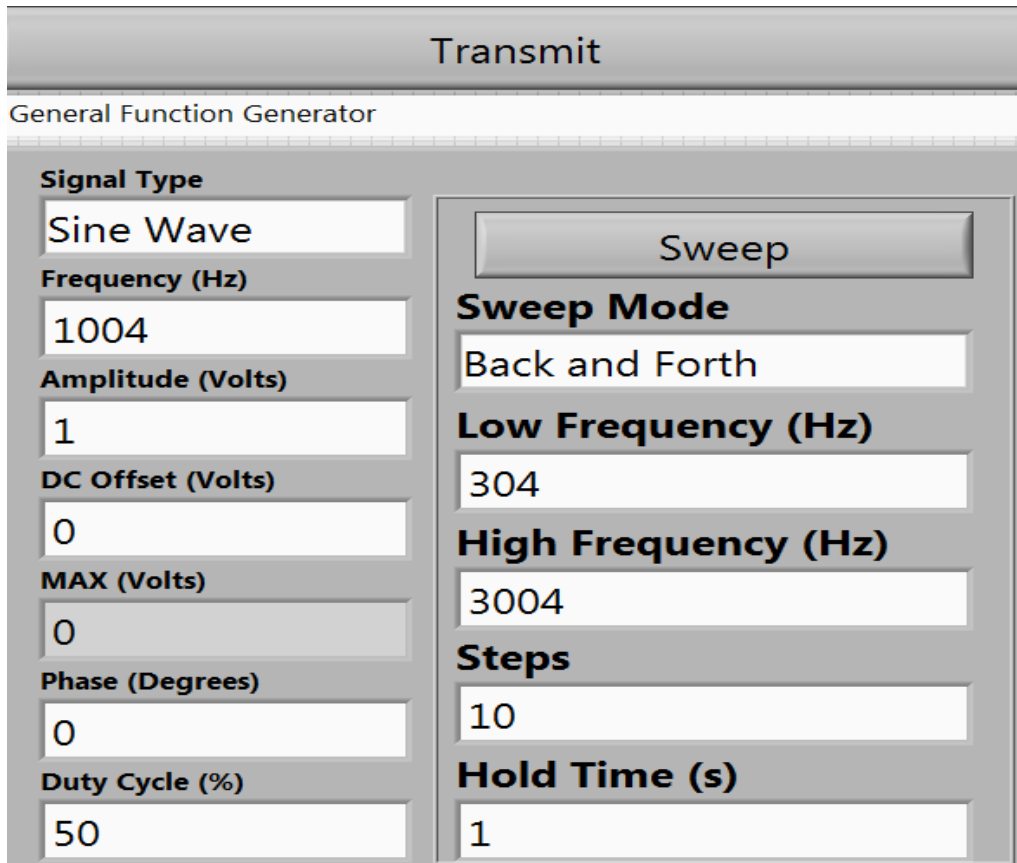
Figure 4-22: Function Generator, General Function Generator Front Panel

4.3.4 Noise Generation

Noise Generation provides White Gaussian noise and a single tone sinusoidal signal for the purpose of evaluating a test set's ability to measure the signal to noise level. The noise signal is band limited with a customizable upper and lower cutoff frequency. The total noise power is dependent on the bandwidth and the amplitude of the noise signal. The total noise power is calculated prior to being added to the test tone using the Noise in Band sub-VI to calculate the total power between the two cutoff frequencies. The controls provided allow the user to set the components of the holding tone, the noise band, and the maximum noise amplitude, **Figure 4-23**.
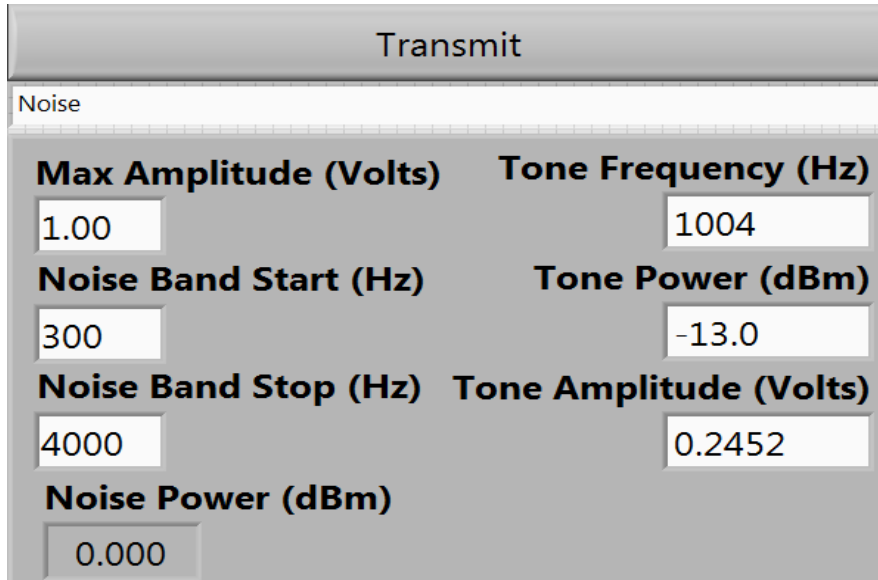
Figure 4-23: Function Generator, Noise Front Panel

4.3.5 Frequency Bursts

The bursts are single tone signals which are active for a percent of time determined by a duty cycle. This type of signal is used to test the averaging capability of the Impairment Measurement Tests. If a test does not have a long enough averaging period then the results will be sporadic and unreliable. If the averaging period is too long the results will not properly measure sporadic noise levels. The Function Generator has the capability of generating two sets of single tone parameters that will be alternated between at a rate set by the gating frequency. The controls provided allow the user to set the signal components of the two tones, the duty cycle of each tone, and the gating frequency, **Figure 4-24**.
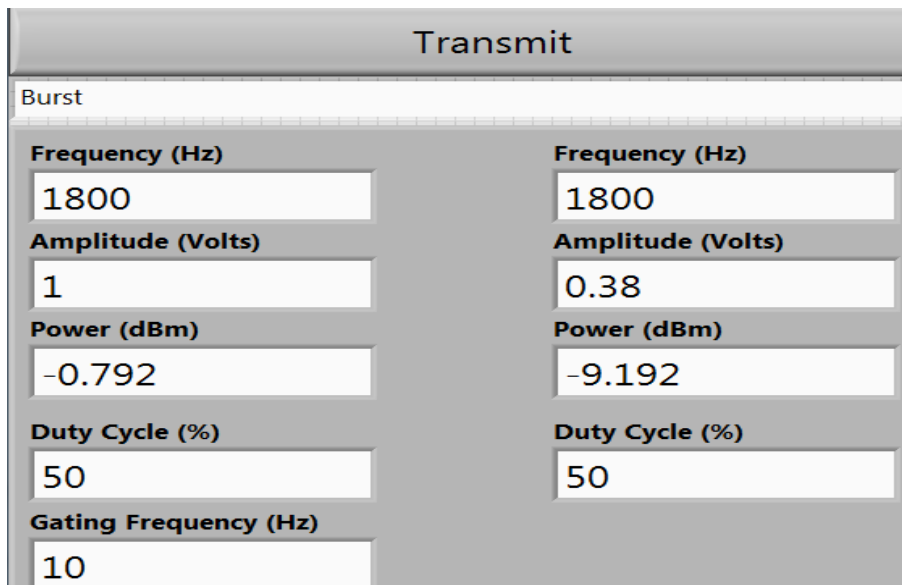
Figure 4-24: Function Generator, Burst Front Panel.

4.3.6   Amplitude Modulation Generation

Amplitude modulation is the process of changing the amplitude of a high frequency signal, known as the carrier signal, proportionally to a lower frequency signal, known as the message signal. Amplitude modulated signals are used to carry messages over a higher frequency range, allowing multiple signals to be transmitted at the same time using different frequency bands. The Function Generator can only produce a single carrier frequency with a single tone message according to **Equation 4-4 [3]**. The virtual test set primarily uses Amplitude Modulation to validate a test set's ability to properly differentiate between Amplitude and Phase Jitter. The provided controls allows the user to set the signal components of the carrier and message tones, **Figure 4-25**, and the displayed power is calculated according to **Equation 4-3.**

$$AM\ Signal = A_c \sin(2\pi\ f_c t + \phi_c) * \frac{A_m \sin(2\pi f_m t + \phi_m) + A_c}{A_c} \tag{4-4}$$

Where A_c is the amplitude of the carrier, f_c is the frequency of the carrier, φ_c is the initial phase of the carrier, A_m is the amplitude of the message, f_m is the frequency of the message, φ_m is the initial phase of the carrier, and t is time.
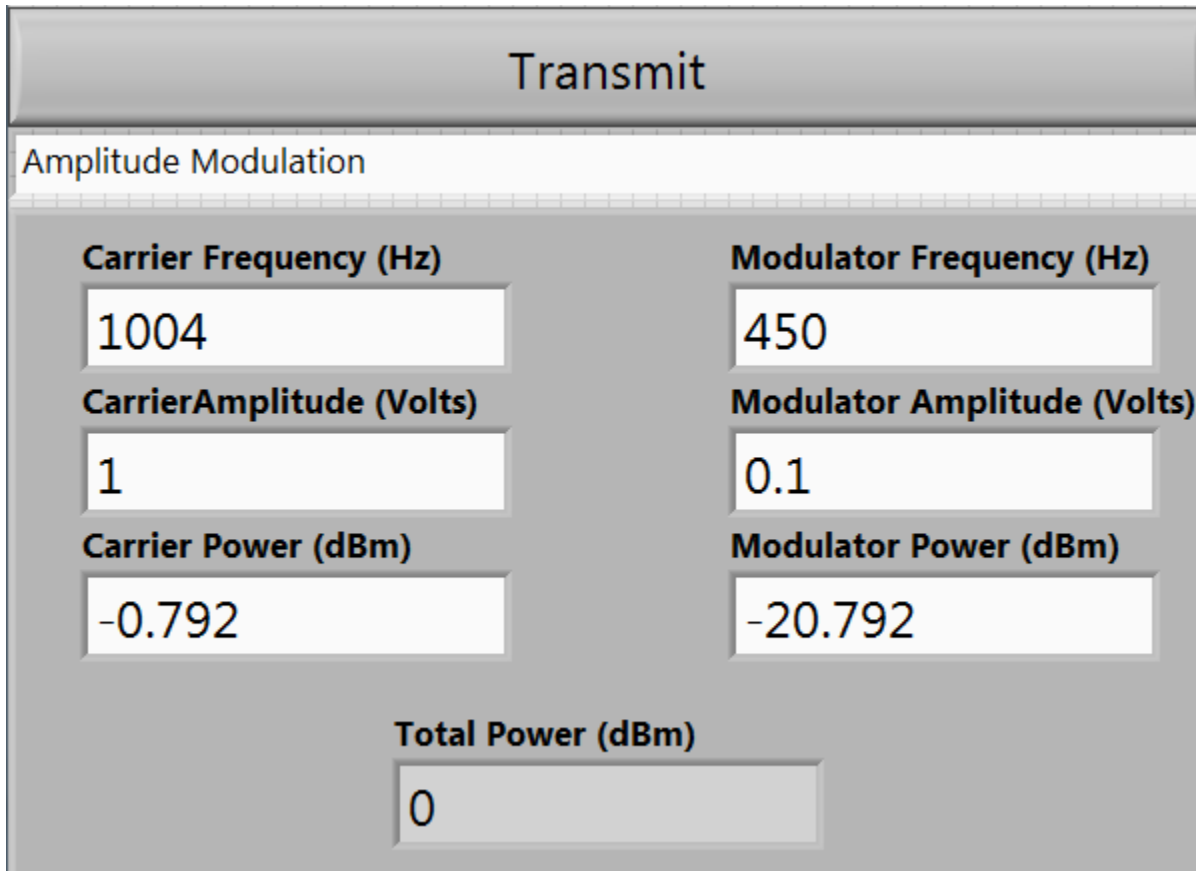
Figure 4-25: Function Generator, Amplitude Modulation Front Panel.

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4.3.7   Frequency Modulation Generation

Frequency modulation is the process of changing the frequency of a high frequency signal, known as the carrier signal, proportionally to a lower frequency signal, known as the message signal. Frequency modulated signals are used to carry messages over a higher frequency range, allowing multiple signals to be transmitted at the same time using different frequency bands. The Function Generator can only produce a single carrier frequency with a single tone message that is mathematically generated point by point according to **Equation 4-5** for frequency modulation and **Equation 4-6** for phase modulation **[3]**. The virtual test set primarily uses Frequency Modulation to validate a test set's ability to properly differentiate between Amplitude and Phase Jitter. The provided controls allow the user to set the signal components of the carrier and message signals, change the deviation constant, and set a duty cycle for the modulation, **Figure 4-26**.

$$FM\ Signal = A_c \sin(2\pi f_c(A_m \sin(2\pi f_m t + \phi_m))t + \phi_c) \tag{4-5}$$

$$PM\ Signal = A_c \sin(2\pi f_c t + \phi_c(A_m \sin(2\pi f_m t + \phi_m))) \tag{4-6}$$

Where in both equations A_c is the amplitude of the carrier, f_c is the frequency of the carrier, ϕ_c is the initial phase of the carrier, A_m is the amplitude of the message times the deviation constant, f_m is the frequency of the message, ϕ_m is the initial phase of the carrier, and t is time.

Figure 4-26: Function Generator, Frequency Modulation Front Panel.

4.3.8 Impulse Noise Generation

Impulse noise is any kind of undesired instantaneous or near instantaneous change to the signal. The Function Generator is capable of generating a two tone signal with square pulses, gain hits, or phase hits added to the signal. Square pulses are high amplitude noise spikes that are directly added to the signal, gain hits are sudden changes to the amplitude of the signal, and phase hits are sudden changes to the phase of the signal. Square pulses, gain hits, and phase hits can each have their amplitude and duration changed along with the two tone components, **Figure 4-27**. These signals are primarily used to test and validate a test set's sensitivity to impulse noise.
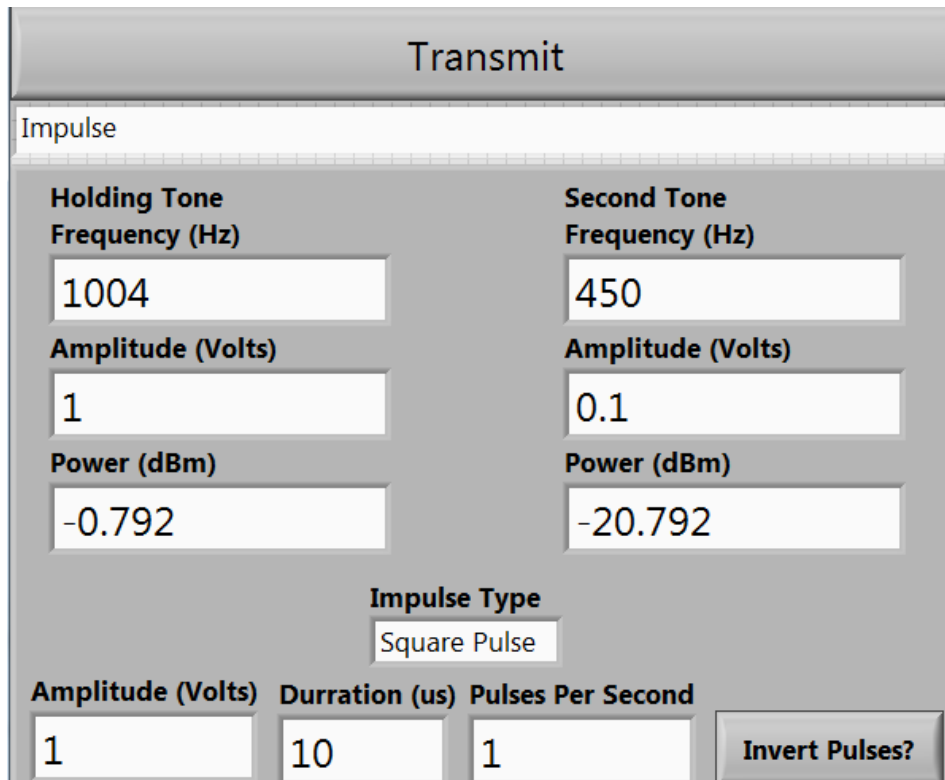
Figure 4-27: Function Generator, Impulse Front Panel.

4.4 Transmission Impairment Measurement

A Transmission Impairment Measurement Set is used to transmit and receive test signals on a transmission line to measure impairments that occur on the line. The Transmission Impairment Measurement section of the virtual test set is capable of performing the analog transmission impairment tests as specified by IEEE Std. 743-1995 **[1]**. In addition, when the virtual test set is looped back to itself or connected to another virtual test set it is capable of performing a series of tests automatically, communicating with the Function Generator of the virtual test set to change signals for use in different tests, and generating a report based on the results. The Front Panel of the Transmission Impairment Measurement VI is split into the measurement tab, where the user can select

measurements and change measurement settings, and the graphs tab, where the received signal can be viewed in the time and frequency domain, **Figure 4-28**. In addition, the user is able to select the filter used in various tests. The time domain and frequency domain graphs can display the received signal before or after the filter.
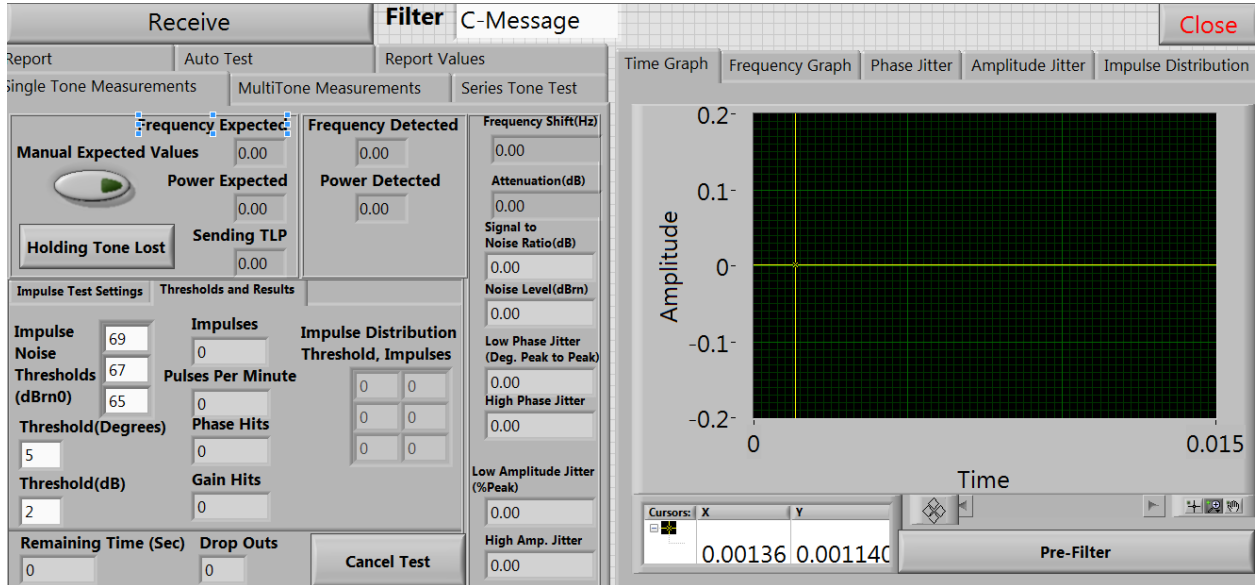
Figure 4-28: Transmission Impairment Measurement Front Panel.
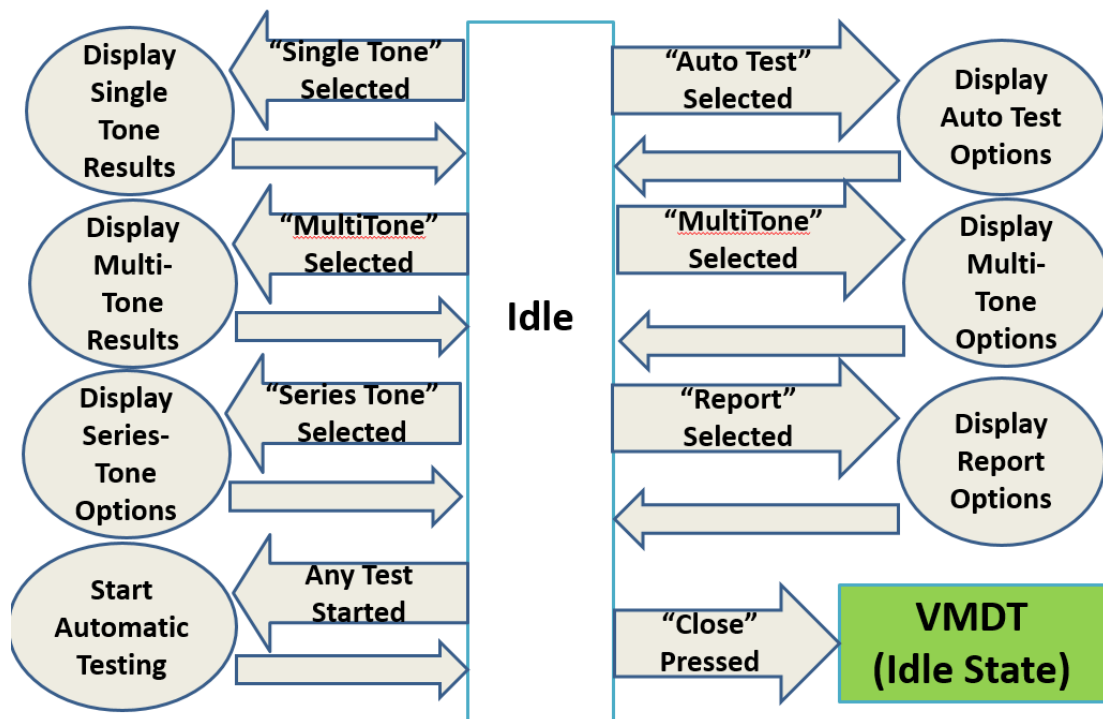
Figure 4-29: Transmission Impairment Measurement User Flow Chart

The purpose of the Transmission Impairment Measurement VI is to perform the analog signal impairment tests available to a Halcyon TIMS. These test were designed to meet IEEE standards and must provide the user with a range of settings and thresholds for the many tests the virtual test set provides. In addition the virtual test set is capable of performing tests automatically, saving the results of any test taken, and generating a report based on the results. To accomplish these goals, the Transmission Impairment Measurement section of the virtual test set is broken into tabs. When the user selects a new tab the virtual test set displays the options associated with and begins performing any continuous measurements associated with the tab, **Figure 4-29**. The tabs and their function include:

1. Single Tone Tests: This section is used to perform impairment measurements on continuous single tone signals.

2. Multi-Tone Tests: This section is used to perform impairment measurements on continuous multi-tone signals.

3. Series Tone Tests: This section is used to perform impairment measurements using a series of continuous single tone signals.

4. Report Setup: This section is where non-numeric values of a report are entered and where a report can be manually generated.

5. Auto Test: This section is where a series of tests can be setup to be ran automatically with a report optionally generated after all tests are completed.

6. Report Values: This section is where values can be latched for use in a manual report.

7. Graphs Display: This section displays graphs of the received signal as well as graphical representations of some of the tests.

4.4.1   Single Tone Tests

Single tone testing is accessed by selecting 'Single Tone Tests' from the tab. This section is used to perform impairment measurements on continuous single tone signals. From here the user can measure:

- Frequency Shift
- Attenuation
- Signal to Notched Noise Ratio
- Noise Level
- Phase Jitter
- Amplitude Jitter
- Impulse Noise:
    - Impulses
    - Gain Hits

- Phase Hits
- Drop Outs

The Single Tone Measurement tab is further broken down into subsections, **Figure 4-30**. The 'Expected Signal' section determines the values frequency shift and attenuation are based off of and can be set manually or automatically, which uses the tone transmitted with the function generator. The 'Detected Signal' section displays the frequency and power of the detected signal. The 'Continuous Measurements' section displays the results of all the single tone measurements that are made continuously. Finally, the Impulse Testing sub-tab allows the user to configure the settings for an impulse noise test.
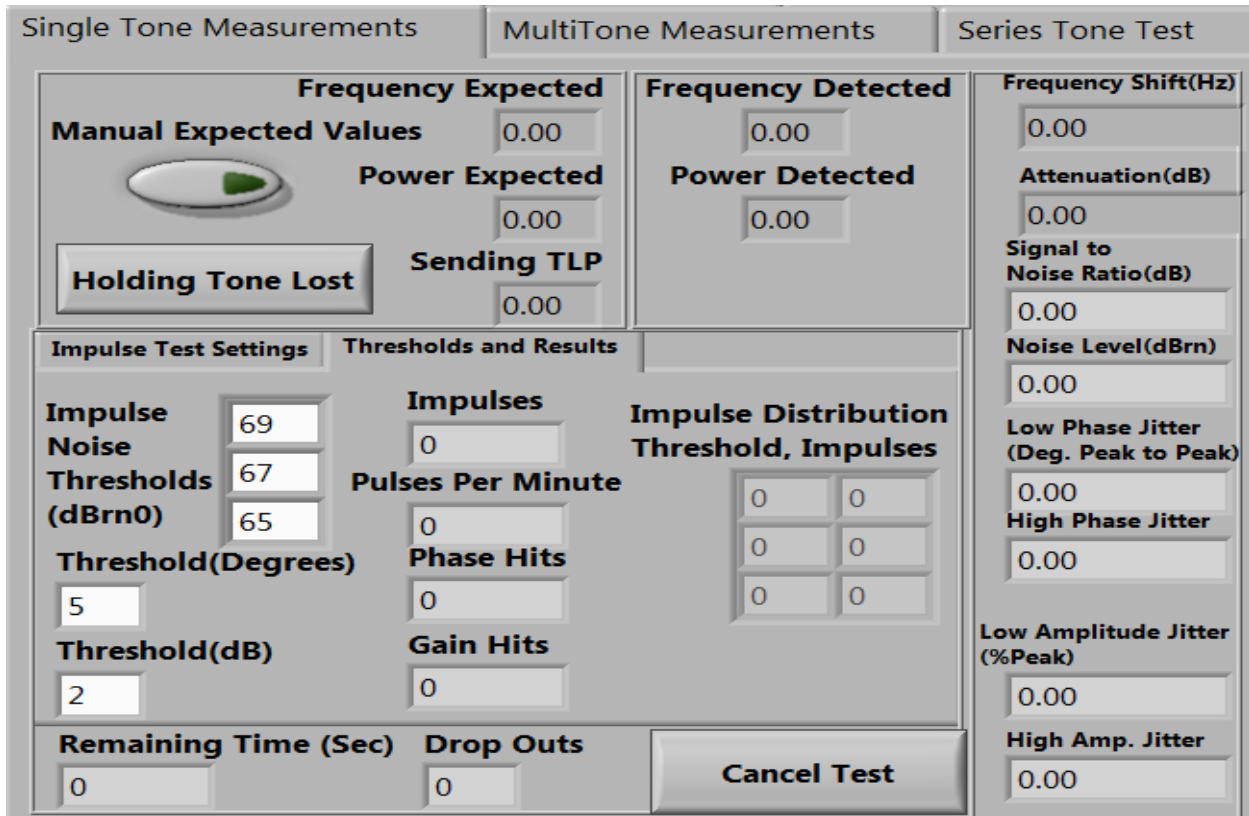
Figure 4-30: Transmission Impairment Measurement Tab Control, Single Tone Measurements Tab

4.4.1.1 Frequency Shift

Frequency Shift is the changed in frequency a signal undergoes between the point it was transmitted and the point it was received, **Equation 4-7**. The received frequency is directly measured by the virtual test set however the virtual test set must be given the original frequency to calculate the difference. If the virtual test set is in loop back mode the virtual test set will use the frequency being transmitted by the function generator as the original frequency. If the virtual test set is not in loop back

mode the user may press 'Manually Selected Values' to enter the expected frequency manually, **Figure 4-31**, which will be used as the transmitted frequency.

$$received\ frequency - transmitted\ freqency = change\ in\ frequency \qquad (4\text{-}7)$$

	Frequency Expected	Frequency Detected	Frequency Shift(Hz)
Manual Expected Values	0.00	0.00	0.00
	Power Expected	**Power Detected**	**Attenuation(dB)**
	0.00	0.00	0.00
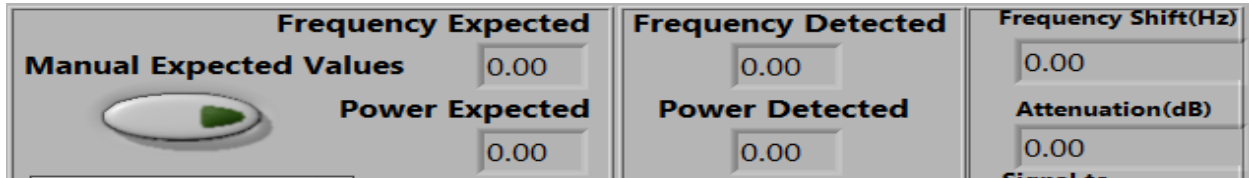
Figure 4-31: Single Tone Measurements Tab, Frequency Shift Section

4.4.1.2 Attenuation

Attenuation is the changed in signal power between the point it was transmitted and the point it was received, **Equation 4-8**. The received signal power is directly measured by the virtual test set however the virtual test set must be given the original power to calculate the difference. If the virtual test set is in loop back mode the virtual test set will use the power transmitted by the function generator as the original power. If the virtual test set is not in loop back mode the user may press 'Manual Expected Values' to enter the expected power manually, which will be used as the transmitted power. Frequency Shift and Attenuation are measured at the same time and are typically measured after a C-Message Filter is applied, **Figure 4-32**.

$$transmitted\ power - received\ power = attenuation \qquad (4\text{-}8)$$
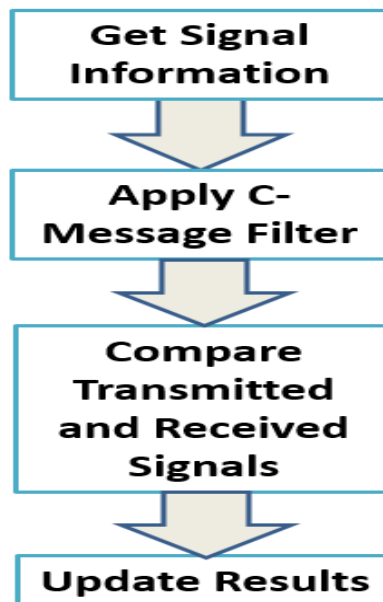
Figure 4-32: Frequency Shift and Attenuation Flow Chart.

4.4.1.3 Signal to Notched Noise Ratio

This test determines the signal to noise ratio by comparing the total power of the received signal before and after being passed through a notch filter, **Equation 4-9**. A notch filter is a narrow band-stop filter centered on the test tone frequency and is used to eliminate the test tone being sent through the system. The test tone, known as the holding tone, is necessary to generate noise that is only produced when a signal is being sent through the system but must be eliminated before measuring the strength of the noise signal. For the purposes of audio range testing the notch filter is commonly paired with a C-message filter, which mimics the pass band of human hearing. This special case of Signal to Notched Noise Ratio Test is known as a C-Notched Noise Ratio Test, **Figure 4-34**. Because any noise that passes through the C-Message filter will be audible on the receiving end of the system, this test is useful for determining how much noise in the system will be audible to users.

$$P_r - P_n = Notch\ NR \tag{4-9}$$

Where: P_r=Power of received signal, P_n=Power of notched signal
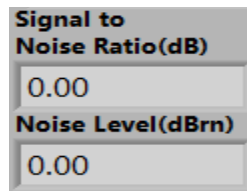
Figure 4-33: Single Tone Measurements Tab, Noise Section
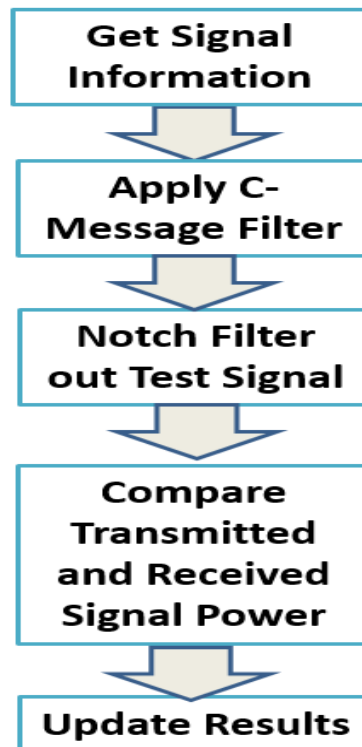
Figure 4-34: Signal to C-Notched Noise Test Flow.

4.4.1.4 Noise Level

This test determines the noise level of a single tone signal. Often measured in tandem with a notched noise ratio test, this test measures the total power in a signal after it has been passed through a notched filter. The noise power is calculated according to **Equation 4-10** and is largely dependent on the bandwidth of the filter. The Noise Level and Signal to Noise Ratio tests are performed at the same time and the results are displayed together, **Figure 4-33**.

$$P_N = 10Log\left(\frac{1000(A_{RMS})^2}{I}\right)$$ (4-10)

Where: P_N= the noise power of the signal, A_{RMS}= the Root Mean Square of the signal, and I is the Impedance.

4.4.1.5 Phase Jitter

This tests measures irregular changes in the phase of a sinusoidal signal and is often paired with a measure of amplitude jitter, **Figure 4-35**. The received signal is phase demodulated to produce a Phase Change vs Time graph, using **Equation 4-11**, which is displayed in the graphs section of the front panel. The Phase Change vs Time signal is passed through bandpass filters to isolate frequency ranges which are the most common source of Phase Jitter. These frequencies are 4-300Hz for Low Frequency Phase Jitter and 20-300Hz for High Frequency Phase Jitter. The difference between maximum and minimum of the resulting signal is taken and displayed in Degrees Peak to Peak. This test is performed to check the ability of the system to recreate the phase of a signal.

$$P_h(t) = tan^{-1}(\frac{H(f)(t)}{f(t)})$$ (4-11)

Where $P_h(t)$ is the Phase vs Time signal, f(t) is the original signal, and H(f)(t) is the Hilbert transform of the signal.
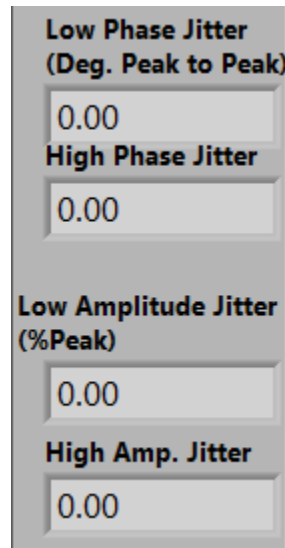
Figure 4-35: Single Tone Measurements Tab, Jitter Section

4.4.1.6 Amplitude Jitter

This tests measures changes in the amplitude of a sinusoidal signal and is often paired with a measure of phase jitter. The received signal is amplitude demodulated, using **Equation 4-12**, to obtain the Amplitude vs Time signal. The average of the Amplitude vs Time signal is subtracted to obtain the Amplitude Change vs Time graph which is displayed in the graphs section of the front panel. Finally this Amplitude Change vs Time signal is passed through bandpass filters to isolate frequency ranges which are the most common source of Jitter. These frequencies are 4-300Hz for Low Frequency Amplitude Jitter and 20-300Hz for High Frequency Amplitude Jitter. The resulting signal divided by the original amplitude and the maximum deviation is displayed as % Peak. Both Amplitude Jitter and Phase Jitter are performed using the same steps and are often passed through a C-Message Filter before being measured, **Figure 4-36**.

$$A(t) = \sqrt{H(f)(t)^2 + f(t)^2} \qquad (4\text{-}12)$$

Where A(t) is the Amplitude vs Time signal, f(t) is the original signal, and H(f)(t) is the Hilbert transform of the signal.
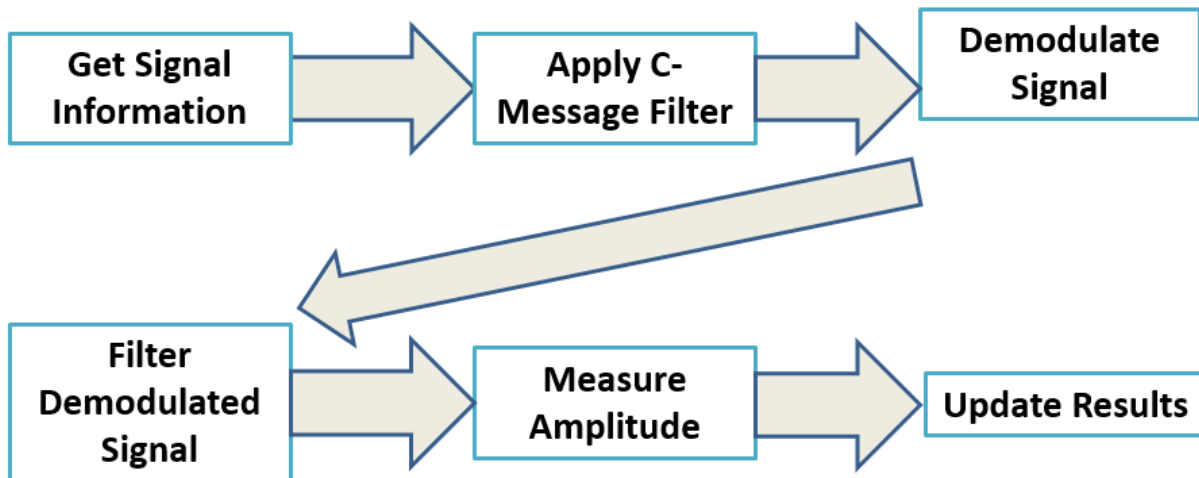
Figure 4-36: Jitter Measurement Flow Chart

4.4.1.7 Transient Measurements

These tests measures high amplitude but short duration changes to audio range signals. These tests are broken into Impulse Noise, Gain Hits, Phase Hits, and Dropouts and are often measured in parallel with one another. Each of these tests are performed by sending a holding tone through the system for a period of time and counting each time the signal deviates by more than a given threshold, **Figure 4-38**. Typically these tests are run for 15 minutes. The transient measurement controls are placed in sub-tabs on the single tone measurements tab, **Figure 4-37**. The first tab provides the settings for the timed test while the second tab provides the thresholds settings and displays the results.
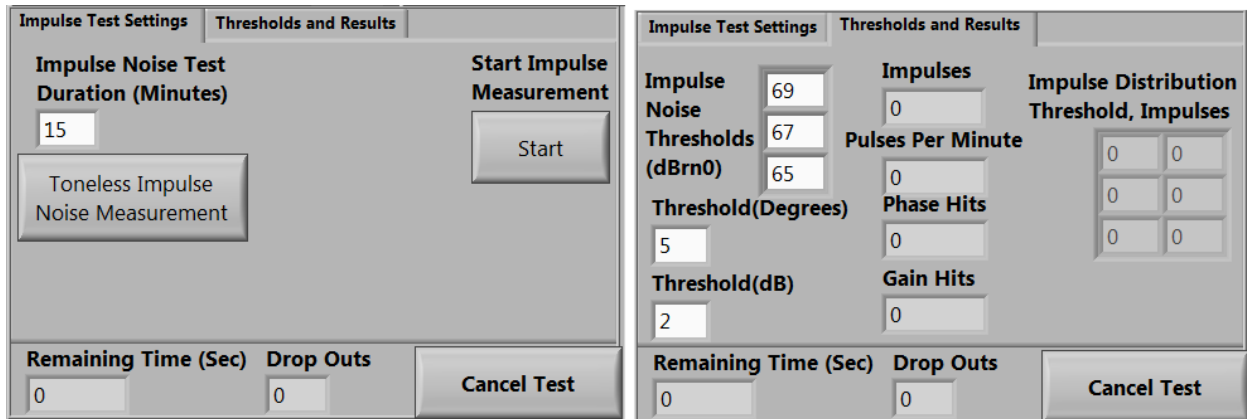
Figure 4-37: Single Tone Measurements Tab, Impulse Noise Tab
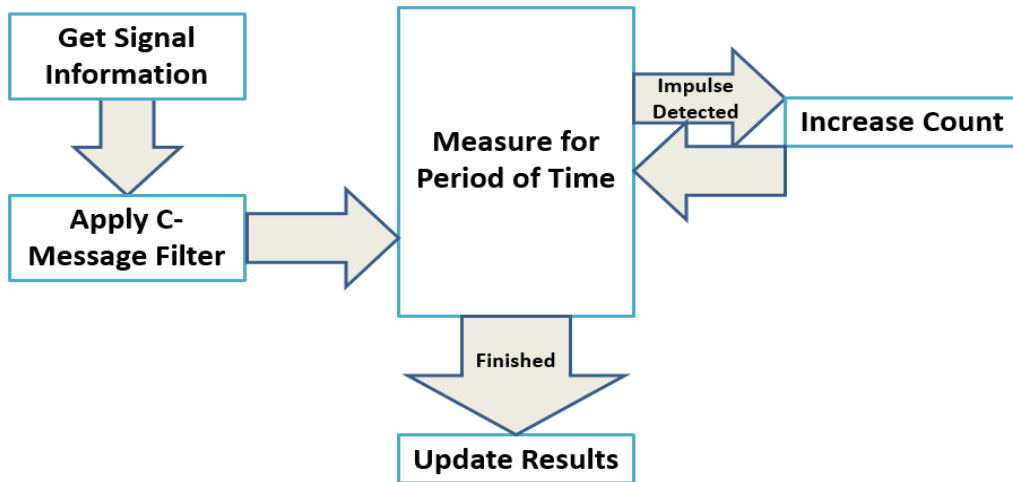
Figure 4-38: Impulse Noise Test Flow Chart.

4.4.1.7.1 Impulse Noise

Impulse Noise is defined as a high amplitude noise signal that lasts less than 4ms. Typically caused by sudden changes to the system, such as relays being flipped or lightning strikes, these noise pulses cause irritating pops and clicks in audio signals. Impulse noise can be measured two ways: with a holding tone and without a holding tone. Without a holding tone, the test measures the signal after it has been passed through a filter, typically a C-Message filter, and counts each time the signal exceeds a set threshold. With a holding tone, the signal is passed through a notch filter to eliminate the holding tone before the impulse measurement is made. In addition, when an impulse noise test is performed with a holding tone impulses will be ignored during a dropout in the signal. The virtual test set provides three thresholds for impulse noise which all have exclusive independent counters, meaning that an impulse exceeding the middle threshold but not the largest one will not be counted by any other threshold. However, by IEEE standards, any impulse exceeding the highest threshold by at least 6 dB will be counted by all thresholds [1]. In addition to counting the impulses, the virtual test set generates a histogram of the impulses indicating when each impulse was counted.

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4.4.1.7.2 Gain Hits

Gain Hits are sudden changes to the amplitude of an audio range signal and is characterized by a spike in volume on the receiving end. Gain hits are measured by sending a holding tone through the system and continuously measuring the amplitude of the signal. Gain Hits are not counted during a Dropout in the signal and when they occur they must last longer than the qualification interval, typically 3.5 periods of the holding tone, before they are counted.

4.4.1.7.3 Phase Hits

Phase Hits are sudden changes to the phase of an audio range signal. Phase hits are measured by sending a holding tone through the system and continuously measuring the phase of the signal. Gain Hits are not counted during a Dropout in the signal and when they occur they must last longer than the qualification interval, typically 3.5 periods of the holding tone, before they are counted.

4.4.1.7.4 Drop Outs

Dropout are a loss in power greater than 12 dB compared to the original power of the signal. Dropouts must last longer than the qualification interval, typically 3.5 periods of the holding tone, before they are counted. Furthermore, once a dropout has occurred no transient measurements may be made until at least one second after the dropout ends.

4.4.2 Multi-Tone Tests

Multi-Tone tests are tests that are performed using signals with two or more simultaneous tones. The virtual test set utilizes the standard 23-Tone signal in all of its multi-tone tests. These tests include Intermodulation Distortion (IMD) and Envelope Delay Distortion (EDD). The 23-Tone signal has predetermined frequency and initial phase settings listed in **Table 4-4**. The Multi-Tone tab displays the results of the IMD and EDD tests as well as the power level of each of the 23-Tones received, **Figure 4-39** **[1]**.

Table 4-4: 23-Tone Test Components [IEEE Std. 743]

Tone	Δf Mult.	Frequency in Hz	$\Delta\varnothing$ Mult.	Phase in deg.	Tone	Δf Mult.	Frequency in Hz	$\Delta\varnothing$ Mult.	Phase in deg.
1	13	203.125	14	219.13	13	133	2078.125	13	203.48
2	23	359.375	7	109.57	14	143	2234.375	5	78.26
3	33	515.625	18	281.74	15	153	2390.625	20	313.04
4	43	671.875	1	15.65	16	163	2546.875	12	187.82
5	53	828.125	2	31.30	17	173	2703.125	15	234.78
6	63	984.375	16	250.43	18	183	2859.375	19	297.39
7	73	1140.625	10	156.52	19	193	3015.625	8	125.22
8	83	1296.875	4	62.61	20	203	3171.875	17	266.08
9	93	1453.125	6	93.91	21	213	3328.125	22	344.35
10	103	1609.375	3	46.96	22	223	3484.375	11	172.17
11	113	1765.625	9	140.87	23	233	3640.625	21	328.70
12	123	1921.875	0	0.00	—	—	—	—	—

Figure 4-39: Transmission Impairment Measurement Tab Control, MultiTone Measurements Tab

4.4.2.1   Intermodulation Distortion

Intermodulation Distortion (IMD) occurs when non-linearity in the system generates harmonics from the signal that are characterized as noise frequencies in the signal. Because the 23-Tone signal has preset frequency and phase spacing's, the 2nd and 3rd harmonics of the system will always occur at predictable frequencies. 2nd and 3rd IMDs are calculated by comparing the sum of the power in the harmonics to the sum of the power of the 23-Tone signal following the process shown in **Figure 4-40**.
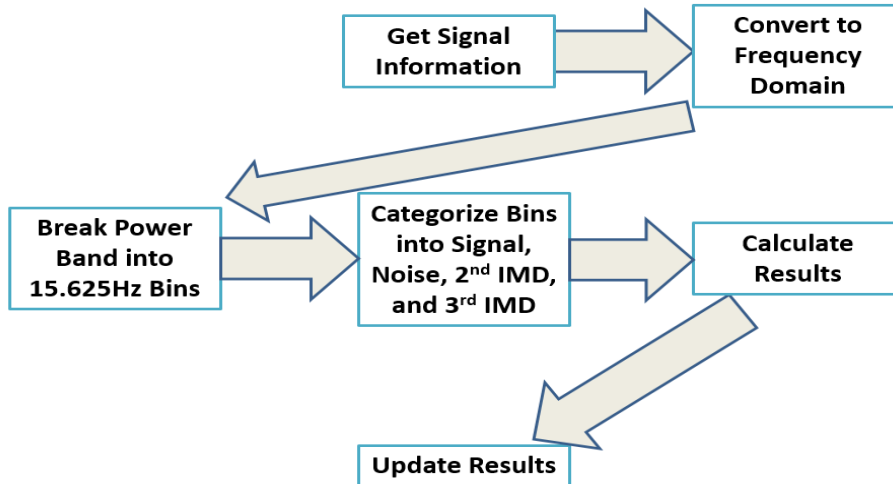
Figure 4-40: Intermodulation Distortion Test Flow Chart.

4.4.2.2 Envelope Delay Distortion

Envelope Delay Distortion is caused when frequency components of a signal are delayed by different amounts. Because the 23 tone signal initially has set phase differences between tones the relative delay between tones can be measure by comparing the phase difference to the initial phase difference of the 23 tone signal according to **Equation 4-13**. The delay of adjacent tones is averaged and the tone with the lowest delay is used as a reference following the process shown in **Figure 4-41**. For most tests, only the delays of the middle 22 pairs are reported with the final envelope delay distortion result being the difference between the smallest and largest delay.
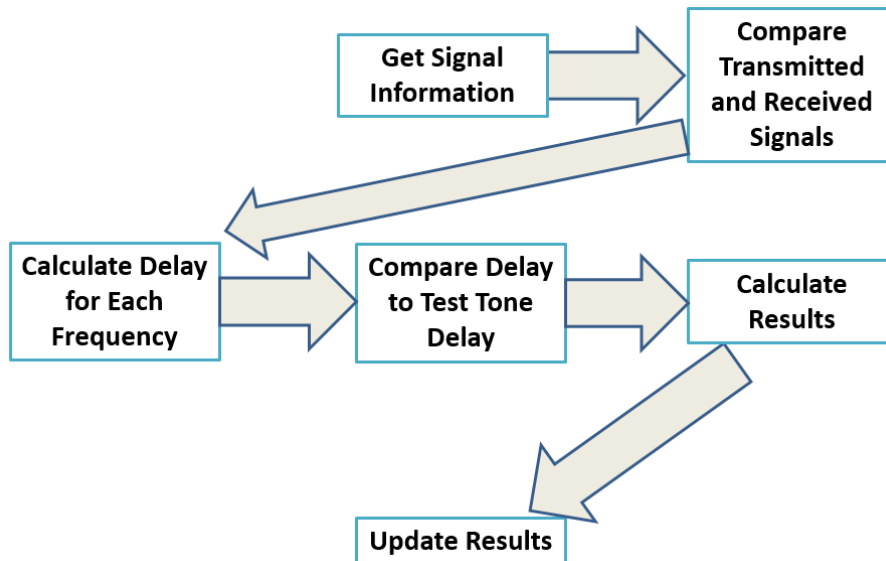
Figure 4-41: Envelope Delay Distortion Flow Chart

$$D = 10^6\left(\frac{\mathrm{acos}\left(\cos\left(\frac{\pi\left(\left(p_{1m}-p_{2m}\right)-\left(p_{1i}-p_{2i}\right)\right)}{180}\right)\right)}{2\pi*156.25}\right) \qquad (4\text{-}13)$$

Where D is the delay in microseconds, the p_ms are the measured phases of tone pair, the p_is are the initial phases of the tone pair, and 156.25 is the frequency difference between the tone pair.

4.4.3   Series Tone Tests

The Series Tone Test tab allows users to configure, start, cancel and display the results of a series tone test, **Figure 4-42**. Series Tone tests are tests that are performed using a sequence of single tone signals. The virtual test set is capable of automatically cycling through any series the user sets. This is done using a queue dedicated to requesting new tones from the function generator. On the receiving end, the virtual test set is capable of receiving the tones in any order. When a new tone is received the tone power is compared to the recorded power of the tone and the old tone is overwritten if the new tone is greater, **Figure 4-43**. Once each tone in the series has been detected the virtual test set will cycle through the series one more time, this ensures far end test equipment that may have had a delayed start will receive all of the signals in the series. While capable of performing attenuation slope measurements with any series, the virtual test set primarily uses the series tone test for 3 Tone Slope, Attenuation Distortion, and Echo Return Loss measurements.
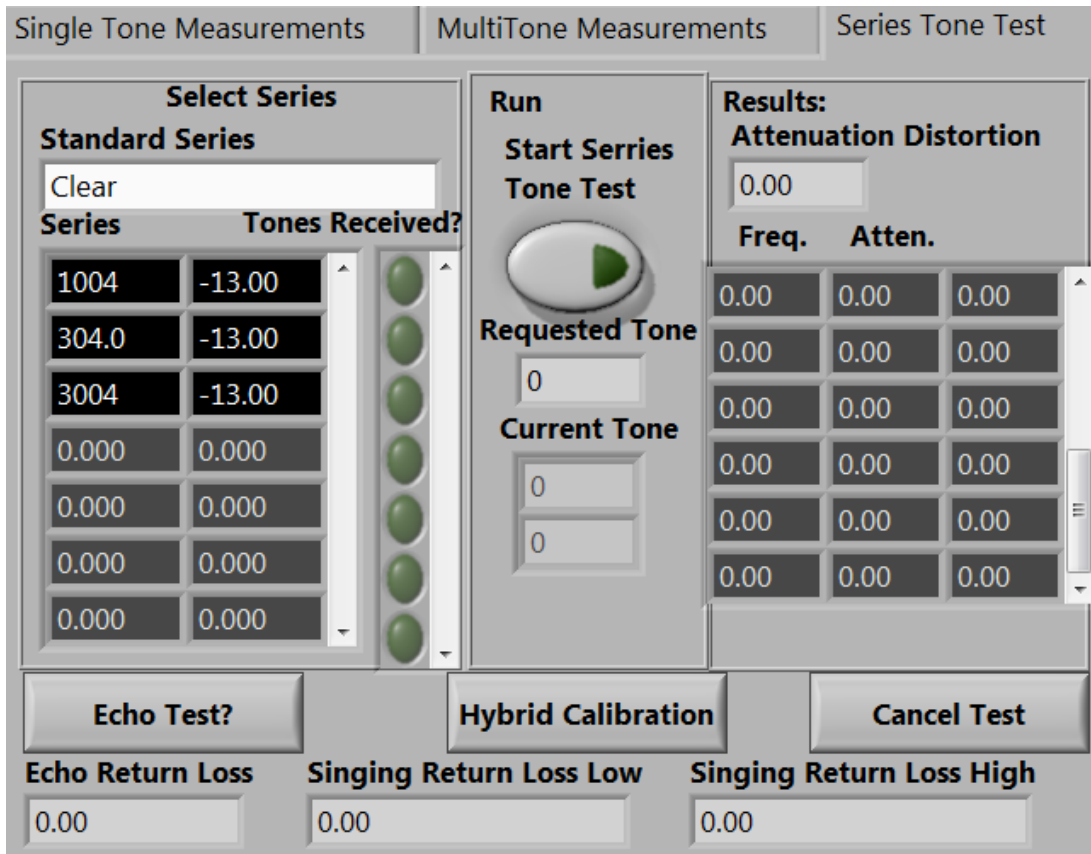
Figure 4-42: Transmission Impairment Measurement Tab Control, Series Tone Test Tab
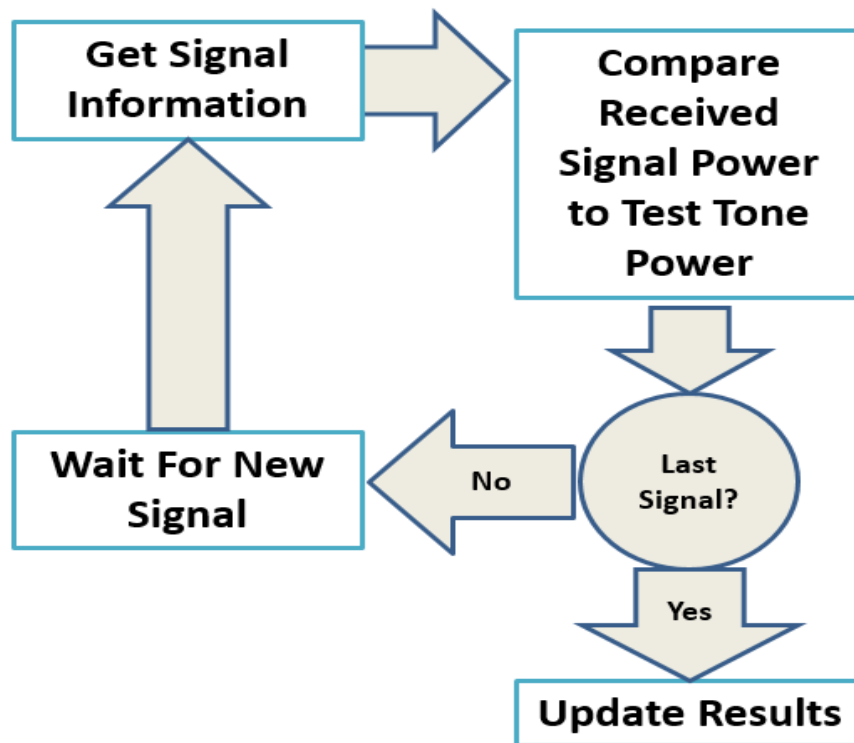
Figure 4-43: Series Tone Test Flow Chart

4.4.3.1   3 Tone Slope

The 3 Tone Slope is a series tone measurement that uses three standard frequencies. 304, 1004, and 3004Hz representing the low end, middle, and high end of the audio range spectrum. When the virtual test set performs the 3 Tone Slope test, or any other attenuation slope measurement, the first tone in the series is used as the reference tone whose attenuation is read as zero. For the 3 Tone Slope test the 1004Hz tone is used as a reference to which the attenuation of the other tones are compared.

4.4.3.2   Attenuation Distortion

Attenuation Distortion is a series tone measurement that uses 30 tones in 100Hz increments from 304Hz to 3004Hz, covering the full audio range spectrum. When the virtual test set performs the Attenuation Distortion test, or any other attenuation slope measurement, the first tone in the series is used as the reference tone whose attenuation is read as zero. For Attenuation Distortion the 1004Hz tone is used as a reference to which the attenuation of the other tones are compared. The relative and absolute attenuation for each tone is stored in a table and the final attenuation distortion result is calculated from the difference between the maximum and minimum attenuated tones.

4.4.3.3 Echo Return Loss

Echo Return Loss is a series tone measurement that measures how much of a signal is reflected back to the virtual test set. Echo Loss is usually caused by signal reflection due to impedance mismatching in a system and can cause unwanted signal loss and ring-back in a system. In order to separate the reflected signal from the transmitted signal it is necessary to use the virtual test set's internal hybrid circuit, or an external one, to enable the virtual test set to transmit and receive over the same two wires. The virtual test set performs echo return loss measurements by transmitting a series of tones covering the range of signals specified by the IEEE Echo Return Loss, Singing Return Loss Low, and Singing Return Loss High filters **[1]**. When the series tone test is indicated to be an Echo Return Loss test, the ideal series is passed through the filters in the 'Start Series Tone Test' event case and the results are saved. As the virtual test set receives tones from the series it passes them through each of the filters and saves the results alongside the attenuation slope measurements. Once the entire series is found the ideal results are compared to the actual results and converted to a dB result associated with each filter according to **Equation 4-14**. Additionally, the virtual test set is able to save the results when performing the multi-part IEEE hybrid circuit validation test.

$$L = 10 \log \left(\frac{\sum V_i^2}{\sum V_m^2} \right) - C \tag{4-14}$$

Where L is the echo loss in dB, V_i is the ideal peak amplitude of a tone passed through the appropriate filter, V_m is the measured peak amplitude after the a received tone has passed through the appropriate filter, and C is a calibration constant for the hybrid circuit.

4.4.4 Auto Test

The Auto Test tab allows a series of tests and/or a report to be generated from the results, **Figure 4-44**. Tests are broken into Single Tone, Multi-Tone, and Series Tone tabs and can be selected individually by pressing the corresponding button. In addition, pressing the report button will configure a report to be generated after all testing has completed. Certain tests and the report have additional options that can be changed for the tests. For example, the duration of an impulse noise test and the report format can be selected. Furthermore, a full testing regiment and corresponding test values can be configured simply by selecting a testing regiment from the Standard Testing Regiment menu. The virtual test set has two standard testing regiments built in: the Interim Technical Performance Record and the Key Performance Parameters. Once the user has selected all the tests they wish to perform they can add the tests to the testing queue to by clicking 'Add Tests,' testing will begin automatically. The tests remaining in the auto test queue will be displayed in the 'Test In Queue' indicator while the current test being performed is displayed in the 'Current Test' indicator. Additional tests can be selected and queued at any time, however reports will only contain the most recent results for each test. Finally, any tests may be canceled, paused, or resumed at any time.
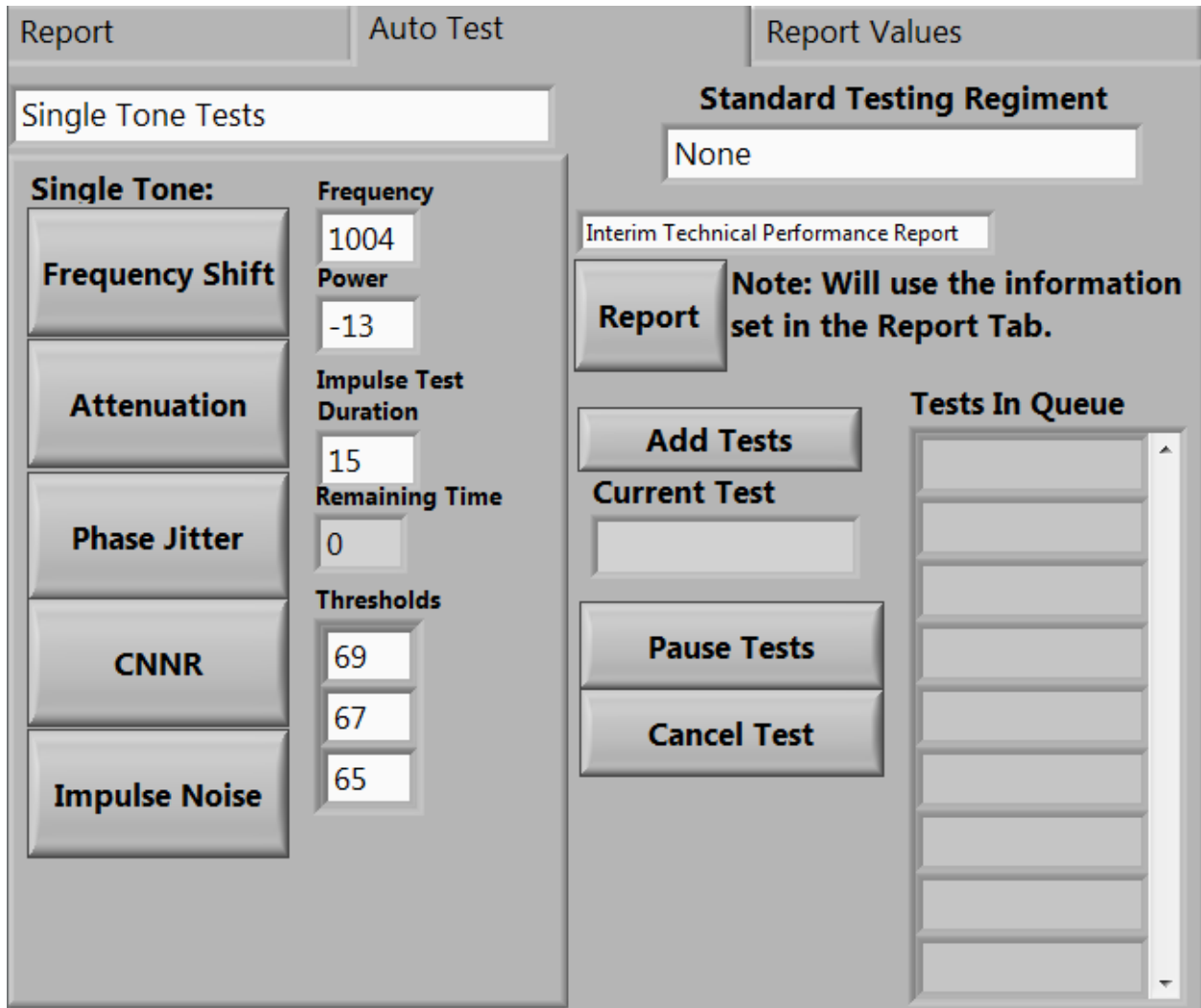
Figure 4-44: Transmission Impairment Measurement Tab Control, Auto Test Tab

4.4.5 Report Values

The Report Values tab is where tests results for a report are saved, **Figure 4-45**. During automatic testing information will automatically be entered into this tab after a test completes. During manual testing the user must click the latch buttons to copy the most recent test results into the corresponding fields. Each test has its own latch button. In addition, each of the single tone and multi-tone tests, which share signals between tests, can latch multiple results with a single button.
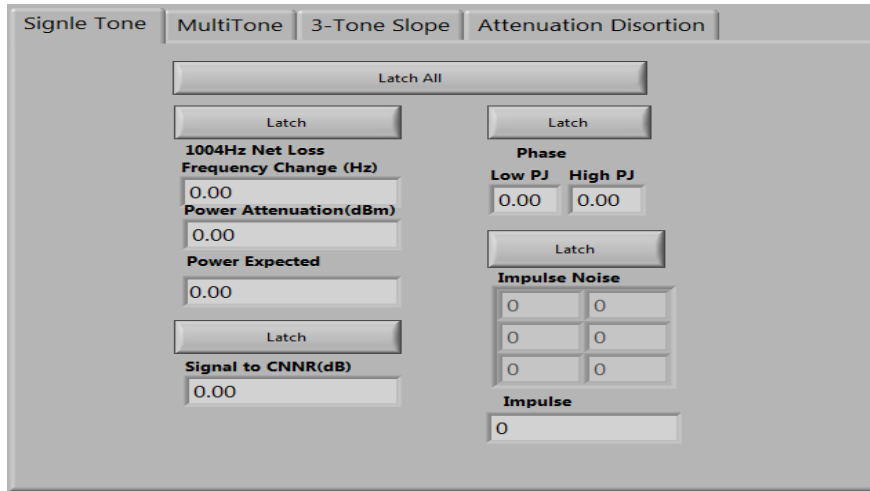
Figure 4-45: Transmission Impairment Measurement Tab Control, Report Values Tab

4.4.6 Report Generation

The Report tab is where non-numeric test information is entered and stored for a report and is also where the report can be generated, **Figure 4-46**. The virtual test set has two major analog impairment testing formats built in: the Interim Technical Performance Report (ITPR), **Figure 4-47**, and the Key Performance Parameters Report, **Figure 4-48**. Both of these tests require information about the test circuit in addition to the test results. This information includes: the test circuit name, the facility the test circuit is located, the location of the test circuit, the facility the test signal is transmitted from, the facility that is receiving the test signal, whether the test is in end to end or looped mode, and whether the circuit handles voice or voice and data. A report is generated when either the report case is queued in the automatic test queue or the user clicks 'Generate Report.' When a report is generated all of the most recent measurements corresponding to the report format are passed to the report along with the circuit information on the Report tab. In addition, each of the measurements are checked to see if they comply with the standards of the report, **Table 4-5** for the ITPR and **Table 4-6** for the key performance parameters report, and any values outside of the standard are highlighted in red. Once the report has been completed a word document is opened for user review.
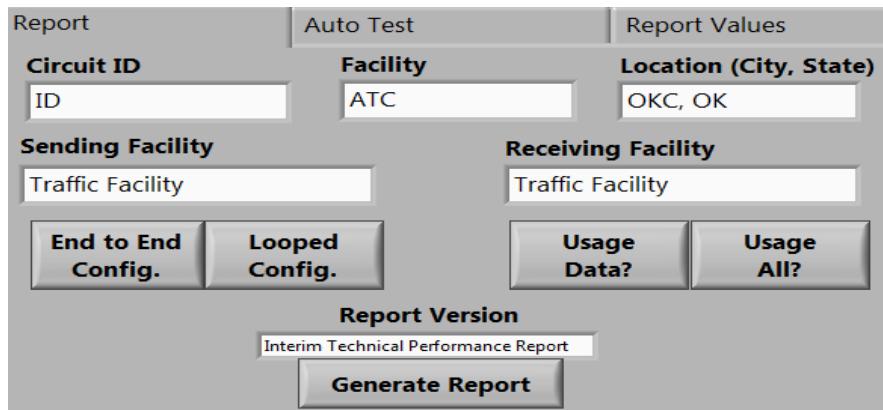
Figure 4-46: Transmission Impairment Measurement Tab Control, Report Tab

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INTERIM TECHNICAL PERFORMANCE RECORD — Fiber Optic Transmission System (FOTS) Analog VF Services

FACILITY	DATES		SUPERVISOR'S SIGNATURE
	FROM	To	

LOCATION

		Circuit ID			Sending Facility		Receiving Facility		

		Configuration		Usage		Line Test Set-Up (1004 Hz)			
		End-to-End	Looped	All		Send Point		Receive Point	
Date	Time					TLP	dBm	TLP	dBm

	1004 Hz Net Loss/ Frequency Shift		Attenuation Distortion 304 to 3004 Hz		3 Tone Slope Attenuation Distortion Or Attenuation Distortion 404 to 2804 Hz		Signal-to-C-notched Noise Ratio)	Remarks	Initials
			304 Hz – 3004 Hz		404 Hz – 2804 Hz				
	Freq. Shift	Level (dB)	Minimum Level (dB)	Maximum Level(dB)	Minimum Level (dB)	Maximum Level (dB)	Level (dB)		
Nominal	0 Hz	0.0 dB	1004 Hz Net Loss	1004 Hz Net Loss	1004 Hz Net Loss	1004 Hz Net Loss	≥ 34 dB		
Minimum	-1Hz	-1.5 dB		-1.0 dB		- 1.0 dB	> 32 dB		
Maximum	+1Hz	+1.5 dB	+5.0 dB		+2.0 dB				

Receive Level

Freq (Hz)	304	404	504	604	704	804	904	1004	1104	1204	1304	1404	1504	1604	Remarks	
Attenuation dBm																
Attenuation dBm																
Attenuation dBm																
Freq (Hz)	1704	1804	1904	2004	2104	2204	2304	2404	2504	2604	2704	2804	2904	3004	Remarks	
Attenuation dBm																
Attenuation dBm																
Attenuation dBm																

Interim Fiber Optic Transmission System (FOTS) Analog VF Services TPR Form (1 of 2)

INTERIM TECHNICAL PERFORMANCE RECORD — Fiber Optic Transmission System (FOTS) Analog Data Services

FACILITY	DATES		SUPERVISOR'S SIGNATURE
	FROM	TO	

LOCATION

		Circuit ID			Sending Facility		Receiving Facility		

		Configuration		Usage		Line Test Set-Up (1004 Hz)			
		End-to-End	Looped	Data		Send Point		Receive Point	
Date	Time					TLP	dBm	TLP	dBm

	Impulse Noise	Envelope delay distortion	Phase Jitter		Intermodulation Distortion		Remarks	Initials
	At Threshold Noted (dBrnc0)	804 Hz – 2804 Hz	4–300 Hz	20–300 Hz	Second Order	Third Order		
Nominal	≤ 15 counts in 15 Min. at 65 dBrnC0	≤650 µsec	≤8 °	≤3 °	≥ 46 db	≥ 49 db		
Minimum					> 45 db	> 48 db		
Maximum	≤ 15 counts in 15 Min. at 67 dBrnC0	≤ 700 µsec	<9 °	<4 °				

Envelope Delay Distortion

Freq (Hz)	750	906	1063	1219	1375	1531	1688	1845	2000	2156	2313	2469	2625	2781	Remarks
Delay (µ sec)															
Delay (µ sec)															
Delay (µ sec)															

Interim Fiber Optic Transmission System (FOTS) Analog Data Services TPR Form (2 of 2)

Figure 4-47: Interim Technical Performance Record Report Sheet.

Table 4-5: Fiber Optic Transmission System Transmission Line Standards

Measurement	Standard
Frequency Shift	+/- 1Hz
1004Hz Net Loss	+/- 1.5dB
C-Notched Noise Ratio	> 32dB
Low Frequency Phase Jitter	< 9º Pk-Pk
High Frequency Phase Jitter	< 4º Pk-Pk
Impulse Noise	< 15 at 67dBrnc0
3 Tone Slope	-1dB < Attenuation Difference > 2dB
Attenuation Distortion	-1dB < Attenuation Difference > 5dB
Envelope Delay Distortion	Max-Min </= 700uSec
2nd Order Intermodulation Distortion	> 45dB
3rd Order Intermodulation Distortion	> 48dB

Figure 4-48: Key Performance Parameters Report Sheet.

Table 4-6: Key Transmission Line Parameter Standards

Measurement	Standard
1004Hz Net Loss	+/- 1.5dB
C-Notched Noise Ratio	> 32dB
3 Tone Slope	-1dB < Attenuation Difference > 2dB

4.5 Oscilloscope

The Oscilloscope function is meant to look and function like a Tektronix oscilloscope, **Figure 4-49**. Each of the buttons and knobs mimic the mechanical actions and display controls of the standalone Oscilloscope. The virtual test set's oscilloscope is also capable of performing various periodic measurements and acting as a basic spectrum analyzer. Like a standalone oscilloscope, the front panel of the Oscilloscope VI consists of the main display, where the captured signal and measurements are displayed, and the controls section, where the capture and display settings are changed. The oscilloscope will automatically display new signals as they are received and updates the display settings as they are changed, **Figure 4-50**.
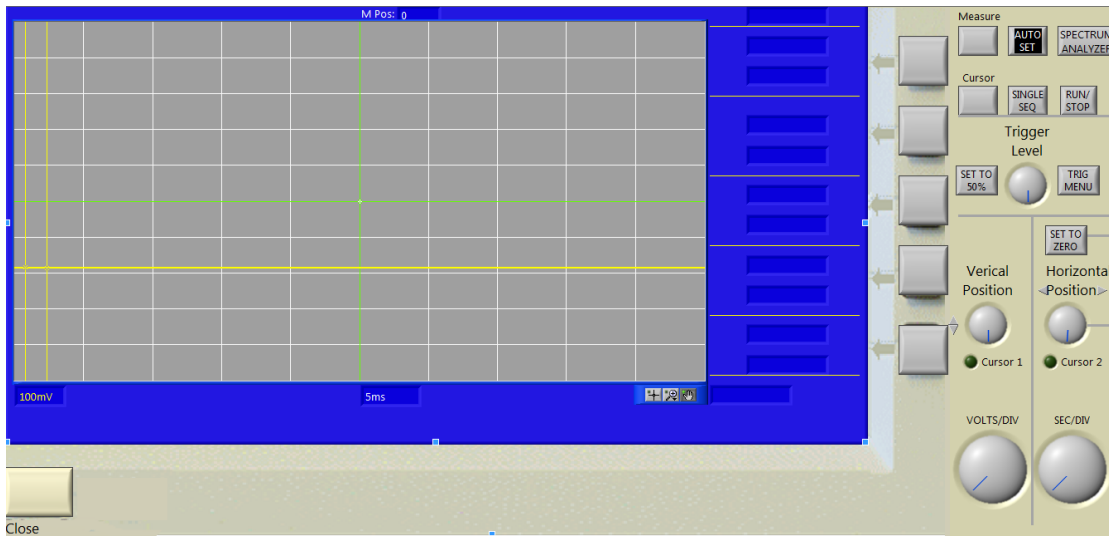
Figure 4-49: Oscilloscope Front Panel
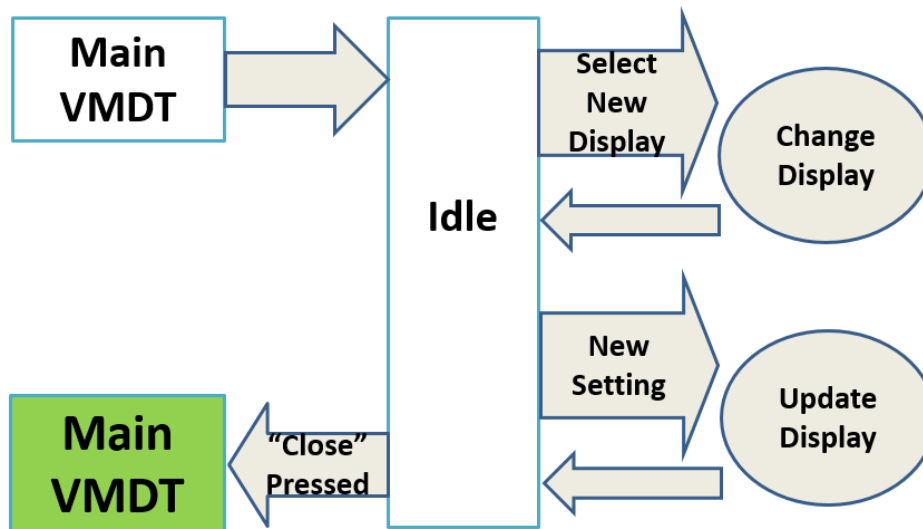
Figure 4-50: Oscilloscope/Spectrum Analyzer Flow Chart

4.6 Selective Signaling

The Selective Signaling function is meant to transmit, receive, and decode SS-1 signals. Selective Signaling is used on party-lines, phone lines where audio is transmitted to all users at once, to notify a specific user of a call. Selective Signaling signals consist of 2 or 3 digit signals represented as a series of high and low frequency pulses **[4]**. The front panel consists of the signal settings, the dial pad, the pulse durations display, and the graphs display, **Figure 4-51**. The signal settings section allows the user to configure the signal parameters of the signal, the number of digits in a code, whether or not 1 is a valid digit, and whether the audio signal will be played when it is transmitted. The accepted codes will change based on the settings and SS-1 codes are automatically sent as the user enters the digits, **Figure 4-52**.
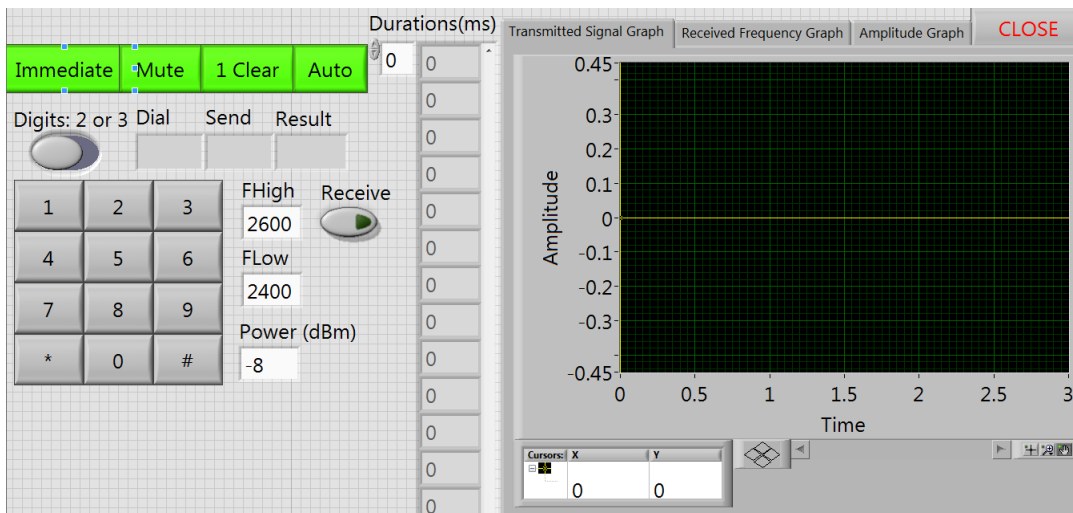
Figure 4-51: Selective Signaling Front Panel
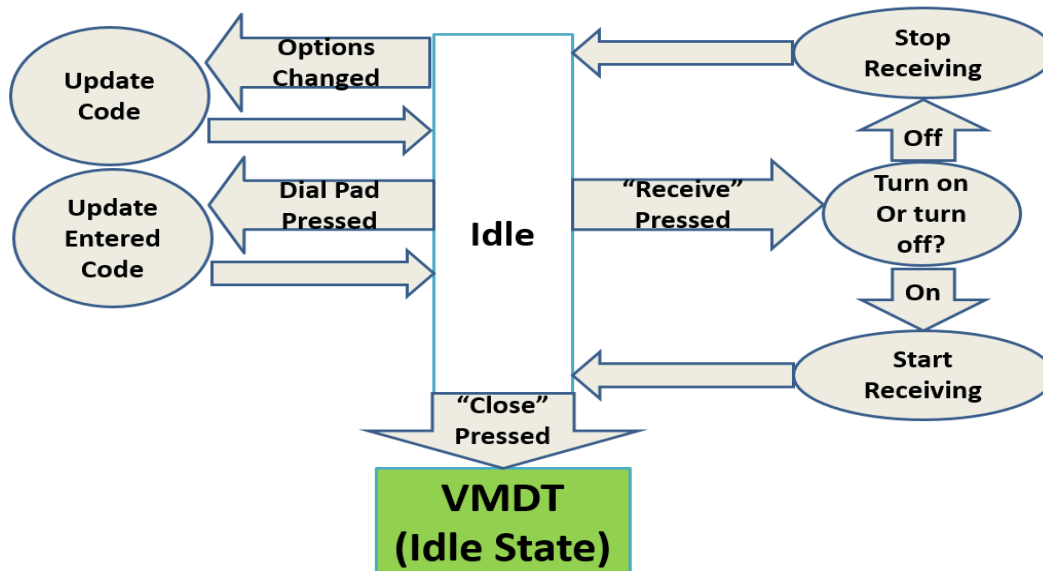
Figure 4-52: Selective Signaling User Flow

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4.7 Frequency Shift Keying

The Frequency Shift Keying (FSK) function is meant to transmit, receive, and decode Bell 202 FSK signals. FSK is used to transmit digital data over analog systems. FSK signals consist of digital data represented as a series of high and low frequency pulses, known as mark and space pulses respectively. **[5]**. The front panel consists of the signal settings, text entry and display, and the captured signal display sections, **Figure 4-53**. The signal settings section allows the user to configure the signal parameters of the signal. The text entry and display section allows the user to transmit a text message and displays any text message that is received. The captured signal display section consists of tabs displaying the captured signal at each stage in the decoding process ending in a binary display of the captured bits. The accepted codes will change based on the settings and FSK codes are sent and received upon user command, **Figure 4-54**.
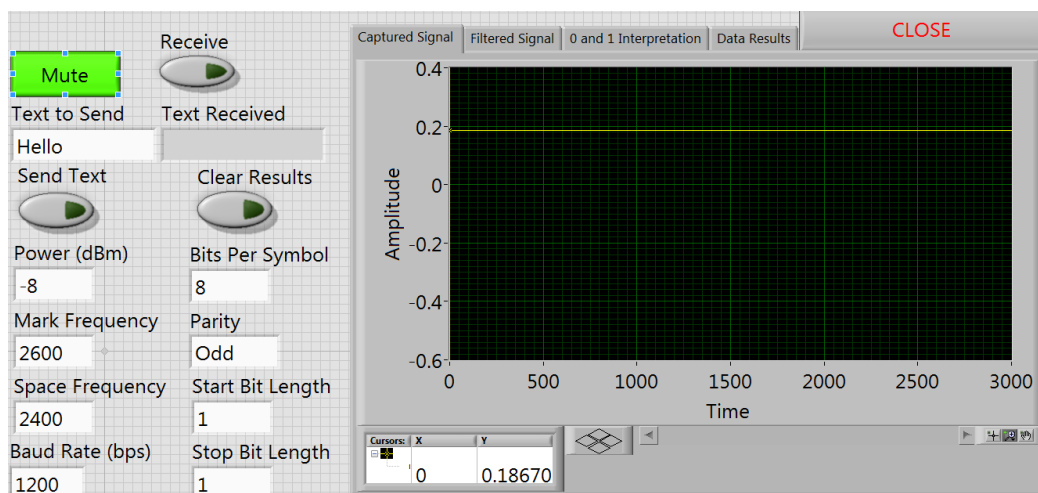
Figure 4-53: Frequency Shift Keying Front Panel
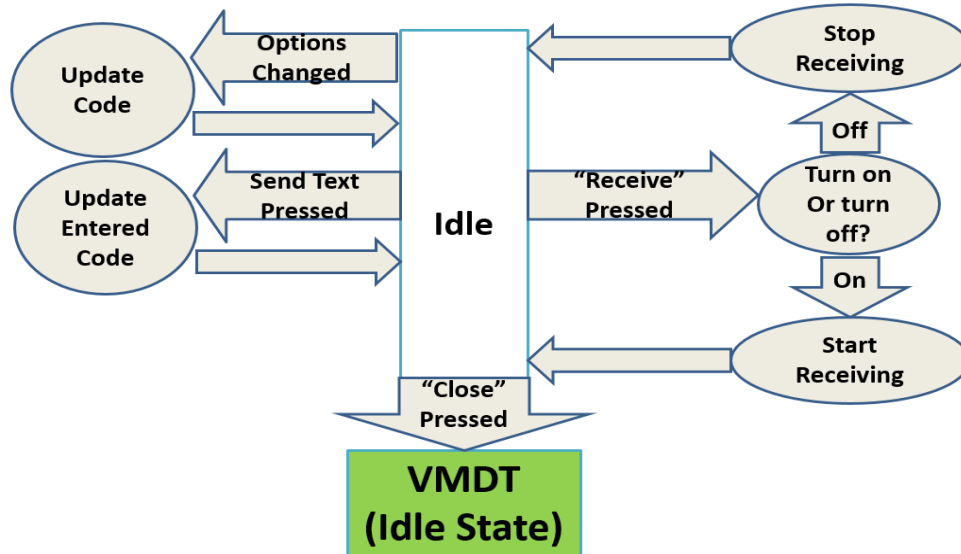
Figure 4-54: Frequency Shift Keying User Flow Chart

4.8 Digital Multi-Meter

The Digital Multi-Meter function is meant to look and function like a standalone Digital Multi Meter, **Figure 4-55**. Like a standalone Digital Multi-Meter, the center knob is used to select the measurements including voltage, current, and resistance measurements. In addition, a drop down menu selects the display range while a simple illustration demonstrates the physical connection need for the chosen measurement. Measurements are continuously made when the user is not changing the measurement settings, **Figure 4-56**.

Figure 4-55: Digital Multi-Meter Front Panel
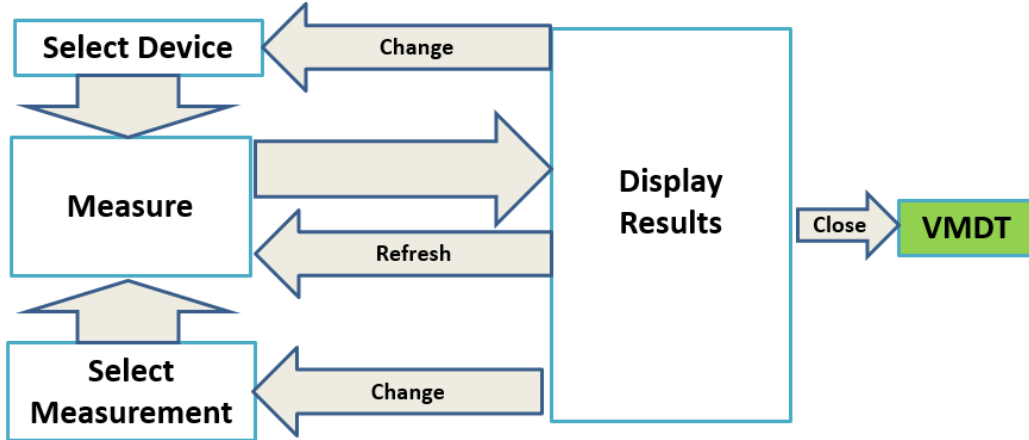
Figure 4-56: DMM Flow Chart

4.9 Calibration

The Calibration function is designed to keep the virtual test set measurements within IEEE specifications by accounting for any drift in the hardware. The virtual test set requires an external reference signal to calibrate its input but is capable of calibrating its own output except in 2-wire mode, where a second virtual test set is required, **Table 4-7**. At least one initial calibration is required to account for the amplification, impedance matching, and 2-to-4 wire converter circuits. The front panel consists of the calibration selection, instruction display, and the calibration graphs display, **Figure 4-57**. The calibration selection section allows the user to select the calibration settings used to perform a new calibration of view a previous calibration. Measured amplitude graph displays a continuous slope of values used for the calibration, from the maximum to the minimum. The difference from ideal graph displays the difference between the measured amplitude graph and the values an ideal signal would produce. The calibration curve graph displays the values transmitted/received by the analog output/input pins vs the values transmitted to/measured from the transmission line. Finally, the difference curve displays the difference between the values displayed on the calibration curve graph vs the ideal values. More details about calibrating the virtual test set are provided in **Appendix A**.

Table 4-7: Calibration Settings

Calibration Settings			Signal	External Requirements
Type	Mode	Impedances (Ohms)	Max/2-Volt	External Requirements
Input	Bridge	NA	8V/2V, 0.5Hz Triangle Wave	Calibrated Signal Source
Input	Standard 4-Wire	600	8V/2V, 0.5Hz Triangle Wave	Calibrated Signal Source
Input	2-Wire	600, 900, 1200	8V/2V, 1004Hz Sine Wave	Calibrated Signal Source, Impedance Matching
Output	Bypass	NA	8.4V/2V, 0.5Hz Triangle Wave	Calibrated Input

Output	Standard 4-Wire	100, 125, 600, 900, 1200	8.4,5.8,4.3V/2V 0.5Hz Triangle Wave	Calibrated Input
Output	2-Wire	600, 900, 1200	8.4V/2V, 1004Hz Sine Wave	2nd Input Calibrated VIRTUAL TEST SET
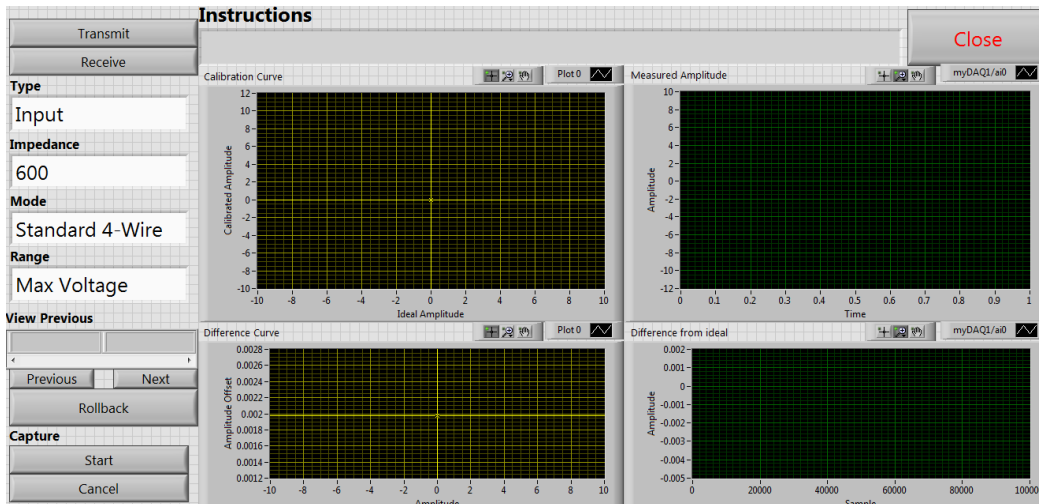
Figure 4-57: Calibration Front Panel

5   Testing

Before any new test set is accepted by the industry it must meet established performance standards. The test equipment requirements for specific systems are detailed in the maintenance handbook for the system while more general standards and guidelines are provided by IEEE standards. IEEE Standard 743-1995 details several audio range analog tests to verify the quality of signals being transmitted on an analog communication channel. In order to verify that the virtual test set meets these guidelines it is necessary to compare the performance of the virtual test set with other transmission test sets as well as perform validation tests specified by the IEEE standards. This section will go into detail about the tests that were performed and what the tests demonstrate.

5.1   Single Tone Generation and Measurement

The most common tests performed on analog systems use single tone measurement tests. A single test can determine if the circuit is connected, how much of the signal power is lost through the system, and what the signal to noise ratio is. For this reason, the accuracy requirements for generating and receiving single tones signals are the strictest of the IEEE analog measurement requirements [1]. For voice band measurements the requirements surrounding the 1002 Hz to 1022 Hz range are particularly strict because it is the range of frequencies the human ear is most sensitive to. Tones within this range of frequencies that meet specific noise requirements are known as holding tones and are essential to many different audio range analog tests.

Single tone generators that are to produce a holding tone must produce the tone within 0.1 dB (~2%) of the intended power level, the frequency must not drift by more than 1 Hz, the signal must contain almost no jitter, and must have a signal to distortion ratio of at least 50 dB (I.E the signal power is at least 100000 times stronger than the noise and harmonic signal powers). Single tone generators that produces tones other than the holding tone will have fewer noise requirements for tones outside of the 1002-1022 Hz and -19-0 dBm ranges. Tones that are not to be used as holding tones only need to be within 0.2 dB of their intended power, within 0.1% of the intended frequency, and have a signal to distortion ratio of at least 40 dB. For audio range test equipment the signal generator must at least be able to produce tones from 300-3900 Hz and -40-10 dBm while maintaining all of the accuracy requirements described above.

The test set also must meet accuracy requirements for measuring single tone signals and must meet these requirements under a wide range of noise levels. For holding tone signals the test set must be accurate to within 0.1 dB of accuracy and within 0.2 dB for other tones between 200 and 15k Hz. For frequencies outside of the audio range, the test set will be allowed to measure tones at any frequency so long as it maintains an accuracy of 0.5 dB. In addition to the accuracy requirements, the test set must accept tones with at least 7.4 Volts of amplitude and be able to measure levels as low as -49.9 dBm for any impedance setting.

Figure 5-1: Halcyon TIMS [TesComUSA.com]

To verify the virtual test set is capable of meeting the accuracy requirements specified by IEEE the virtual test set was looped to itself to measure its own output and the result was compared to the frequency and power of the Halcyon test set, **Figure 5-1 [6]**. Five different measurement sets were taken. The first test was meant to verify the output level accuracy of the virtual test set and consisted of

a single 1004 Hz tone at varying power levels and impedance settings, only the 600 and 900 ohm impedances were tested because those are the most common impedance settings for audio range testing. The second test was to verify the signal to noise level of tones generated by the virtual test set and also consisted a 1004 Hz tone at varying levels. The third test was to verify the virtual test set's frequency output accuracy by keeping the power level constant while varying the frequency of the virtual test set. The fourth test verified the flatness of the virtual test set's output by measuring the power level as the output power level was kept constant across a wide range of frequencies. The final single tone level test checked the level generation and measurement accuracies by varying the virtual test set's output power and frequency over the full range specified by IEEE and comparing the results measured by the Halcyon. The measurements made during each of these tests are available in **Appendix C** while the results are given in **Tables 5-1, 5-2, 5-3, and 5-4**.

Table 5-1: Single Tone Generator

Test	Virtual Test Set	Standard
Output Level Range	At least 12.1-(-58) dBm	10-(-40) dBm
Output Level Accuracy	+/- 0.02 dBm	+/- 0.1 dBm 0-(-19), +/- 0.2 else
Frequency Stability	No change measured.	<0.1% deviation per hour
Total Distortion	40-60 dB below tone level	>40 dB below tone level
Output Frequency Accuracy	At least +/- 0.01, 200-40k Hz	+/- 1 Hz 50-9999, +/- 10 Hz 10k+
Frequency Adjustment Increments	As low as 0.02 Hz for any frequency range.	<4 Hz 4k-, <10 Hz 4k-10k, <100 Hz 10K+
Frequency Range	0-50k Hz	At least 50-3900 Hz
Flatness	Varies by 0.2 200-10k	Vary by 0.2 dBm, 200-15k Hz

The virtual test set failed the flatness requirement of a single tone generator, only being able to maintain 0.2 dB flatness up to 10 kHz. This is caused by the virtual test set output dropping in power at higher frequencies. This can be fixed by adding a frequency response component to the calibration process. However, 4000 Hz is the highest required frequency for voice range tests so the virtual test set still meets the single tone generation requirements for voice range testing.

Table 5-2: Holding Tone Generator

Test	Virtual Test Set	Standard
Phase Jitter	0.03 Degrees	<0.2 Degrees
Amplitude Jitter	0.05 %	<0.2 %
Impulse Noise	None Detected	>20 dB below tone
Phase Hits	None Detected	<3 Degrees
Gain Hits	None Detected	<0.5 dB
Total Distortion	60 dB	>50 dB

Table 5-3 Tone Level Measurement

Value	Virtual Test Set	Standard
100 Ohm Range	27.0-(-62) dBm	24.4-(-49.9) dBm
135 Ohm Range	25.7-(-63) dBm	23.1-(-49.9) dBm

600 Ohm Range	19.2-(-70) dBm	16.6-(-49.9) dBm
900 Ohm Range	17.4-(-72) dBm	14.9-(-49.9) dBm
Accuracy 20-200 Hz	+/- 0.01 dB	+/- 0.5 dB
Accuracy 200-15K Hz	+/- 0.2 dB	+/- 0.2 dB
Accuracy >15k Hz	+/- 0.4 dB	+/- 0.5 dB
Holding Tone 0-(-19) dBm	+/- 0.02 dB	+/- 0.1 dB

Table 5-4 Frequency Measurement and Accuracy

Test	Virtual Test Set	Standard
Accuracy	+/- 0.01 Hz	+/- 1 Hz <10k, +/- 10 Hz >10k
Level Range	At least -62-17.4 dBm	-50-10 dBm
Measurement Range	0-50k Hz	At least 50-4k Hz

5.2   Noise Measurement

Noise is any unwanted signal that can be picked up by the receiving equipment. Noise can be random or deterministic but every signal will contain some level of noise. At sufficiently low levels noise is negligible compared to the transmitted signals but at high levels noise can cause sever irritation to listeners of audio range signals as well as cause errors on digital systems. High noise levels can also cause errors in other measurements so many measurements provide noise level indications in addition to the measurement results.

Noise signals are typically the sum of a wide frequency range of low power signals. As a result noise measurements made across a wider frequency range will measure higher noise level than measurements made across a narrower frequency range. For this reason, selecting the proper filter is crucial for measuring the correct noise level. The IEEE standard provides specifications for C, D, 3.4k Hz Flat, 15k Hz Flat, E, F, and G filters however the virtual test set does not use the E, F, and G filters because those filters are used for data systems instead of voice systems **[1]**. Furthermore, the C, D, and 3.4k Hz Flat filters are often used for measurements that use holding tones and noise measurements must be made in conjunction with a notch filter to eliminate the holding tone. For the purposes of noise measurement IEEE standards will permit any notch filter that attenuates the 995-1025 Hz frequency range by more than 50 dB and attenuates the range outside of 862-1182 Hz by less than 3 dB but more than -0.5 dB. Regardless of the filter configuration used, IEEE standards mandates that audio range noise level measurements accept a noise range of at least 10-90 dBrn and be accurate to within 1 dB.

Often more important than a noise level measurement itself is a Signal to Noise, or Signal to Total Distortion, Ratio test. A signal to noise ratio test measures the ratio of the signal and noise signal powers where noise includes random noise, harmonics of the signal, and any other unwanted signals. Per IEEE standards, the test set must be able to measure signal to noise ratios between 10-45 dB and be accurate to within 1 dB for ratios below 40 dB and 2 dB for ratios above 40 dB.

To verify the virtual test set is capable of meeting the accuracy requirements specified by IEEE the virtual test set was looped to itself to measure its own output, which includes simulated impairments,

and the result was compared to the noise level and signal to noise ratio measurements of the Halcyon test set. Four sets of tests were performed to determine whether the virtual test set meets IEEE noise measurement standards. For the first test a wide band noise signal was applied to the virtual test set and Halcyon at different power levels to demonstrate the noise level measurement range and accuracy of the virtual test set under the various filter settings. The second test used a 1.8 kHz signal with gated attenuation of different gating frequencies to verify the noise averaging time of the noise level measurement test. The third test had the leads switched while a 75% duty cycle, 300 Hz square wave was applied to verify that the polarity of the leads has no effect on the final noise level measurement. For the final set of tests, a 1004 Hz tone and wide range noise of different power ratios was applied to the virtual test set and Halcyon with C-Notched filter settings to verify the signal to noise measurement range and accuracy of the virtual test set. The measurements made during each of these tests without results specified by IEEE are available in **Appendix C** while the results are given in **Tables 5-5, 5-6, and 5-7**.

Table 5-5 Noise Measurement Range and Accuracy

Filter	Virtual Test Set	Standard
C-Message	8-112.2 dBrn	10-90 dBrn
C-Notched	8-112.2 dBrn	10-90 dBrn
3.4kHz Flat	8-112.2 dBrn	10-90 dBrn
3.4kHz Flat Notched	8-112.2 dBrn	10-90 dBrn
D-Filter	8-112.2 dBrn	10-90 dBrn
D-Notched	8-112.2 dBrn	10-90 dBrn
15kHz Flat	8-112.2 dBrn	10-90 dBrn
Accuracy	+/- 0.4 dB	+/- 1 dB

Table 5-6 Noise Measurement Test Signals

Signal	Virtual Test Set	Standard
1.8kHz, 50% Duty Cycle of 8.4 dB Gain Loss, 10 Hz Gate	85.8 dBrn +/- 0.2	<1 dB Variation
1.8kHz, 50% Duty Cycle of 8.4 dB Gain Loss, 2 Hz Gate	84 dBrn +/- 4	>3 dB Variation
300 Hz 20% Duty Square Wave into 3.4 kHz Flat Filter	No Change When Inverted	+/- 1 dB of inverted

Table 5-7 Signal to Noise Ratio

Test	Virtual Test Set	Standard
Measurement Range	At least 10-45 dB	10-45 dB
Accuracy (10-40 dB)	+/- 0.4 dB	+/- 1 dB
Accuracy (40-45 dB)	+/- 0.5 dB	+/- 2 dB

5.3 Jitter Measurement

Jitter is any unwanted oscillation of a signal. Jitter is often caused by modulation of the signal by low frequency noise signals and is represented as phase and amplitude oscillations. The two main measurements of jitter, Phase Jitter and Amplitude Jitter, are used to quantify the jitter effects on a signal by passing a holding tone through a system. The holding tone is then phase and amplitude

demodulated and the resulting signal is analyzed. Filters are used to single out the two frequency bands of interest, 4-300 Hz and 20-300 Hz. The filters used for phase and amplitude jitter measurements must be properly weighted to meet IEEE specifications **[1]**.

To verify the virtual test set is capable of meeting the accuracy requirements specified by IEEE the virtual test set was looped to itself to measure its own output, which includes simulated impairments, and the measured result was compared to the jitter measurements made by the Halcyon test set. Seven sets of tests were performed using to determine if the virtual test set's phase and amplitude jitter measurements meet IEEE specifications. The first set of tests used a 1 kHz signal as a holding tone and a 1.1 kHz signal as a source of jitter using an array of power levels for both tones to demonstrate the range and accuracy of the virtual test set's measurements. The second test uses a 1 kHz holding tone with voice band noise to verify that random noise will not be measured as jitter. The third test used a 1 kHz 0 dBm holding tone and a second tone of varying frequency near the holding tone and -20 dBm of power to verify the frequency weighting of the filters used for phase and amplitude jitter measurement. The fourth test used a 1 kHz 0 dBm holding tone and a second tone of varying low frequency and -20 dBm of power to verify that low frequency signals will not be measured as jitter. The fifth test used a 1 kHz signal as a holding tone and a 1.1 kHz signal as a source of jitter using an array of total signal powers while keeping the relative power of the signals to verify that the relative power of the signals will not affect the final jitter measurement. The sixth test used an amplitude modulated signal for phase jitter measurement to verify that amplitude jitter will not be measured as phase jitter. The seventh test used a phase modulated signal for amplitude jitter measurement to verify that phase jitter will not be measured as amplitude jitter. The eighth test used a gated phase modulating signal of varying gating frequencies to verify the averaging time of the phase modulation measurement. The ninth test used a gated amplitude modulating signal of varying gating frequencies to verify the averaging time of the amplitude modulation measurement. The tenth test used 1 kHz, 1.1 kHz, and 1.17 kHz tones as specified in the IEEE standard to verify the effect of two tone interference on phase and amplitude modulation. The measurements made during each of these tests without results specified by IEEE are available in **Appendix C** while the results are given in **Tables 5-8, 5-9, 5-10, and 5-11**.

Table 5-8 Phase Jitter Range and Accuracy

Test	Virtual Test Set	Standard
Accuracy	+/-0.2°	+/-5% +/-0.2°
Tone Level Range	At least -62-17.4 dBm	-40-10 dBm
Tone Frequency Range	At least 1002-1022 Hz	1002-1022 Hz
Measurement Range	At least 0-37° Pk-Pk	0-25° Pk-Pk

Table 5-9 Phase Jitter Test Signals

Signal	Virtual Test Set	Standard
250-3.5 kHz noise 30 dB below 1 kHz Sine Wave	3.3° Pk-Pk	<4° Pk-Pk
0 dBm 1 kHz Sine Plus -20 dBm 1000.4 or 999.6 Hz Sine	0.2° Pk-Pk LPJ	<1° Pk-Pk LPJ
0 dBm 1 kHz Sine Plus -20 dBm 1001 or 999 Hz Sine	0.2° Pk-Pk LPJ	<3° Pk-Pk LPJ
0 dBm 1 kHz Sine Plus -20 dBm 1002 or 998 Hz Sine	2.4° Pk-Pk LPJ	<8° Pk-Pk LPJ
0 dBm 1 kHz Sine Plus -20 dBm 1002 or 998 Hz Sine	0° Pk-Pk HPJ	<1° Pk-Pk HPJ

0 dBm 1 kHz Sine Plus -20 dBm 1004 or 996 Hz Sine	11.0° Pk-Pk LPJ	9.2-12.1° Pk-Pk LPJ
0 dBm 1 kHz Sine Plus -20 dBm 1005 or 995 Hz Sine	0.2° Pk-Pk HPJ	<3° Pk-Pk HPJ
0 dBm 1 kHz Sine Plus -20 dBm 1010 or 990 Hz Sine	2.2° Pk-Pk HPJ	<8° Pk-Pk HPJ
0 dBm 1 kHz Sine Plus -20 dBm 1008-1240 Hz Sine	11.0-11.5° Pk-Pk LPJ	10.8-12.2° Pk-Pk LPJ
0 dBm 1 kHz Sine Plus -20 dBm 760-992 Hz Sine	11.4-11.5° Pk-Pk LPJ	10.8-12.2° Pk-Pk LPJ
0 dBm 1 kHz Sine Plus -20 dBm 1020-1240 Hz Sine	11.0-11.5° Pk-Pk HPJ	10.8-12.2° Pk-Pk HPJ
0 dBm 1 kHz Sine Plus -20 dBm 760-980 Hz Sine	11.1-11.5° Pk-Pk HPJ	10.8-12.2° Pk-Pk HPJ
0 dBm 1 kHz Sine Plus -20 dBm 1300 or 700 Hz Sine	10.2-10.4° Pk-Pk	10-12.2° Pk-Pk
0 dBm 1 kHz Sine Plus -20 dBm 1500 or 500 Hz Sine	2.8° Pk-Pk	<3° Pk-Pk
0 dBm 1 kHz Sine Plus -20 dBm 0-300 Hz Sine	<0.2° Pk-Pk	<1° Pk-Pk
0 dBm 1 kHz Sine Plus -20 dBm 1.5-4 kHz Sine	<2.8° Pk-Pk	<3° Pk-Pk
0 dBm 1 kHz Sine Plus -20 dBm 4+ kHz Sine	<0.1° Pk-Pk	<1° Pk-Pk
1 kHz Sine Plus 20 dBm lower 1.1 kHz Sine -40-10 dBm Total Signal Power	11.4-12.0° Pk-Pk	<0.7° Pk-Pk difference
10% AM, 0 dBm 1 kHz Carrier, 2-900 Hz Message	0.05-0.98° Pk-Pk	<1° Pk-Pk
PM, 1 kHz Carrier, 50% Duty Message, 5 Hz Gate	$\frac{2(13.88-13.65)}{(13.88+13.65)}=2\%$	$\frac{2(max-min)}{(max+min)}<10\%$
PM, 1 kHz Carrier, 50% Duty Message, 1 Hz Gate	$\frac{2(13.59-0.05)}{(13.59+0.05)}=199\%$	$\frac{2(max-min)}{(max+min)}>40\%$
0 dBm 1 kHz Sine Plus 1.1 kHz Sine Plus 1170 kHz Sine	19.8° Pk-Pk	18.5-19.8° Pk-Pk

Table 5-10 Amplitude Jitter Range and Accuracy

Test	Virtual Test Set	Standard
Accuracy	+/-0.2% Peak	+/-5% +/-0.2% Peak
Level Range	At least -62-17.4 dBm	-40-10 dBm
Frequency Range	At least 1002-1022 Hz	1002-1022 Hz
Measurement Range	At least 0-33% Peak	0-25% Peak

Table 5-11 Amplitude Jitter Test Signals

Signal	Virtual Test Set	Standard
250-3.5 kHz noise 30 dB below 1 kHz Sine Wave	2.6% Pk	<4% Pk
0 dBm 1 kHz Sine Plus -20 dBm 1001 or 999 Hz Sine	0.2% Pk LAJ	<2.6% Pk LAJ
0 dBm 1 kHz Sine Plus -20 dBm 1002 or 998 Hz Sine	2.2% Pk LAJ	<7% Pk LAJ
0 dBm 1 kHz Sine Plus -20 dBm 1002 or 998 Hz Sine	0% Pk HAJ	<0.9% Pk HAJ
0 dBm 1 kHz Sine Plus -20 dBm 1004 or 996 Hz Sine	9.9% Pk LAJ	8-10.6% Pk LAJ
0 dBm 1 kHz Sine Plus -20 dBm 1005 or 995 Hz Sine	0.2% HAJ	<2.6% HAJ
0 dBm 1 kHz Sine Plus -20 dBm 1010 or 990 Hz Sine	2.0% Pk HAJ	<7% Pk HAJ
0 dBm 1 kHz Sine Plus -20 dBm 1008-1240 Hz Sine	9.8-10.2% Pk LAJ	9-10.6% Pk LAJ
0 dBm 1 kHz Sine Plus -20 dBm 760-992 Hz Sine	10.1-10.2% Pk LAJ	9-10.6% Pk LAJ
0 dBm 1 kHz Sine Plus -20 dBm 1020-1240 Hz Sine	9.8-10.2% Pk HAJ	9.4-10.6% Pk HAJ
0 dBm 1 kHz Sine Plus -20 dBm 760-980 Hz Sine	9.9-10.2% Pk HAJ	9.4-10.6% Pk HAJ
0 dBm 1 kHz Sine Plus -20 dBm 1300 or 700 Hz Sine	8.9-9.0% Pk	8.7-10.6% Pk
0 dBm 1 kHz Sine Plus -20 dBm 1500 or 500 Hz Sine	2.1% Pk	<2.6% Pk
0 dBm 1 kHz Sine Plus -20 dBm 0-300 Hz Sine	<0.1% Pk	<0.9% Pk
0 dBm 1 kHz Sine Plus -20 dBm 1.5-4 kHz Sine	<2.9% Pk	<2.6% Pk

0 dBm 1 kHz Sine Plus -20 dBm 4+ kHz Sine	<0.1% Pk	<0.5% Pk
1 kHz Sine Plus 20 dBm lower 1.1 kHz Sine -40-10 dBm Total Signal Power	10.1-10.4% Pk	<0.6% Pk difference
11.5º Pk-Pk PM, 0 dBm 1 kHz Carrier, 2-900 Hz Message	0.05-0.9% Pk	<0.9% Pk
AM, 1 kHz Carrier, 50% Duty Message, 5 Hz Gate	4.03% Peak, Steady	$\frac{2(max-min)}{(max+\text{min})}$ <10%
AM, 1 kHz Carrier, 50% Duty Message, 1 Hz Gate	4.03% Peak, Steady	$\frac{2(max-min)}{(max+\text{min})}$ >40%
0 dBm 1 kHz Sine Plus 1.1 kHz Sine Plus 1170 kHz Sine	19.65% Pk	18.5-19.8% Pk

5.4   Transient Measurements

A transient is any short duration change in a signal. The IEEE standards provides specifications for measuring impulse noise, phase hits, gain hits, and dropout transients [1]. Like the signal to noise ratio test, transient measurements that use a holding tone must also use a notch filter to eliminate the holding tone. However, unlike with noise level measurements, impulse noise measurements must have a notch filter that exactly matches the 1010 Hz notch filter specified by IEEE in order to meet the impulse requirements and must be used in conjunction with a C or D message filter for audio range measurements. For impulse noise measurements the test set must be able to accept impulse noise from 30-110 dBrn and be accurate to 1 dB with or without a holding tone. Drop outs are measured at a fixed level 12 dB below the initial level of the holding tone and must be accurate to within 1 dB after the qualification interval of 3.5 periods of the holding tone. Gain hits measurements must accept a threshold range between 2 and 10 dB and be accurate to within 0.5 dB after the qualification interval of 3.5 periods of the holding tone. Phase hits measurements must accept a threshold range between 5 and 45 degrees and be accurate to within 0.5 degrees plus 10% after the qualification interval of 3.5 periods of the holding tone.

Figure 5-2: HST 3000 [ViaviSolutions.com]

      To verify the virtual test set is capable of meeting the accuracy requirements specified by IEEE the virtual test set was looped to itself to measure its own output, which includes simulated impairments, and the measured result was compared to the measurements made by the HST-3000 test set, **Figure 5-2 [7]**, for impulse measurements and the Halcyon test set for phase and gain hit measurements. Nine sets of tests were performed to determine if the virtual test set transient measurements are within IEEE standards. The first test used 8us square pulses of varying amplitude to verify the threshold range and accuracy of impulse noise measurements with and without a holding tone. The second test uses tones and pulses specified by IEEE to verify the impulse and frequency responses of the C, D, and Notch filters. The third test used 5ms phase hits of varying amplitude to verify the threshold range and accuracy of phase hits. The fourth test used a second tone and phase hits as specified by IEEE to verify that a second tone will not trigger a phase hit. For the fifth test a 10dB 5ms gain hit was applied to verify that a gain hit will not trigger a phase hit. The sixth test used 5ms gain hits of varying amplitude to verify the threshold range and accuracy of gain hits. The seventh test used a second tone and gain hits as specified by IEEE to verify that a second tone will not trigger a gain hit. For the eighth test a 180 degree 5ms phase hit was applied to verify that a phase hit will not trigger a gain hit. The ninth test used a second tone and dropouts as specified by IEEE to verify that a second tone will not trigger a dropout. The measurements made during each of these tests without results specified by IEEE are available in **Appendix C** while the results are given in **Tables 5-12, 5-13, 5-14, 5-15, 5-16, 5-17, 5-18, and 5-19**.

Table 5-12 Impulse Noise Measurement Range and Accuracy

Test	Virtual Test Set	Standard
Measurement Range	8-112.2 dBrn	30-110 dBrn
Accuracy	+/- 1 dB	+/- 1 dB

Table 5-13 Impulse Noise Test Signals

Signal	Virtual Test Set	Standard
1 kHz 0 dBm Sine Wave, C-Message Filter	Counts at 93 dBrn	93 dBrn max
1 V 80 us square pulse, C-Message Filter	Counts at 83 dBrn	76 dBrn max
1.8 kHz 0 dBm Sine Wave, C-Notched Filter	Counts at 91 dBrn	91 dBrn max
1 V 80 us square pulse, C-Notched Filter	Counts at 83 dBrn	76 dBrn max
1 kHz 0 dBm Sine Wave, D-Message Filter	Counts at 93 dBrn	93 dBrn max
1 V 80 us square pulse, D-Message Filter	Counts at 84 dBrn	85 dBrn max
2 kHz 0 dBm Sine Wave, D-Notched Filter	Counts at 93 and 92 dBrn	93 dBrn max
1 V 80 us square pulse, D-Notched Filter	Counts at 84 dBrn	84 dBrn max

The virtual test set failed the impulse tests for the C-Message, C-Notched, and D-Message filters. The D-message filter is within the +/- accuracy range allowed for impulse readings however the C-Filter responses are significantly different than the expected results. This could be explained by the C-Filter not meeting the impulse response requirements of a C-Filter. However, after extensive testing, creating a filters using the exact zero and pole locations specified by the IEEE standard, and considering the significantly different results for the D-Filter, which is similar to the C-Filter, it was concluded that the result specified by IEEE is incorrect.

Table 5-14 Phase Hit Range and Accuracy

Test	Virtual Test Set	Standard
Measurement Range	5-45º	5-45º
Accuracy	+/- (20% + 0.5º)	+/- (10% + 0.5º)

Table 5-15 Phase Hit Test Signals

Signal	Virtual Test Set	Standard
-20 dBm 1004 Hz Sine with 25º 5 ms Hits Every Second Plus -30 dBm 300 Hz Sine	One count per second	One count per second
-20 dBm 1004 Hz Sine with 25º 5 ms Hits Every Second Plus -22 dBm 120 Hz Sine	One count per second	One count per second
-20 dBm 1004 Hz Sine with 25º 5 ms Hits Every Second Plus -30 dBm 3100 Hz Sine	One count per second	One count per second
10 dB Gain Hit	<10º	<10º

Table 5-16 Gain Hit Range and Accuracy

Test	Virtual Test Set	Standard
Measurement Range	2-10 dB	2-10 dB

Accuracy	+/- 0.5 dB	+/- 0.5 dB

Table 5-17 Gain Hit Test Signals

Signal	Virtual Test Set	Standard
-20 dBm 1004 Hz Sine with -6 dB 5 ms Hits Every Second Plus -30 dBm 300 Hz Sine	One count per second	One count per second
-20 dBm 1004 Hz Sine with -6 dB 5 ms Hits Every Second Plus -22 dBm 120 Hz Sine	One count per second	One count per second
-20 dBm 1004 Hz Sine with -6 dB 5 ms Hits Every Second Plus -30 dBm 3100 Hz Sine	One count per second	One count per second
180° Phase Hit	<2 dB	<2 dB

Table 5-18 Dropout Range and Accuracy

Test	Virtual Test Set	Standard
Measurement Range	12 dB	12 dB
Accuracy	+/- 1 dB	+/- 1 dB

Table 5-19 Dropout Test Signals

Signal	Virtual Test Set	Standard
-20 dBm 1004 Hz Sine with -20dB 5 ms Hits Every Second Plus -30 dBm 300 Hz Sine	One count per second	One count per second
-20 dBm 1004 Hz Sine with -20dB 5 ms Hits Every Second Plus -22 dBm 120 Hz Sine	One count per second	One count per second
-20 dBm 1004 Hz Sine with -20dB 5 ms Hits Every Second Plus -30 dBm 3100 Hz Sine	One count per second	One count per second

5.5   Echo Loss and Hybrid Circuit

Echo loss is the ratio of the original signal power reflected back to the transmitting circuit and is typically caused by unmatched impedance. A hybrid circuit converts a 4-wire transmission line into a 2-wire transmission line. The only IEEE specifications for a Hybrid Circuit are based on the results of an echo loss test [1]. The IEEE specifications of an echo lost test is based on the frequency response of three filters. One filter for echo return loss, one for low frequency singing return loss, and one for high frequency return loss.

The virtual test set uses a series of tones in its echo loss test so the range and accuracy of the virtual test set's echo loss test is the same as for single tone measurement. The only test performed using echo loss was the test specified by IEEE to validate a hybrid circuit. In this test, the test set is set to 900 Ohm impedance and echo loss tests are performed when the hybrid circuit is un-terminated and when the hybrid circuit is terminated with a 900 ohm resistor. The results are given in **Tables 5-20 and 5-21**.

Table 5-20 Echo Return Loss Range and Accuracy Measurements

Test	Virtual Test Set	Standard
Output Range	At least 12.1-(-58) dBm	-6-(-10) dBm
Output Signal to Noise Ratio	60 dB	>30 dB
Measurement Range (4-Wire)	17.4-(-70) dBm	-50-0 dBm
Measurement Range (2-Wire)	20.9-(-66.5) dBm	-40-0 dBm
Measurement Accuracy	+/- 0.2 dBm	+/- 0.5 dBm

Table 5-21 Hybrid Circuit Test

Result	Virtual Test Set	Standard
Echo Return Loss	28.3 dB	27.9-28.7 dB
Singing Return Loss Low	18.9 dB	18.5-19.3 dB
Singing Return Loss High	36.6 dB	36.2-37 dB

5.6 23-Tone Generation and Level Measurement

The 23-Tone signal is a standard signal where the frequency and phase difference between each of the tones is predetermined. This test signal is used for intermodulation distortion (IMD) and envelope delay distortion (EDD) tests. To meet IEEE specifications the test set must be able to generate the 23-tone signal with a total signal power level between -40 and -6 dBm with a signal to noise ratio of at least 45 dB and each of the individual tones must be generated within 1 dB of each other with a total signal power level accurate to within 0.2 dB **[1]**. The receive specifications for a 23-Tone test has the same level range requirements as the transmit specifications but the accuracy requirements depends on the 3rd harmonic IMDs.

To verify the virtual test set is capable of meeting the accuracy requirements specified by IEEE the virtual test set was looped to itself to measure its own output and the result was compared to the 23-Tone power level measured by the Halcyon test set. Two sets of tests were performed to determine if the virtual test set meets the IEEE requirements for generating and measuring 23-tone signals. The first test had the virtual test set transmit and receive a 23-tone signal at varying power levels to verify the virtual test set's range and accuracy with no 3rd harmonic IMDs. The second test used a 1.2 kHz signal of varying amplitude as a source of 3rd harmonic IMD to verify the accuracy of the virtual test set's 23-Tone level measurement with the presence of 3rd harmonic IMDs. The measurements made during each of these tests are available in **Appendix C** while the results are given in **Table 5-22**.

Table 5-22 23-Tone Signal Range and Accuracy

Signal	Virtual Test Set	Standard
Output Level Range	1.5-(-45) dBm	-40-(-6) dBm
Output Level Accuracy	+/- 0.02 dBm	+/- 0.1 dB
Output Signal to Noise Ratio	80 dB	>45 dB
Receive Level Range	5.5-(-41) dBm	-40-(-6) dBm
Level Measurement Accuracy	+/- 0.02 dBm	+/- 0.1 dB
Level Measurement Accuracy (3rd IMD 45-40 dB)	+/- 0.02 dBm	+/- 0.4 dB
Level Measurement Accuracy (3rd IMD 39-35 dB)	+/- 0.02 dBm	+/- 0.7 dB

Level Measurement Accuracy (3rd IMD 34-30 dB)	+/- 0.02 dBm	+/- 1 dB

5.7 Envelope Delay Distortion and Intermodulation Distortion

Envelope delay distortion and intermodulation distortion both use the 23-Tone signal to measure distortion in a signal. Envelope delay distortion measures the phases of the 23-Tones to quantify the amount of delay distortion caused by a system. A test set measuring EDD must be able to measure EDD up to 10000us with an accuracy within 10us or 40us under high 3rd harmonic IMD. Intermodulation distortion is a measure of the power level at specific frequencies caused by the 2nd and 3rd harmonics of the 23-Tone signal which represents the amount of IMD a system would cause. A test set measuring IMD must be able to measure 2nd harmonic IMDs and 3rd harmonic IMDs in the range of 20 to 60 dB with an accuracy between 1 and 3 dB depending on the IMD level.

To verify the virtual test set is capable of meeting the accuracy requirements specified by IEEE the virtual test set was looped to itself to measure its own output, which includes simulated impairments, and the measured result was compared to the EDD and IMD measurements made by the Halcyon test set. Fourth sets of tests were performed to determine if the virtual test set could measure EDD and IMD to and accuracy specified by IEEE **[1]**. The first test changed to initial phase of the 1921.87Hz tone in the 23-Tone signal by varying amounts as a source of envelope delay distortion to verify the range and accuracy of the virtual test set's EDD test. The second test added a 1.2kHz tone as a source of IMD to verify the accuracy of the virtual test set's EDD test with the presence of a high level IMD. The third test applied a gain loss of 5dB to verify that EDD measurements are not affected by the power of the 23-Tone signal. The forth test applied 1.2kHz and 1185Hz tones of varying amplitudes to the 23-Tone signal as a source of 3rd and 2nd harmonic IMD to verify the range and accuracy of the virtual test set's IMD measurements. The measurements made during each of these tests without results specified by IEEE are available in **Appendix C** while the results are given in **Tables 5-23, 5-24, and 5-25**.

Table 5-23 Envelope Delay Distortion Range and Accuracy

Signal	Virtual Test Set	Standard
Accuracy	+/- 1 us	+/- 10 us
Accuracy (3rd IMD 33-30 dB)	+/- 1 us	+/- 40 us
Measurement Range	12800 us	>10000 us

Table 5-24 EDD Test Signals

Signal	Virtual Test Set	Standard
Signal Gain or Loss of 5 dB	1 us result change	<5 us result change

Table 5-25 Intermodulation Distortion Range and Accuracy

Signal	Virtual Test Set	Standard
Measurement Range	At least 20-80 dB	20-60 dB
Accuracy (60-56 dB)	+/- 2.5 dB	+/- 3 dB
Accuracy (55-47 dB)	+/- 2.5 dB	+/- 2 dB

Accuracy (46-30 dB)	+/- 2.5 dB	+/- 1 dB
Accuracy (29-20 dB)	+/- 2.5 dB	+/- 2 dB

The virtual test set failed the requirements for the IMD measurements at higher IMD readings. The failure is caused by high fluctuations in the measurement readings. The readings can be made more stable by increasing the averaging time used during the test.

6   Future Work

The virtual test set is still a work in progress, more work must be done to get the virtual test set to meet all of the IEEE specifications for analog measurements. Furthermore, after the virtual test set was presented to field technicians it was decided to convert the virtual test set to have a more 'app based' user interface to simplify the use of the virtual test set among other hardware and software changes, the inclusion of the virtual test set into other systems, and the addition of other functions. Changes planned for the virtual test set include:

1. Convert UI to App based system.
2. Convert hardware to be switchless.
3. Combination of Virtual Test Set and Switch Matrix.
4. Combine Analog and Digital Virtual Test Sets.
5. Addition of wireless capability.
6. Complete standalone portability.

Convert UI to App based system: One of the requests of the field technicians is to make the user interface as simple as possible. Up until this point the main focus of the virtual test sets user interface has been to provide all of the functions with as much flexibility as possible. However, most of the field technician's job involves performing the same few set of tests several times for each of the systems they work on. For this reason, the focus of the virtual test set will shift to being 'app based' where the focus will be on providing several specialized functions.

Convert hardware to be switchless: Currently all of the different hardware settings are accessed using physical switches and the user has to ensure the software and hardware settings match. To eliminate possible errors in measurement, save time, and make things easier for the user it has been decided to use relays and digital output pins to automatically switch the settings using relays.

Combination of Virtual Test Set and Switch Matrix: While the virtual test set can be embedded into a computer and connected to a system for testing the connection process is still completely manual. In order to automate the connection process it is necessary to embed the virtual test set into a computer that controls a switch matrix, **Figure 6-1**, that is connected to all the systems the users wish to connect to **[8]**. A LabVIEW based automated test matrix has already been developed as part of the Programmable Integrated Switch Matrix (PrISM) and preliminary testing to combine the two projects has already been completed.
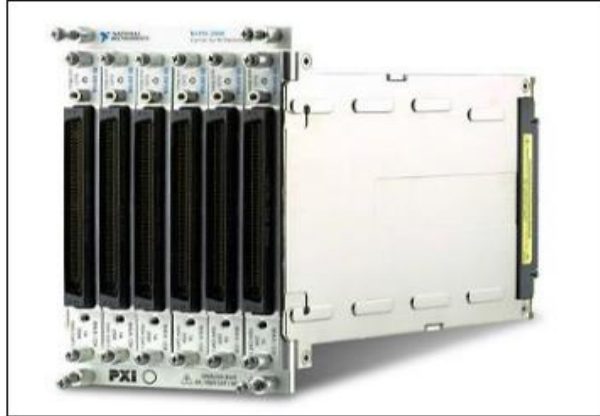
Figure 6-1: Switch Matrix Relay Card [NI.com]

Combine Analog and Digital Virtual Test Sets: The virtual test set was designed to perform audio range audio tests. Another project has developed a LabVIEW based test set for digital systems using a sbRIO, **Figure 6-2 [9]**. A new project is already underway to develop a virtual test set that will combine digital and analog testing capabilities into one test set.


Figure 6-2: SBRIO [NI.com]

Addition of wireless capability: One of the requested additions to the virtual test set was the ability to remotely control the tests the virtual test set performs. This will allow a technician to perform tests at multiple points on the system simultaneously. Implementation of this capability may range from wireless usb connections to wifi based communication but more research needs done before a decision can be made.

Complete standalone portability: The current virtual test set is portable but depends on a laptop for power, a user interface, and computing power. To truly make the virtual test set portable it will need its own built in computer and power supply. Research has been done into the possibility of using a raspberry pi as the computer. If a raspberry pi does will not work a tablet PC should suffice but more research needs done to determine the best option.

7 Conclusion

The virtual test set was developed to measure transmission impairments while meeting IEEE Std 743-1995 for audio range analog tests. This virtual test set can provide the functions of a TIMS as well as an oscilloscope, a spectrum analyzer, a DMM, and a Bell 202 modem while being smaller, lighter, and more cost effective than the hardware based measurement devices, **Figure 7-1**. The virtual test set has shown that it can be embedded into other systems, be updated with additional functions, and provide a user friendly interface.
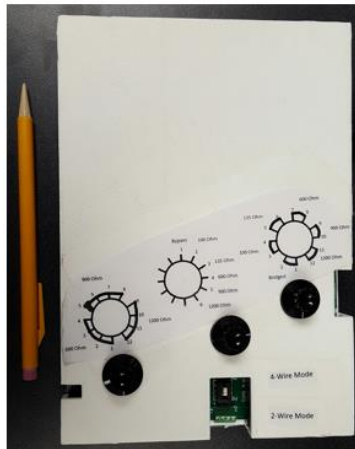
Total Cost: **$1000**
Total Weight: **1.2 lbs**.
Total Volume: **0.03 ft^3**

Total Cost: $20,650
Total Weight: 20.1 lbs.
Total Volume: 0.87 ft^3

Figure 7-1: Virtual Test Set and Hardware Test Equipment Comparison

Software based embedded test sets allows the capabilities of a test set to be expanded to meet increased demands on a system without the need for an entirely new test set for each iteration. Furthermore, virtual test sets are able to automate part of the preventative and corrective maintenance procedures and fewer hardware components allows for smaller, lighter, and cheaper test sets. This reduces the upkeep cost of a system and the burden on technicians in the field. In addition, because the measurements are performed using software, functions can be added, removed, or updated.

8 References

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9   Glossary

0TLP: The reference point all other TLPs in a circuit are compared to.

1004 Hz Net Loss: The attenuation of the 1004 Hz holding tone as it travels through a system.

2nd IMD: Noise signals caused by the second harmonics of a signal.

3rd IMD: Noise signals caused by the third harmonics of a signal.

Alternating Current: an electric current that reverses its direction many times a second at regular intervals.

Altium: An Australian owned public software company that provides PC-based electronics design software for engineers who design printed circuit boards.

Amplitude Jitter: Oscillations in the amplitude of a tone.

Amplitude Modulation: The process of transmitting a message over a high frequency by alternating the amplitude of the carrier based on the message signal.

Analog Signal: A signal that contains an infinite number of possible values.

Attenuation: The change in power of a signal through a filter or a system.

AutoCAD: A commercial computer-aided design (CAD) and drafting software application.

Balanced Line: A transmission line where the impedances on the transmit wire and the return wire are the same.

Block Diagram: The workspace for LabVIEW's graphical programing.

Carrier Frequency: The high frequency signal a message signal rides on during modulation.

C-Notched Noise Ratio: The ratio of signal power before and after being passed through a C-Message filter.

C-Message Filter: A filter designed to mimic the attenuation of human hearing.

Current: A flow of electric charge.

DAQ: A device used to convert physical signals into virtual signals and/or convert virtual signals to physical signals.

Direct Current: An electric current that flows in one direction.

Digital Multi-Meter: An electronic tool used for measuring current and voltage between two points in a circuit.

Drop Out: A dramatic loss of signal power.

Envelope Delay Distortion: The difference in delay between different frequencies.

Frequency Modulation: The process of transmitting a message over a high frequency by alternating the frequency of the carrier based on the message signal.

Frequency Shift Keying: A series of high and low frequency pulses used to transmit digital data over analog lines.

Front Panel: The user interface for a LabVIEW developed VI.

Gain: An increase or decrease in the power of a signal.

Gain Hit: A sudden increase or decrease in the power of a signal.

Graphical Programing: A method of programing where connectable icons represent an application's code.

Hybrid Circuit: A circuit used to convert 4-wire transmission lines to 2-wire transmission lines or 2-wire transmission lines to 4-wire transmission lines.

IEEE: the world's largest technical professional organization dedicated to advancing technology for the benefit of humanity.

IEEE standard 743-1995: An IEEE standard detailing the equipment requirements and measurement techniques for analog transmission parameters for telecommunications.

Impedance: A frequency dependent measure of the difficulty for and electric current to pass through an object or system.

Impulse Noise: Short duration spikes in a signal.

Intermodulation Distortion: Noise signals caused by non-linearity in a system.

LabVIEW: An integrated development environment designed specifically for engineers and scientists building measurement and control systems.

Message Signal: The signal to be transmitted over a higher frequency by modulating the carrier frequency.

Mobile Data Terminal: A portable computer used by technicians when performing work in the field.

Multisim: An advanced, industry-standard, best-in-class SPICE simulation environment used by educators, researchers, and engineers worldwide.

National Instruments: NI provides powerful, flexible technology solutions that accelerate productivity and drive rapid innovation. From daily tasks to grand challenges, NI helps engineers and scientists overcome complexity to exceed even their own expectations. Customers in nearly every industry—from healthcare and automotive to consumer electronics and particle physics—use NI's integrated hardware and software platform to improve our world.

Noise: Undesired frequencies in a signal.

Nyquist Rate: The minimum sample rate required to be able to completely reproduce a signal up to the desired frequency. The Nyquist Rate is two times the maximum frequency.

Operational Amplifier: A DC-coupled high-gain electronic voltage amplifier with a differential input and, usually, a single-ended output.

Operating System: The software that supports a computer's basic functions, such as scheduling tasks, executing applications, and controlling peripherals.

Oscilloscope: A measurement tool used to visualize a signal.

Parallel Processing: The process of a computer performing multiple functions simultaneously.

Phase Hit: A sudden change in the phase of a signal.

Phase Jitter: Fluctuations in the phase of a tone.

Phase Modulation: The process of transmitting a message over a high frequency by alternating the phase of the carrier based on the message signal.

Physical Signal: An electrical signal that exists on a transmission line.

Power: The amount of energy produced or consumed by a system in a given time span.

Queue: A lists of commands that are waiting to be executed. Commands are typically executed in the order they were queued.

Queued State Machine: A type of parallel processing that breaks its parallel processes into specific categories and uses a queue to control the flow of commands.

Resistance: A measure of the difficulty for and electric current to pass through an object or system.

RAM: A form of computer data storage which stores frequently used program instructions to increase the general speed of a system.

Selective Signaling: A series of high and low frequency pulses used to notify a position on a party line.

Spectrum Analyzer: A measurement tool used to visualize the frequency spectrum of a signal.

Sub-VI: A sub-routine of a VI.

Telecommunications: Communication over a distance by cable, telegraph, telephone, or broadcasting.

TIMS: A test set used to measure the impairments of a system by sending controlled tones through the system and measuring the changes in the signal.

Transmission Level Point: A measure of the attenuation at a given point in a system compared to a reference point in the system.

Transmission Line: The physical wires connecting two circuits.

Virtual Instrument: A program that performs a set of functions designed in LabVIEW.

Virtual Signal: A series of data points in a computer that represents a physical signal.

VIRTUAL TEST SET: A virtual test set designed to perform audio range impairment measurements that can be embedded in a system.

10 Appendix A: Hardware Details

10.1 DAQ

In order to meet IEEE standards numerous sample rate and power requirements must be meet, **Table 10-1**. Most voice band analog signals are band limited to 4000 Hz and must be sampled at a rate equal to twice the highest information rate. This rate of sampling is called the Nyquist rate and is the minimum sampling frequency required to extract all the information contained in the original signal. Because only the 4000-Hz bandwidth for the voice signals is specified, the Nyquist rate is 8000 Hz. The MyDAQ, with a sample rate of 200 kHz **[2]**, is able to meet the Nyquist rate for signals up to 100 kHz and easily exceeds the sampling requirements for audio range signals. Because higher sample rates have longer settling times, resulting in less accurate readings, a sample rate of 100 kHz is the default for the virtual test set.

Table 10-1: Transmit and Receive Requirements

Specification	IEEE Requirement	Virtual Test Set
Sample Rate	8kS/s	200kS/s
Transmit Tone Level (100 Ohm)	10-(-40)dBm	17.1-(-58) dBm
Transmit Tone Level (135 Ohm)	10-(-40)dBm	18.4-(-59) dBm
Transmit Tone Level (600 Ohm)	10-(-40)dBm	15.1-(-66) dBm *
Transmit Tone Level (900 Ohm)	10-(-40)dBm	13.4-(-68) dBm *
Transmit Tone Level (1200 Ohm)	NA	12.1(-69) dBm *
Receive Tone Level (100 Ohm)	24.4-(-49.9)dBm	27.0-(-62) dBm
Receive Tone Level (135 Ohm)	23.1-(-49.9)dBm	25.7-(-63) dBm
Receive Tone Level (600 Ohm)	16.6-(-49.9)dBm	19.2-(-70) dBm ***
Receive Tone Level (900 Ohm)	14.9-(-49.9)dBm	17.4-(-72) dBm ***
Receive Tone Level (1200 Ohm)	NA	16.2(-73) dBm ***
Receive Tone Level (Bridged)	16.6-(-49.9)dBm**	19.2-(-70) dBm **

*Values reduced by 3.5dBm in 2-Wire mode.

** Bridged mode assumes an impedance of 600 ohms.

***Values increased by 3.5dBm in 2-Wire mode.

10.1.1 MyDAQ Pinout

The NI MyDAQ reads and writes data as well as provides power through its input and output pins, **Figure 10-1 [2]**. These pins include 2 analog output pins, 2 differential analog input pairs, 8 digital input/output pins, 3 ground pins, a 5V power pin, and a pair of +/-15V pins. The virtual test set utilizes one analog output pin, one analog input differential pair, one ground pin, and the power pins. The analog input pins are capable of directly reading analog signals but the analog output pins require an amplifier circuit in order to meet the IEEE output power requirements **[1]**. The MyDAQ maximum output levels are given in **Table 10-2**.
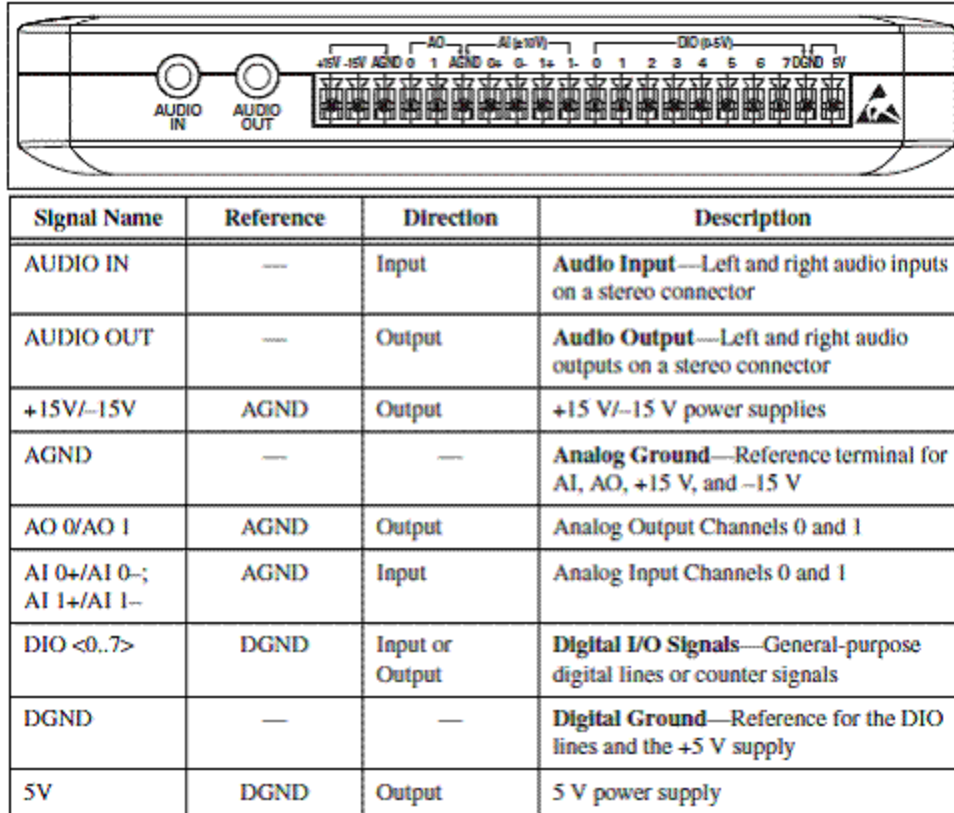
Figure 10-1: MyDAQ Pins [NI.com]

Table 10-2: MyDAQ Output

Output Pin	Voltage Output	Max Current
Analog Out	+/-10V Range	2mA
+/-15V Pins	+/- 15V Respectively	32mA
5V Pin	4.9V	100mA
Digital Pins (High)	>2V	4mA
Digital Pins (Low)	<0.8V	4mA

10.1.2 MyDAQ DMM

The MyDAQ is unique among other NI DAQs in that it has a built in DMM for taking voltage, current, and resistance measurements, **Figure 10-2 [2]**. The 60V range the MyDAQ can tolerate is limited compared to other standalone DMMs but is sufficient for audio range systems. The DMM is not required to meet IEEE audio range test equipment standards but is used to measure on-hook/off-hook and battery backup voltages found in voice switch systems without the need for an extra test device. The full measurement range of the MyDAQ's DMM is give in **Table 10-3**.
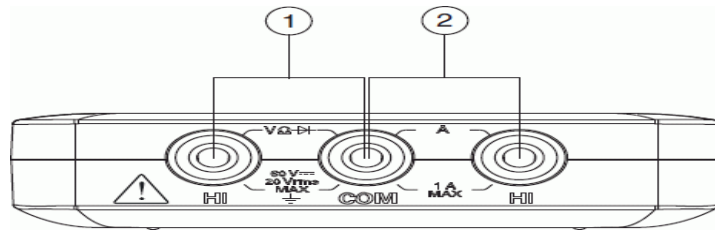
Figure 10-2: MyDAQ DMM Connections [NI.com]

Table 10-3: MyDAQ DMM Measurements

Measurement	Range
DC Voltage	60V
AC Voltage	20Vrms
DC Current	1A
AC Current	1Arms
Resistance	20MOhms
Diode Voltage	2V
Continuity	NA

10.2 Line Interface

In addition to connecting the virtual test set to transmission lines, the Line interface provides amplification, impedance matching, and 2 to 4 wire conversion. The circuit board design for the amplifier, impedance matching, and hybrid circuits was designed in Altium and is shown in **Figure 10-3**.
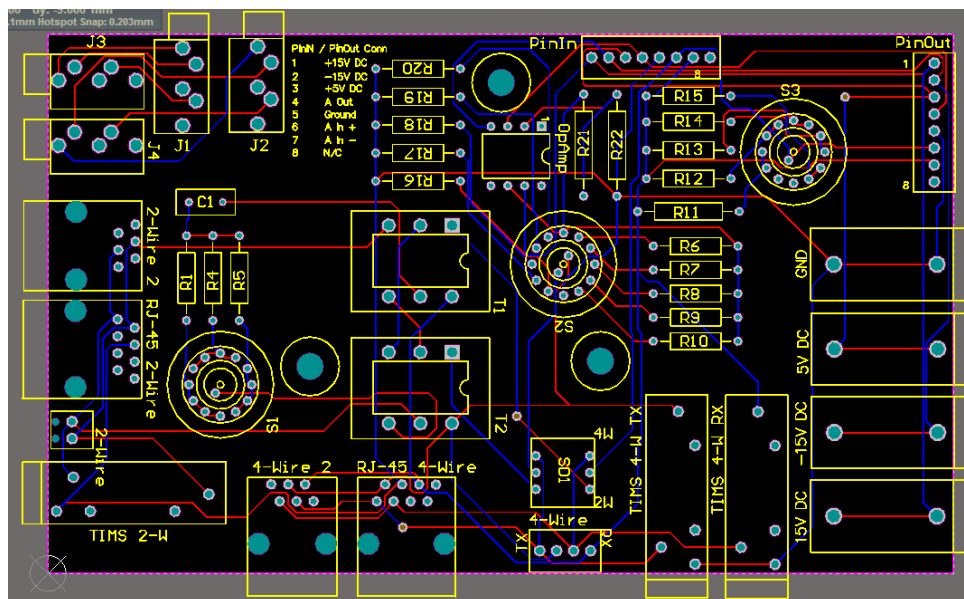
Figure 10-3: 2-4 Wire Converter and Impedance Matcher.

10.2.1 Signal Amplifier

In order to meet the IEEE specifications **[1]**, the virtual test set must be capable of providing specific output power levels for given load resistances, **Table 10-4**. Higher resistances require higher output voltages for a given power level while lower resistances require higher output currents for a given power level. The analog output of the MyDAQ is incapable of meeting these requirements on its own. A simple op-amp voltage amplifier circuit is used to source current from the +/- 15V power pins and provide a 50% voltage boost to the analog output voltage to meet the IEEE output requirements in all settings.

Table 10-4: Output Requirements and Specifications

Output Specification	Peak Source Voltage (Hybrid Circuit)	Peak Source Voltage (No Hybrid)	Peak Voltage (Load)	Peak Current
IEEE Requirement 900Ohm, 10dBm	12.72V	8.48V	4.24V	4.7mA
IEEE Requirement 600Ohm, 10dBm	10.38V	6.92V	3.46V	5.8mA
IEEE Requirement 135Ohm, 10dBm	4.92V	3.28V	1.64V	12.2mA
IEEE Requirement 100Ohm, 10dBm	4.26V	2.84V	1.42V	14.2mA
MyDAQ	10V	10V	10V	2mA
VIRTUAL TEST SET	13V	13V	13V	32mA

The op-amp used in the circuit, **Figure 10-4,** was specially chosen because of its high output voltage swing, typically +/-13V, compared to its supply voltage, +/- 15V **[9]**. In addition, it meets other important requirements such as bandwidth, 200kHz, and output current, up to 38mA. The other elements of the circuit can be replaced as long as they satisfy the requirements for a non-inverting op-amp circuit with 50% amplification, **Equation 10-1**.

$$v_o = \left(1 + \frac{R_2}{R_1}\right) * v_i \qquad (10\text{-}1)$$

Where R_1 is the inverting lead resistance, R_2 is the feedback resistance, and $R_2 = R_1/2$.
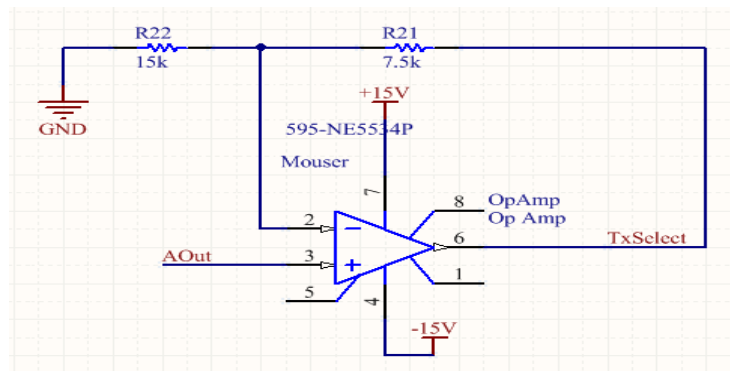
Figure 10-4: Line Interface Signal Amplifier Circuit

10.2.2 Impedance Matching

Impedance matching is the process of matching the impedance, resistance and reactance, of the transmission circuit and the load circuit. For many systems, impedance matching is as simple as adding a load to the transmitting circuit so that the total internal resistance of the source is equal to the load resistance. This maximizes the power transmitted to the load and minimizes the power reflected back to the transmitting circuit [11]. For complex loads, those with a resistive and reactive component, the impedance of the source must be the complex conjugate of the load impedance to cancel out the reactive components. Furthermore, a hybrid circuit used for 2-4 wire conversion requires that the impedance of the 'balance circuit' be equal to the impedance of the load circuit in order to minimize the power reflected back to the source, **Figure 10-5 [12]**. Impedance matching is important to reduce signal reflection and the associated noise on a communication line but comes at the cost of increasing the total power output by the signal generator, double for regular circuits and triple for hybrid circuits.
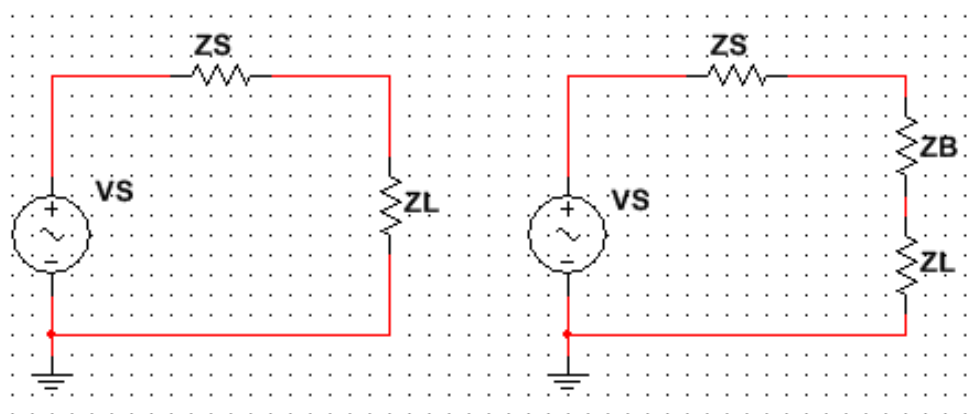
Figure 10-5: Simplified Impedance Matching for normal (Left) and balanced hybrid (Right) circuits. ZS=Source Impedance, ZL=Load Impedance, ZB=Hybrid Balance Impedance.

Impedance matching on the Line Interface is done using resistors to match the standard impedance values and rotary switches to select the different impedances. In addition to providing the standard impedance values, the Line Interface provides a bypass option for the transmit circuit and a bridge option for the receive circuit, **Table 10-5**. Bypass applies voltage directly across the lead terminals to use external impedance matching circuits and to minimize the output power required by the Line Interface. Bridge measures voltage across the line with an open circuit so measurements can be made on an active circuit without interfering with the signals being transmitted over the line. Impedances on the receiving circuit are implemented with a single resistor, **Figure 10-6**. Impedances on the transmitting circuit have to be the same on the positive and negative leads to provide a balanced line and further reduce signal reflection, **Figure 10-7**.

Table 10-5: Virtual Test Set Impedance Values

Rotary Switch	Impedance Values
Transmit	Bypass, 100Ω, 135Ω, 600Ω, 900Ω, 1200Ω
Receive	100Ω, 135Ω, 600Ω, 900Ω, 1200Ω, Bridge
Hybrid Balance	600Ω, 900Ω, 1200Ω
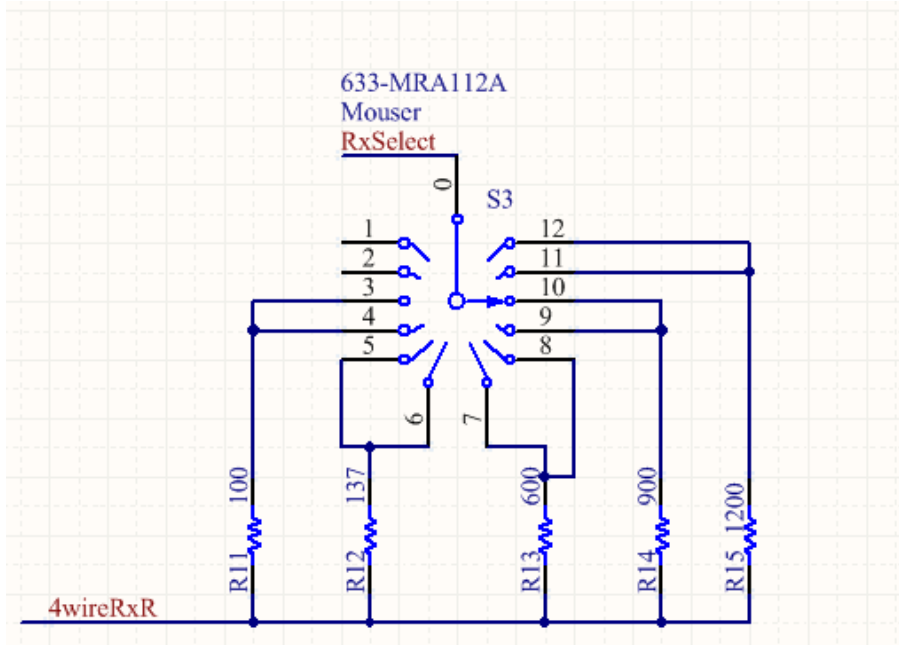
Figure 10-6: Virtual test set impedance matching for the receive circuit.
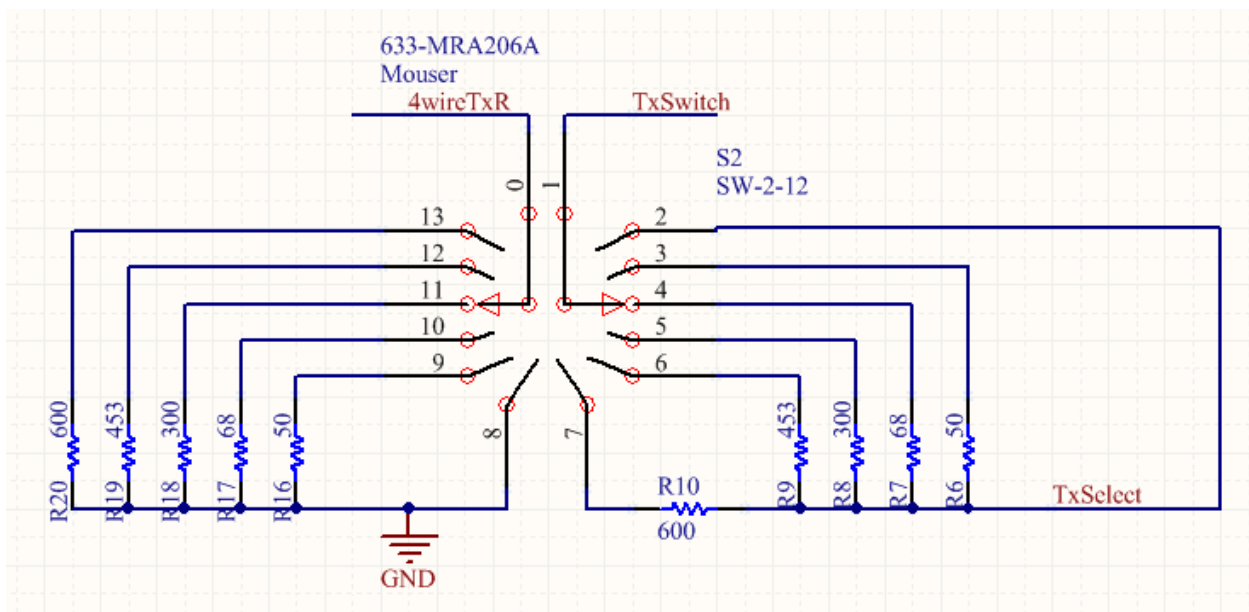
Figure 10-7: Virtual test set impedance matching for the transmit circuit.

10.2.3 Hybrid Circuit

A hybrid circuit is used to connect a 4-wire circuit to a 2-wire circuit. All communication circuits require at least two wires, a signal wire and a return/ground wire. In a 4-wire circuit one 2-wire pair is used for transmitting signals and the other 2-wire pair is used to receive signals. Alternatively, 2-wire circuits use the same pair of wires for transmitting and receiving signals. To connect a 4-wire circuit to a 2-wire circuit a hybrid circuit is necessary to ensure that the transmitted signals are not sent over the receiving line. The hybrid circuit uses transformers to connect the 2-wire, 4-wire transmit, 4-wire receive, and balancing circuits. When a signal enters the hybrid circuit it is split between the two adjacent circuits while undergoing a 90 degree phase shift. No signal power is transferred to the opposite circuit because the signals split between the adjacent circuits undergo opposite 90 degree phase shifts and cancel out, **Figure 10-8 [13]**. In order to ensure the necessary phase shifts and minimize the reflection signals it is necessary to ensure that the impedances of the opposite circuits are matched, the impedances of the 4-wire transmit and receive circuits as well as the 2-wire and balancing circuits must be matched **[14]**. The virtual test set's hybrid circuit includes a SPDT, Single Pull Double Throw, switch to switch between two and 4 wire modes and a rotary switch to select between the standard impedances for the balance circuit, **Figure 10-9**.
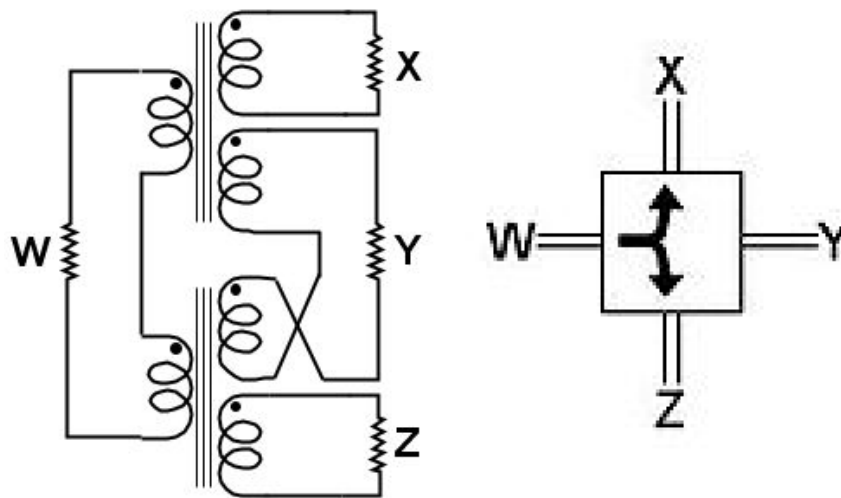
Figure 10-8: Simplified Hybrid Circuit [Wikipedia]
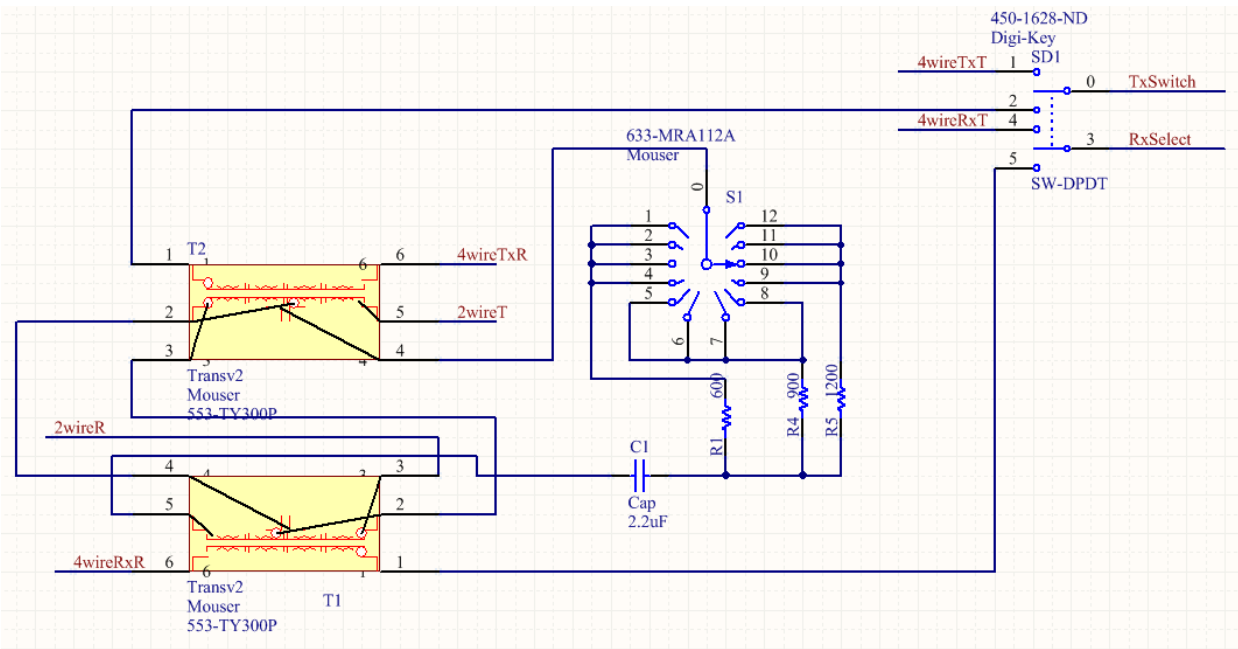
Figure 10-9: Virtual Test Set Hybrid Circuit.

10.3 Calibration

The process calibrating the virtual test set is broken into calibrating the input and calibrating the output according to the different hardware settings. Two virtual test sets and a total of 10 input and 18 output calibrations need to be performed to completely calibrate a virtual test set. Each calibration follows four basic steps: select the hardware settings, make the physical connection, capture the signal, and finish the calibration as shown in **Figure 10-10**.
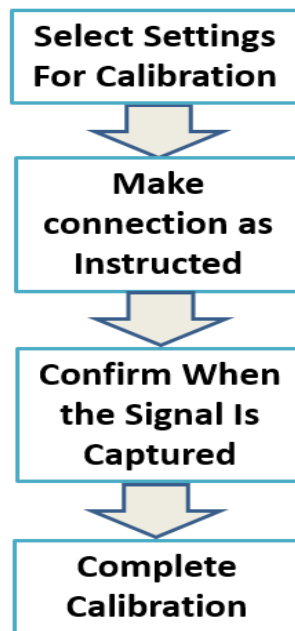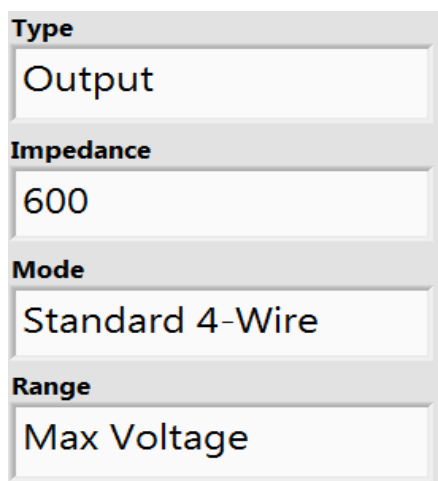
Figure 10-10: Calibration Flow Chart

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10.3.1 Select the Hardware Settings

The first step to performing any calibration is choosing which settings to calibrate for, **Figure 10-11**. Separate calibrations must be performed for each of the hardware setting combinations. The calibration type determines whether input calibration, output calibration, or a self-check will be performed. Mode determines the connection mode that is being calibrated for; 2-Wire, 4-Wire, bridged, or bypass. Bridged mode can only be used during input calibration and bypass mode can only be used during output calibration. Impedance sets the impedance value that will be used during the calibration and the options available depend on the other settings: 4-wire input, bridged, and bypass modes only use 600 ohms; 4-wire output mode uses 100, 125, 600, 900, and 1200 Ohms; and 2-wire mode uses 600, 900 and 1200 Ohms. For proper calibration, the physical hardware settings must match the calibration hardware settings. Once all the settings have been chosen the user clicks the Start Button to begin calibrating.

Figure 10-11: Calibration Settings

10.3.2 Make the physical connection

Once the hardware settings have been selected, the user pushes the Start button to begin calibrating. A simple set of instructions will be displayed to instruct the user on the kind of connection that needs made, **Figure 10-12**. For all input calibrations the specified signal needs to be applied to the input of the virtual test set. This signal is a triangle wave of specified 8V amplitude and 0.5Hz frequency for 4-wire and Bridged mode and is a sine wave of 8V amplitude and 1004Hz frequency for 2-Wire mode. In addition to applying the appropriate signal, the signal source in 2-wire mode must be appropriately impedance matched. 4-Wire and bypass output calibrations are performed simply by looping the virtual test set back to itself and only requires that the virtual test set's input calibrations have already been performed. 2-Wire output mode requires a second virtual test set with calibrated inputs to calibrate. Both virtual test sets must be set to the same calibration settings and the user will be prompted to set one as the virtual test set that is providing calibration and one as the virtual test set

that is being calibrated. The virtual test set that is providing calibration will need to be set to the same physical impedance but will receive using 4-wire mode. Once the connection has been made, the user turns on the virtual test set's transmit and receive to capture the signal.

Figure 10-12: Calibration Instructions.

10.3.3 Capture the Signal

The signal capture process begins when the user turns on the virtual test set's transmit and receive functions. When a recognizable signal is captured by the virtual test set solid sloping lines will be displayed in the Measured Amplitude and Difference from Ideal graphs, **Figure 10-13**. The measured amplitude graph displays the amplitude values of the captured signal from its maximum to its minimum value. The difference from ideal graph displays the difference between the signal that was captured and the ideal signal that should have been applied to the virtual test set for input calibrations or transmitted by the virtual test set for output calibrations. Because of the various amplifications and impedance losses applied by the connector circuit, only the 4-Wire input and Bridged modes will have a flat line for their difference from ideal graph. Once the user confirms the measured amplitude graph displays a smooth and steady line, they click the finish button to finish the calibration.
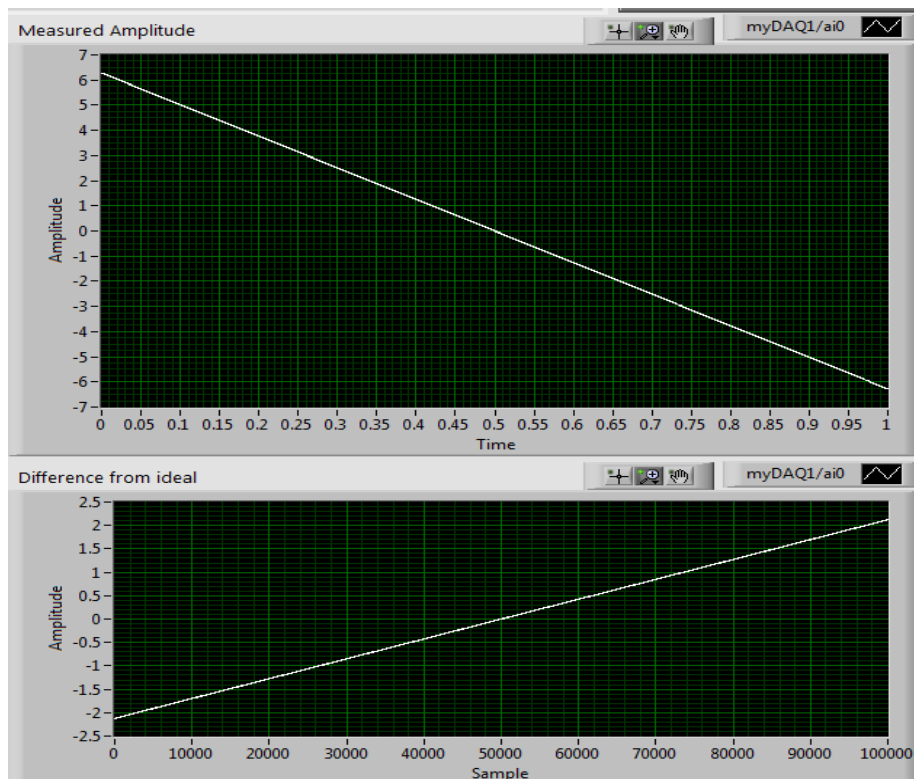
Figure 10-13: Captured signal during calibration.

10.3.4 Finish the Calibration

For most calibrations simply clicking finish will complete the calibration. However, in 2-Wire output mode the calibration constants must be entered manually into the virtual test set being calibrated. These calibration constants are generated by calculating the constants for the best fit polynomial of the difference from ideal graph generated by the virtual test set providing the calibration and are displayed once the Finish button is pressed. The 2-wire output mode calibration is completed once the user confirms they have entered the calibration constants correctly. Additionally, the user will be asked if they wish to save a copy of the values from the Captured Amplitude graph which will be placed in a text file for later review. Once the calibration has successfully completed the new values will be applied to the virtual test set's calibrations and the graphs displaying the actual vs ideal values will be updated, **Figure 10-14**. If a mistake was made at any point during the calibration process, the user may select and roll back to a previous calibration setting.
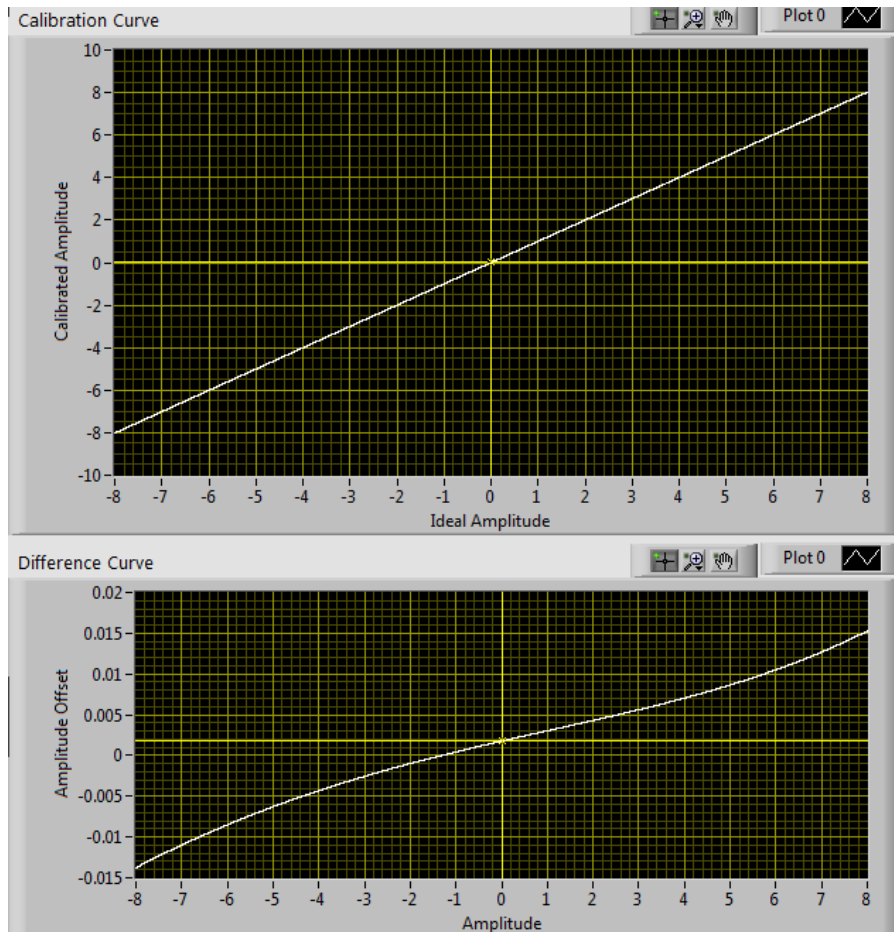
Figure 10-14: Calibration curves.

11 Appendix B: Software Details

11.1 Common Elements

The virtual test set and all of its major working components use Queued State Machines as part of their flow control. Because each major function of the virtual test set works the same way they inevitably have components that are similar. These common components include control events, queue events, and sub-VIs.

11.1.1 Common Control Events

Control events are handled by Event Case Structures. Event Case Structures perform specific actions when the user performs an action with a designated control. The action can be anything from changing the value to the control to mousing over the control. The following events are shared between many of the virtual test set's functions:

1. Idle.
2. Close the Function.
3. Change the displayed help window information.
4. Toggle Transmit.
5. Toggle Receive.

Idle: This event case is triggered when no other actions are made by the user for 200ms. This event case typically does nothing.

Change the displayed help window information: This event case is triggered whenever the mouse touches a new control and updates the information displayed in the help window. The event case uses the label of the control to search for the help information contained in the help text file, which is located in the virtual test set's application folder, **Figure 11-1**. If multiple controls perform the same function they are given the same label and a number. The number is removed before searching for the help information, which allows multiple controls to display the same information.
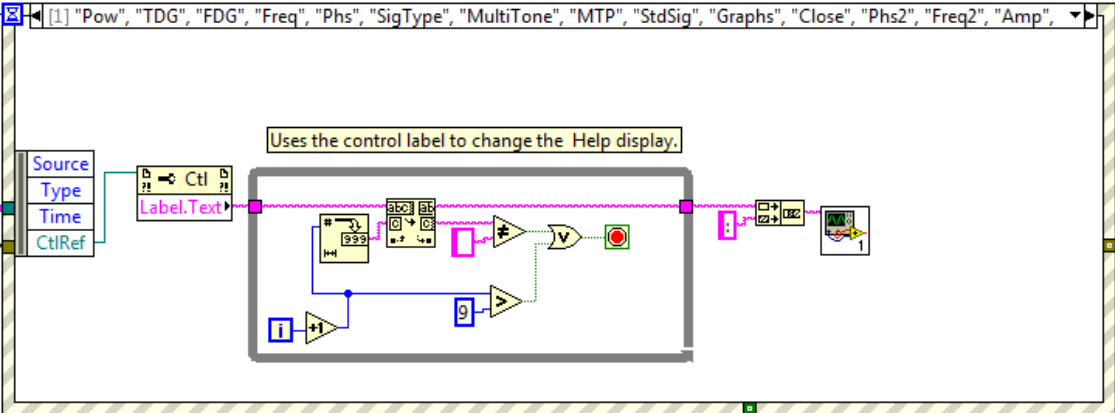
Figure 11-1: Change the displayed help window information Event Case.

Toggle Transmit and/or Receive: This event case is activated whenever the user selects to turn on or off the transmit and/or receive functions. If the function does not have direct control over the virtual test set's transmit or receive functions it must queue a request for the transmit or receive function to be turned on or off, **Figure 11-2**.
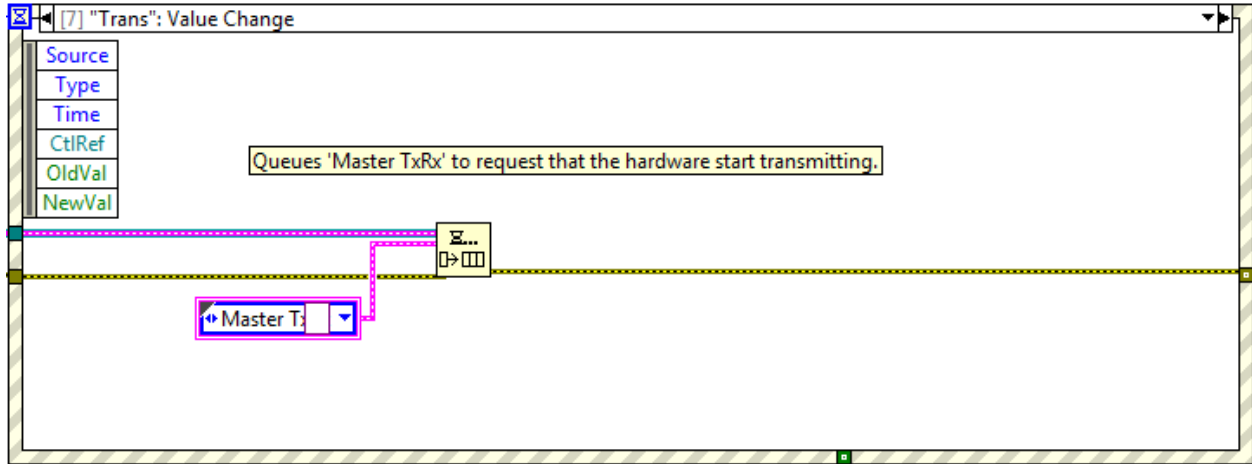
Figure 11-2: Toggle Transmit and/or Receive Event Case

Close: This event is called when the user wishes to close the current function and return to the previous menu. A reference to the function is opened to check if the function window is open and closes it if it is, **Figure 11-3**.
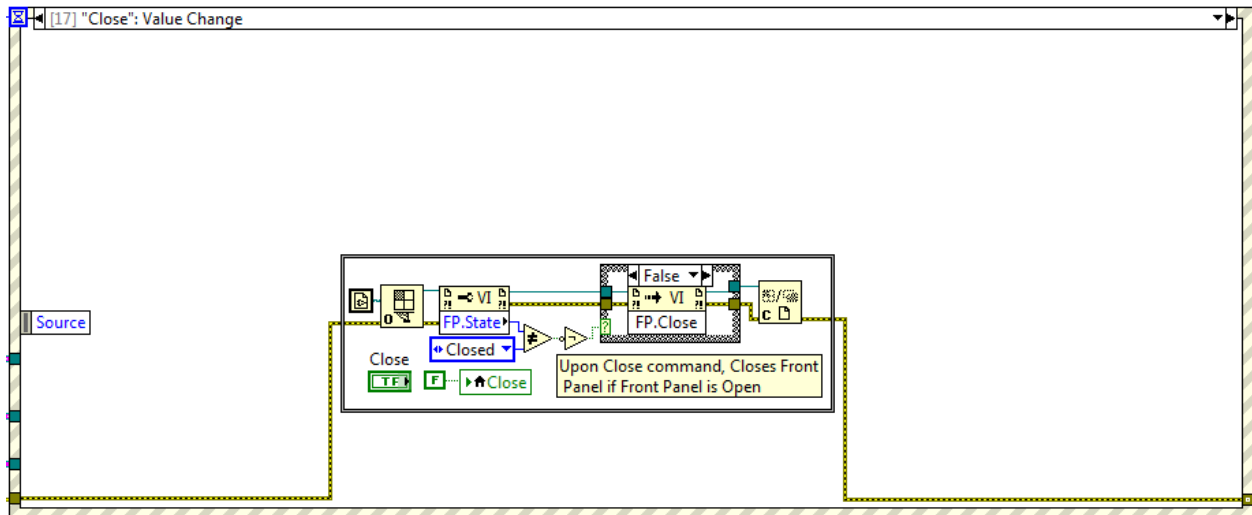
Figure 11-3: Close Event Case

11.1.2 Common Queue Commands

Queue commands for a queued state machine are handled by the function's queue loop, . All Queue Loops are broken into 3 basic components: the de-queue element, the command case structure, and queue management, **Figure 11-4**.
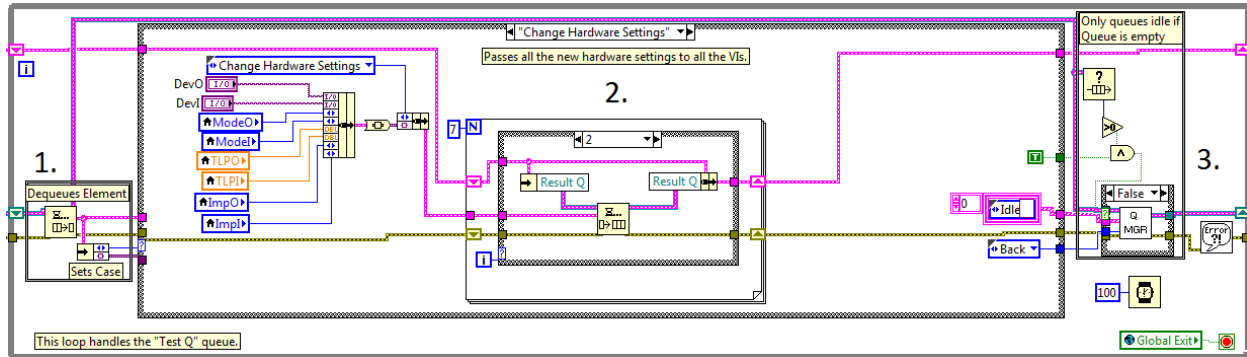
Figure 11-4: Virtual Test Set Queue Loop.

1. The De-Queue Element. This element of the Queue Loop removes the item at the front of the queue and breaks it into the different components used by the Command Case Structure.
2. The Command Case Structure. This element executes various actions depending on the command queued.
3. The Queue Management. This element handles adding commands to either the front or the back of the queue depending on the information sent from the command that was just executed. This is primarily done using the Queue Manager Sub-VI. In the case that an Idle command is queued by the executed command, the idle will only be entered into the queue if the queue is empty. This element also controls the speed that commands are processed and handles any errors that occur.

The queue loop performs specific actions as they are queued by other parts of the program. These actions can be queued by any function with the appropriate queue reference and are executed in the order they are queued. The following queue commands are shared between many of the VIRTUAL TEST SET's functions:

1. Initialize
2. Idle
3. Change Hardware Settings
4. Master TxRx
5. Auto TxRx
6. Error
7. Close Test Suite

Initialize: This command is called when the virtual test set starts and opens a reference to all the other queues the function communicates with, **Figure 11-5**.
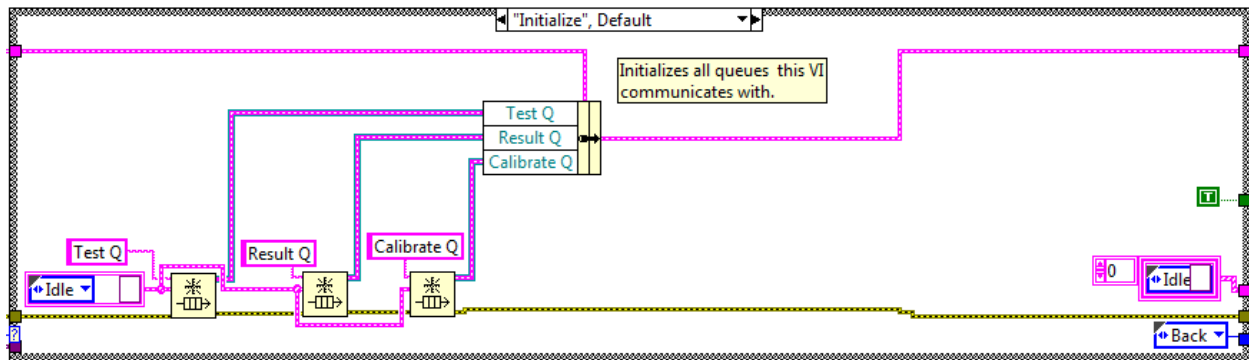
Figure 11-5: Initialize Queue Event

Idle: This command is queued when there are no other commands and waits for a new command to be queued.

Change Hardware Settings: This command is queued whenever the user changes the hardware settings. Every function that receives the command will: stop communicating with the hardware, update the maximum output signal strength, ensure the current signal does not exceed the new limit, and adjust the calibration settings, **Figure 11-6**. This command ensures no hardware errors occur during normal operation of the virtual test set.
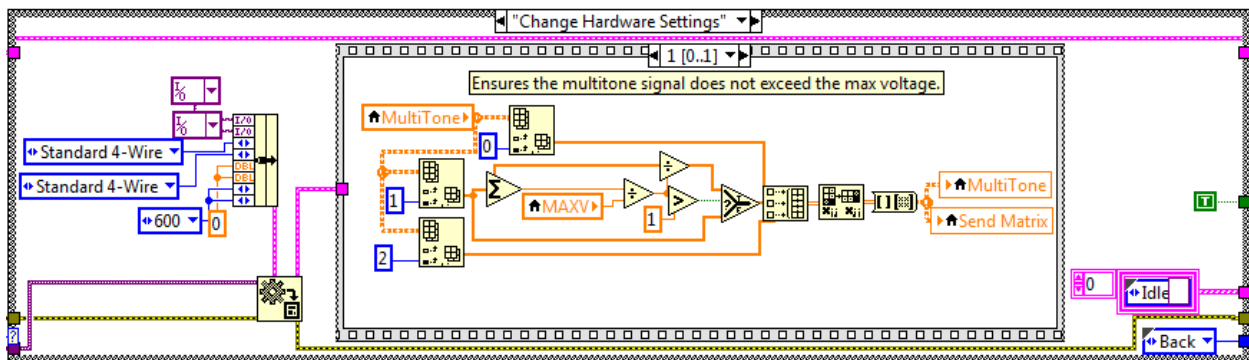
Figure 11-6: Change Hardware Settings Queue Event

Master TxRx: This command is queued whenever one of the virtual test set functions needs to turn on or off the transmit or receive functions. The function requesting the change will queue the command into the virtual test set's main queue, indicating which function it is changing and if it is turning the function on or off, **Figure 11-7**. The main virtual test set responds by notifying all other functions of the change.

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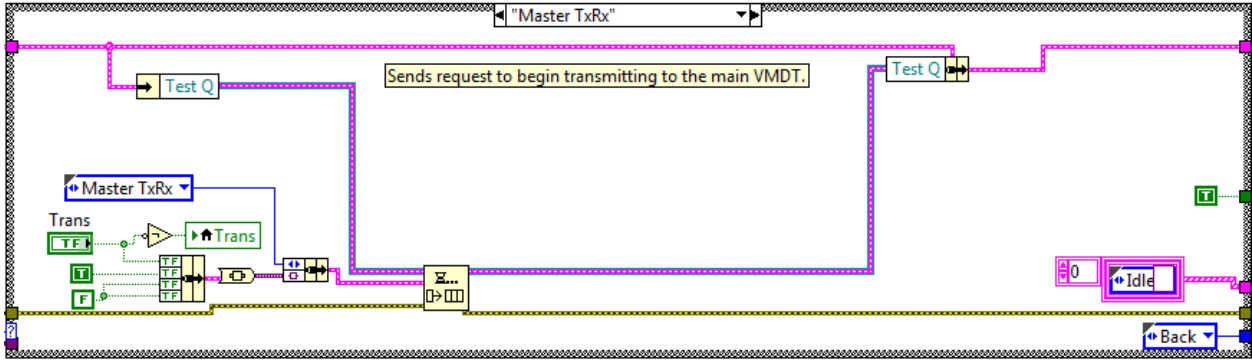
Figure 11-7: Master TxRx Queue Event

Auto TxRx: This command is queued by the main virtual test set queue after receiving a request to turn the transmit or receive on or off. This command updates the indicators of the virtual test set's functions to reflect the change as well as update the communication with the hardware, **Figure 11-8**.
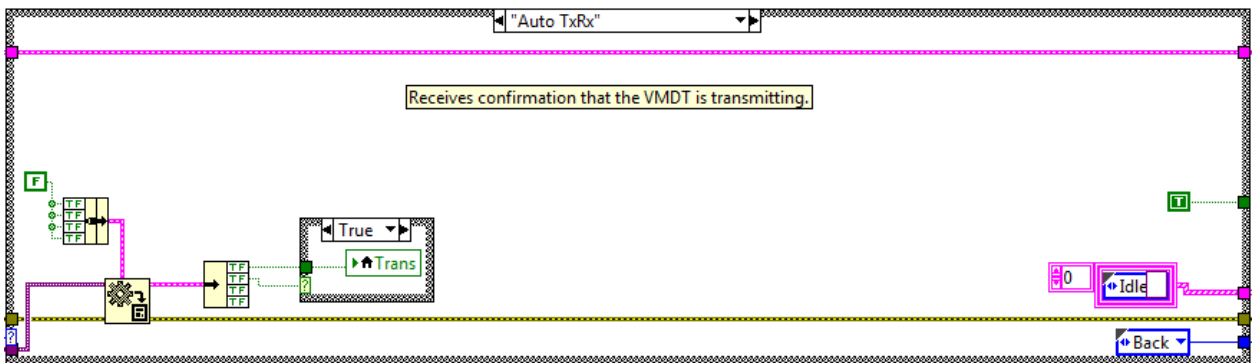
Figure 11-8: Auto TxRx Queue Event

Error: This command is queued whenever an error is encounter. The user is informed of the error and, if possible, resumes normal operation of the virtual test set, **Figure 11-9**.
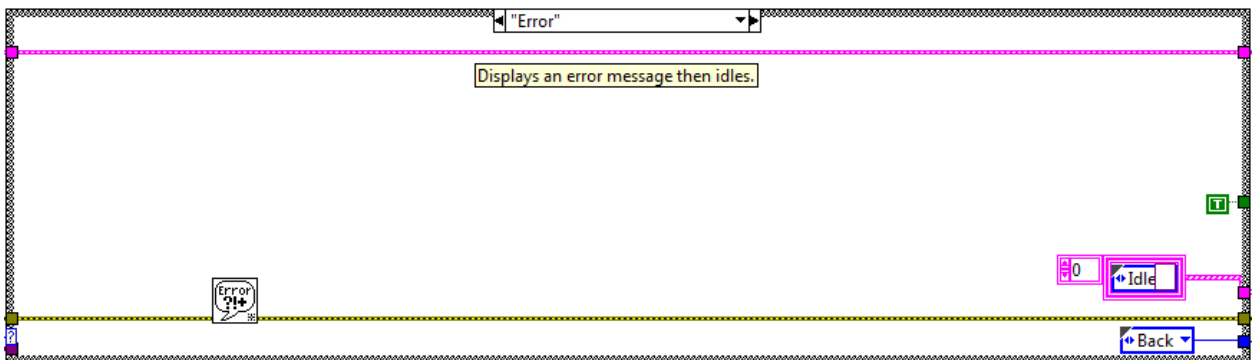
Close Test Suite: This command is queued when the virtual test set is shut down. The command is passed to all of the virtual test set's functions to close their windows, **Figure 11-10**.
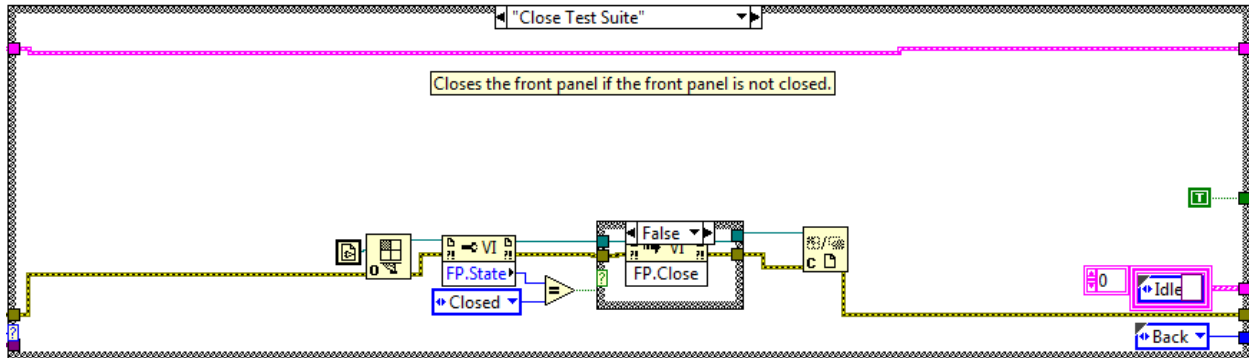
Figure 11-10: Close Test Suite Queue Event

11.1.3 Common Sub-VIs:

Many of the virtual test set's functions need to perform the same actions repeatedly throughout the operation of the virtual test set. The steps needed to perform these actions are made into sub-VIs that can be called any of the virtual test set's functions. Sub-VIs that are called by multiple functions include:

1. Update the help window.
2. Calibrate Transmit.
3. Calibrate Receive.
4. Queue Manager.
5. Convert volts to dBm.
6. Convert dBm to volts.
7. DAQ Analog Output Manager.
8. DAQ Analog Input Manager.
9. Convert dB.

Update the help window: This sub-VI is used to update the information displayed on the help window. It reads the text located within the application folder's 'HelpLogs' file and displays the text between the labels passed to it from other VIs, **Figure 11-11**.
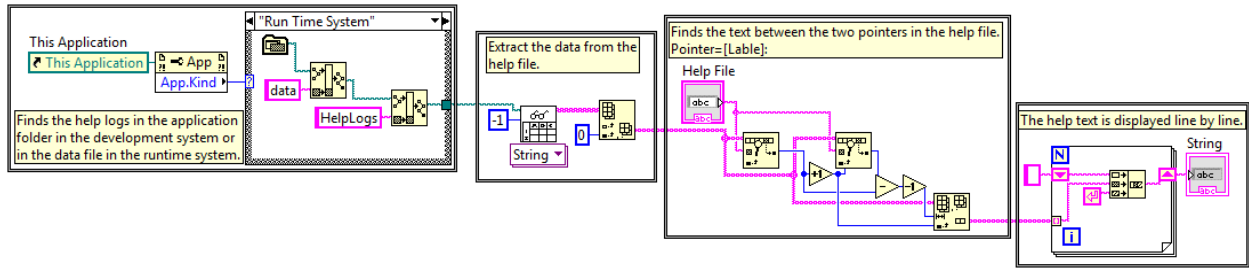
Figure 11-11: Update Help Window Block Diagram

Calibrate Transmit: This sub-VI is used to calibrate the transmitted signal to accurately reproduce the signal the user has set to transmit. This VI is called in two separate cases: when the hardware settings are changed and while the virtual test set is transmitting, **Figure 11-12**. When the hardware settings are changed the 'Output Calibrations' file is read and the hardware settings are used to find the appropriate calibration values as well as calculate the scaling factor for the set TLP using **Equation 11-1**. When the virtual test set is transmitting the VI applies the TLP multiplier, checks if the maximum value will exceed 2 volts after calibration, and applies the high or low voltage calibration, based on the calculated range, according to **Equation 11-2**.
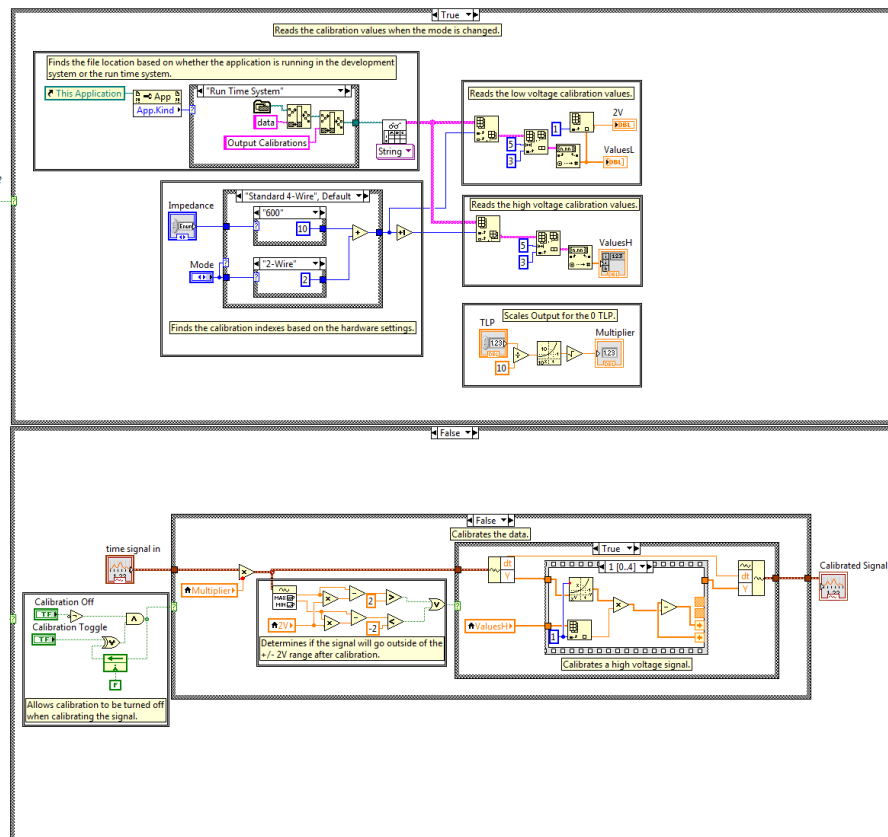
Figure 11-12: Calibrate Transmit Block Diagram, True and False Cases.

$$M = \sqrt{10^{\left(\frac{T_{LP}}{10}\right)}} \qquad \text{(11-1)}$$

Where M is the multiplier and T_{LP} is the transmission level point.

$$f_c(t) = Mf(t) - \left(c_0 + \sum_{n=1}^{4} (Mf(t))^n c_n\right) \qquad \text{(11-2)}$$

Where $f_c(t)$ is the calibrated signal, $f(t)$ is the un-calibrated signal, M is the TLP multiplier, and the c_n are the calibration constants.

Calibrate Receive: This sub-VI is used to calibrate the received signal to accurately represent the signal received by the hardware. This VI is called in two separate cases: when the hardware settings are changed and while the virtual test set is receiving, **Figure 11-13**. When the hardware settings are changed the 'Input Calibrations' file is read and the hardware settings are used to find the appropriate calibration values as well as calculate the scaling factor for the set TLP using **Equation 11-1**. When the virtual test set is receiving the VI applies the TLP multiplier and applies the high or low voltage calibration, based on the set range, according to **Equation 11-3**.
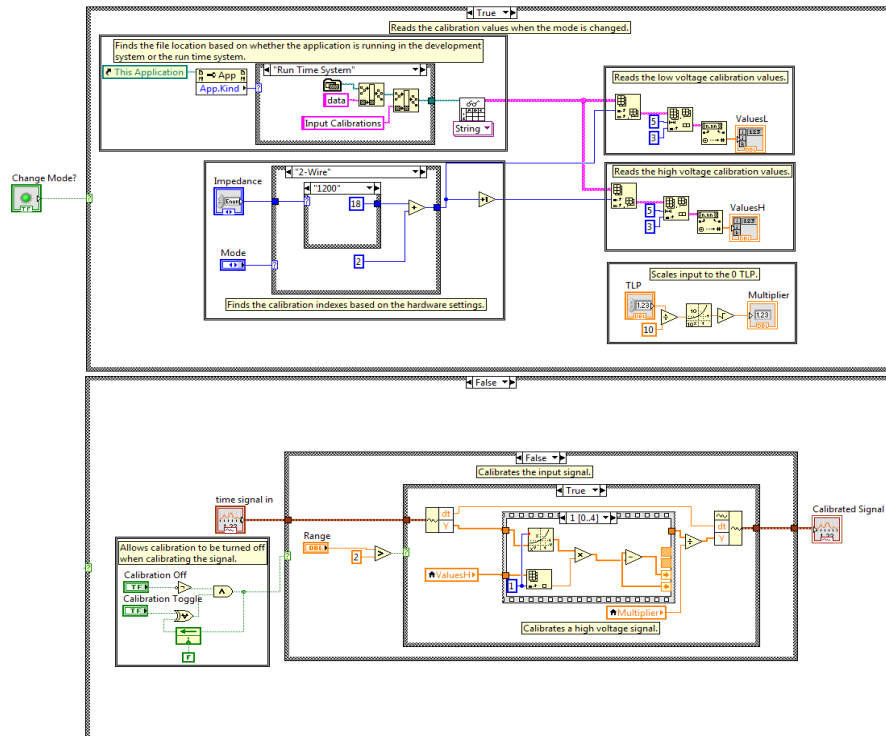
Figure 11-13: Calibrate Receive Block Diagram, True and False Cases.

$$f_c(t) = \left(f(t) - \left(c_0 + \sum_{n=1}^{4} f(t)^n c_n\right)\right)/M \qquad \text{(11-3)}$$

Where $f_c(t)$ is the calibrated signal, $f(t)$ is the un-calibrated signal, M is the TLP multiplier, and the c_n are the calibration constants.

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Queue Manager: This VI places new commands into a queue after a command has been executed, **Figure 11-14**. A previous command may enter a new command into the queue at the back or the front of the queue as necessary and can even flush the queue if necessary. The Error command is entered if any error are detected and if the global stop is active the Exit command will be queued.
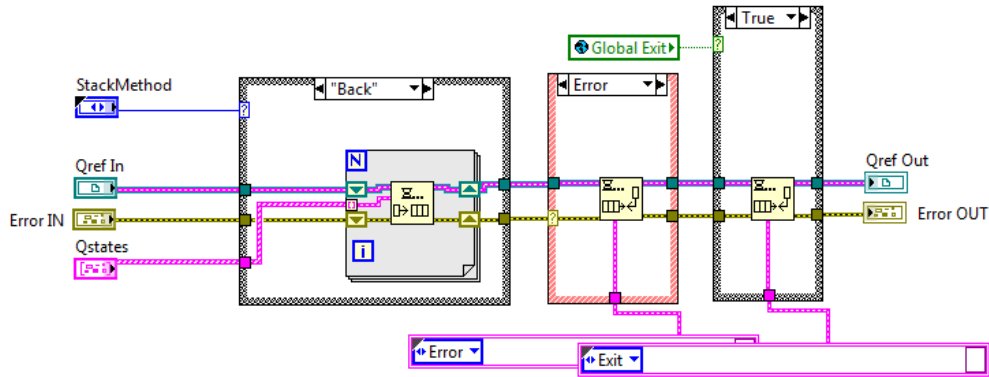

Figure 11-14: Queue Manager Block Diagram

Convert volts to dBm: This VI converts the peak amplitude of a sinusoidal signal to power, **Figure 11-15**, in dBm using the impedance setting of the hardware according to **Equation 11-4**.
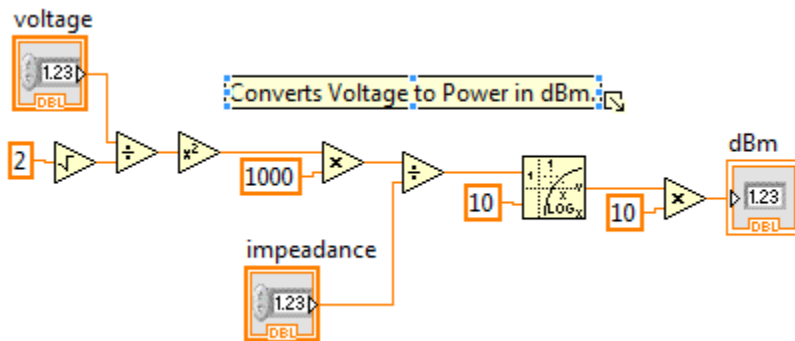
Figure 11-15: Convert volts to dBm Block Diagram

$$P = 10Log(\frac{1000\left(\frac{V_{Peak}}{\sqrt{2}}\right)^2}{I}) \tag{11-4}$$

Where P is the power of the tone in dBm, V_{Peak} is the peak amplitude of the tone in volts, and I is the impedance in ohms.

Convert dBm to volts: This VI converts the power of a sinusoidal signal from dBm to peak amplitude, **Figure 11-16**, using the impedance setting of the hardware according to **Equation 11-5**.
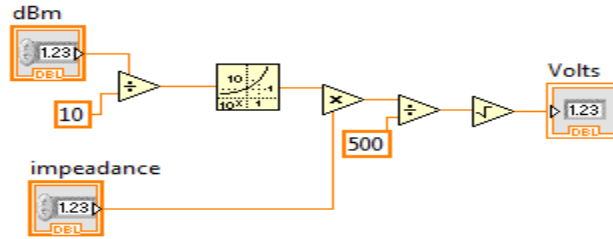
Figure 11-16: Convert dBm to volts Block Diagram

$$V_{Peak} = \sqrt{I10^{P/10}/500} \qquad (11\text{-}5)$$

Where P is the power of the tone in dBm, V_{Peak} is the peak amplitude of the tone in volts, and I is the impedance in ohms.

DAQ Analog Output Manager: The 'DAQ Analog Output Manager' VI is a Sub-VI that was developed to handle the creation and deletion of tasks for the DAQ's analog output and was built with the potential to handle multiple channels at once, **Figure 11-17**. When the VI is called to create a channel, the VI will check if the device is in use. If the device is already in use, the VI will check if the requesting VI is the VI using the device. Multiple VIs cannot communicate with the same device at the same time so the DAQ Manager will reject the request if the device is being used by another VI. If the device is available all of the analog output channels of the device are stored in the 'Channels' variable and the requesting VI's name is stored in the 'VIs' variable to indicate that the channels are being used by the VI. Once the channel has been assigned or an assigned channel needs updated, the DAQ manager clears any tasks previously assigned to the channel and uses the device name, voltage range, the generation mode, and the input signal's sample rate to generate a new task. When the VI the channel is assigned to indicates that it is done with the channel, the DAQ manager clears all channels assigned to the VI.
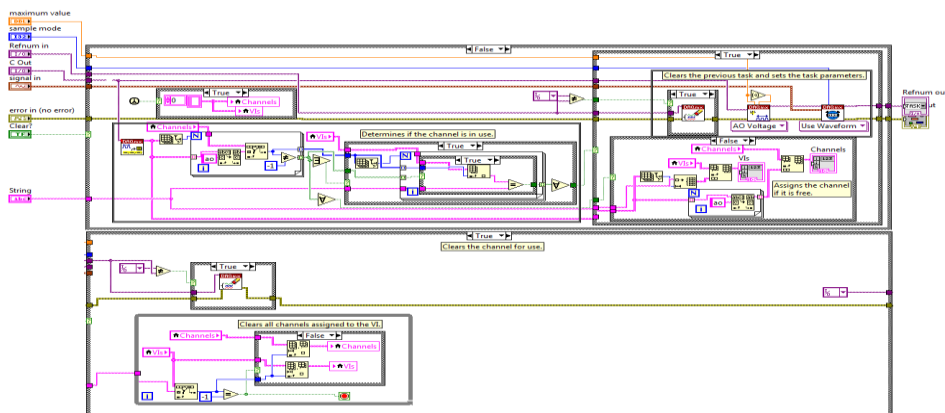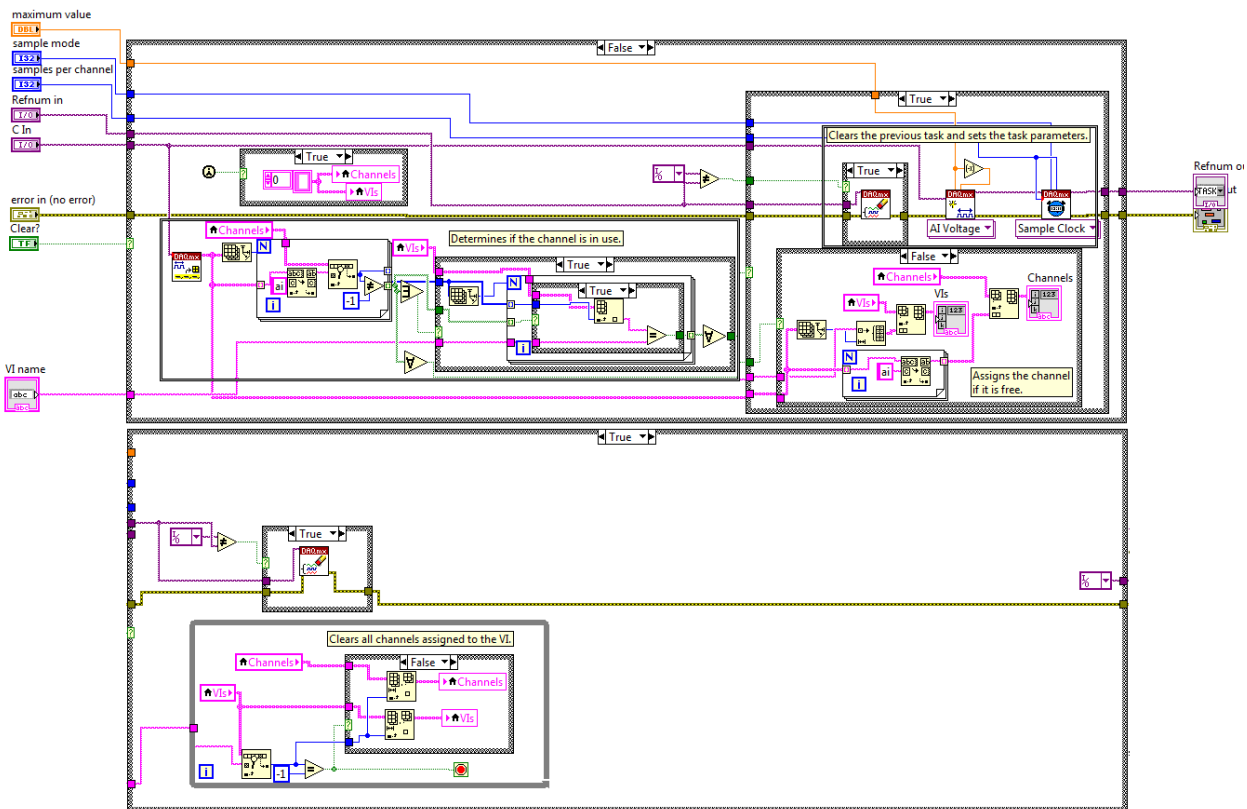

Figure 11-17: DAQ Analog Output Manager Block Diagram

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DAQ Analog Input Manager: The 'DAQ Analog Input Manager' VI is a Sub-VI that was developed to handle the creation and deletion of tasks for the DAQ's analog input and was built with the potential to handle multiple channels at once, **Figure 11-18**. When the VI is called to create a channel, the VI will check if the device is in use. If the device is already in use, the VI will check if the requesting VI is the VI using the device. Multiple VIs cannot communicate with the same device at the same time so the DAQ Manager will reject the request if the device is being used by another VI. If the device is available all of the analog output channels of the device are stored in the 'Channels' variable and the requesting VI's name is stored in the 'VIs' variable to indicate that the channels are being used by the VI. Once the channel has been assigned or an assigned channel needs updated, the DAQ manager clears any tasks previously assigned to the channel and uses the device name, voltage range, the generation mode, and the input signal's sample rate to generate a new task. When the VI the channel is assigned to indicates that it is done with the channel, the DAQ manager clears all channels assigned to the VI.

Figure 11-18: DAQ Analog Input Manager Block Diagram

Convert dB: The convert dB VI is used to convert one dBm measurement to another using the set hardware impedance, **Figure 11-19**. Many functions built into LabVIEW return power measurements for signals but assume an impedance of 1000 ohms. To correct the measurement values the results have to be scaled for the correct impedance using **Equation 11-6**.
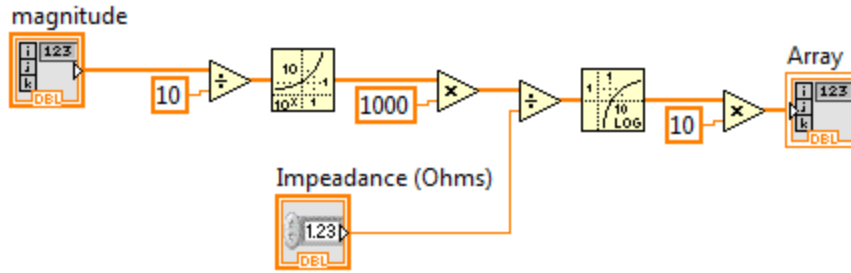
This VI converts dBm from the assumed impedance of 1000 ohms to the correct inpedance.

Figure 11-19: Convert dB Block Diagram

$$P_{Out} = 10Log(1000(10^{\frac{P_{In}}{10}})/I) \qquad (11\text{-}6)$$

Where P_{Out} is the corrected power, P_{In} is the input power, and I is the actual impedance.

11.2 Open First Block Diagram

The block diagram of the Open First VI is simply comprised of a 3 case event structure, **Figure 11-20**. These event cases handle the idle state, the 'setup complete' event, and the 'settings changed' event.
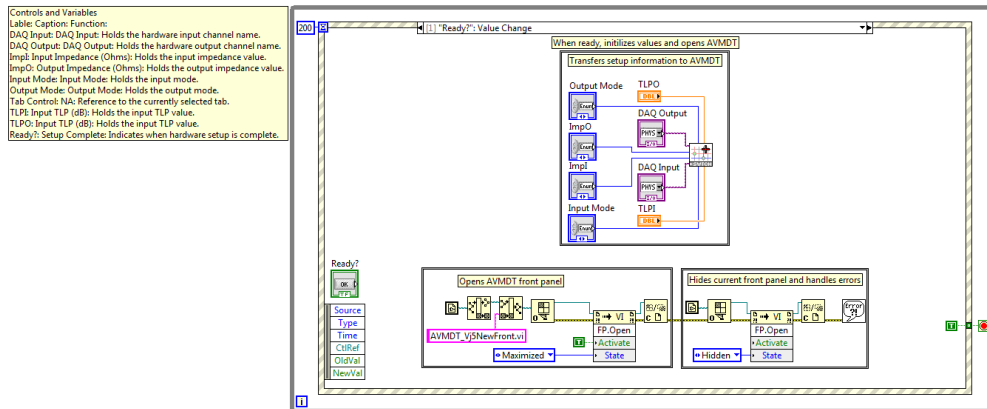
Figure 11-20: Open First Block Diagram

Idle State: This VI does nothing in the idle state.

Settings Changed: This event is activated whenever any of the hardware setting options are changed, **Figure 11-21**. The function of this event is to ensure that in 2-wire mode the settings of the input and output mode match and in bridged or bypass mode the impedance of the receive or transmit circuits respectively are set to 600 Ohms.
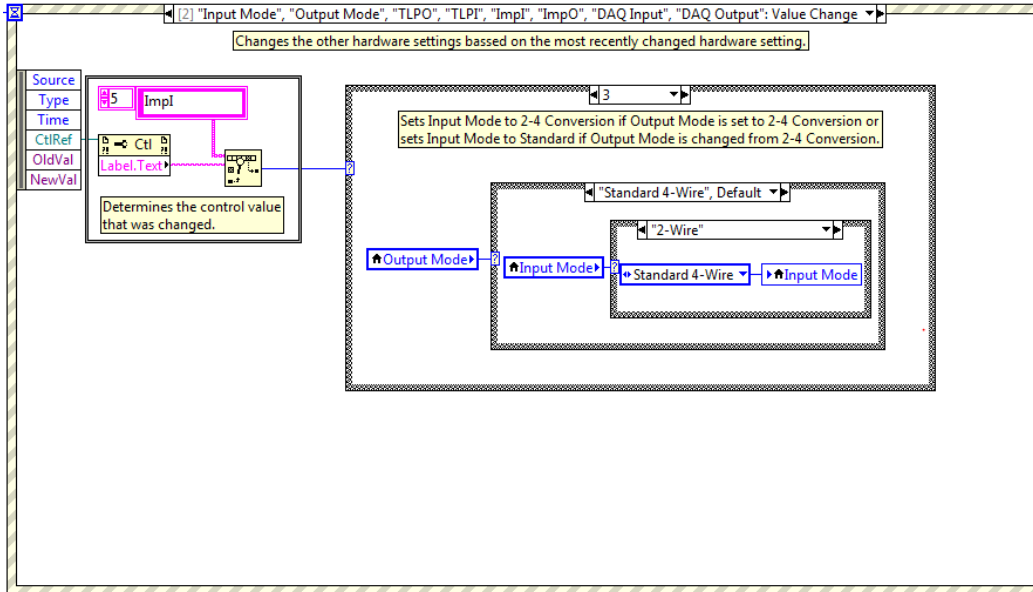
Figure 11-21: Open First Setting Change Event

Setup Complete: The setup complete event occurs when the 'Setup Complete' button is pressed, **Figure 11-22**. This event passes the hardware settings to the VMDT VI, closes the current VI, and opens the VMDT VI.
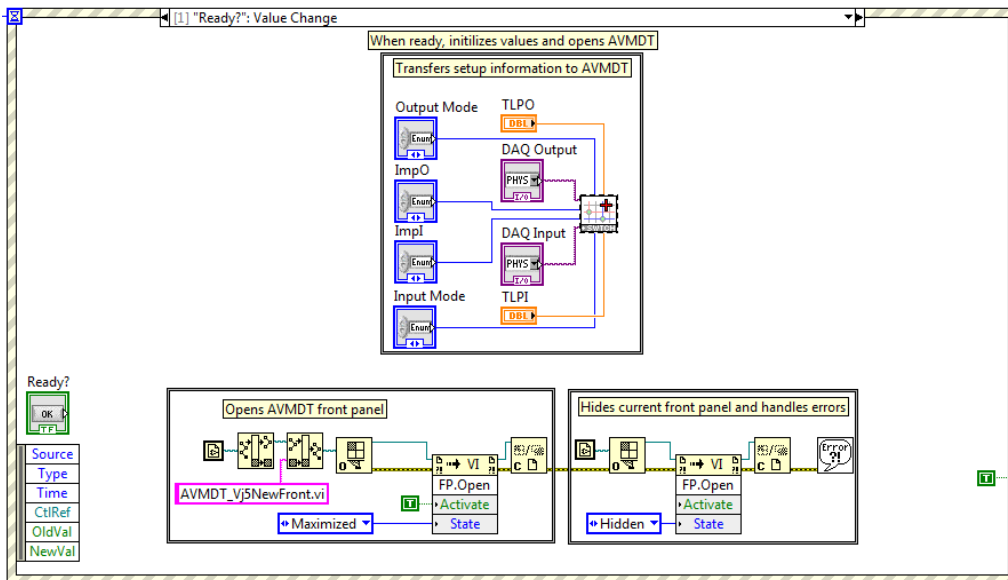

Figure 11-22: Open First Setup Complete Event

11.3 VMDT Block Diagram

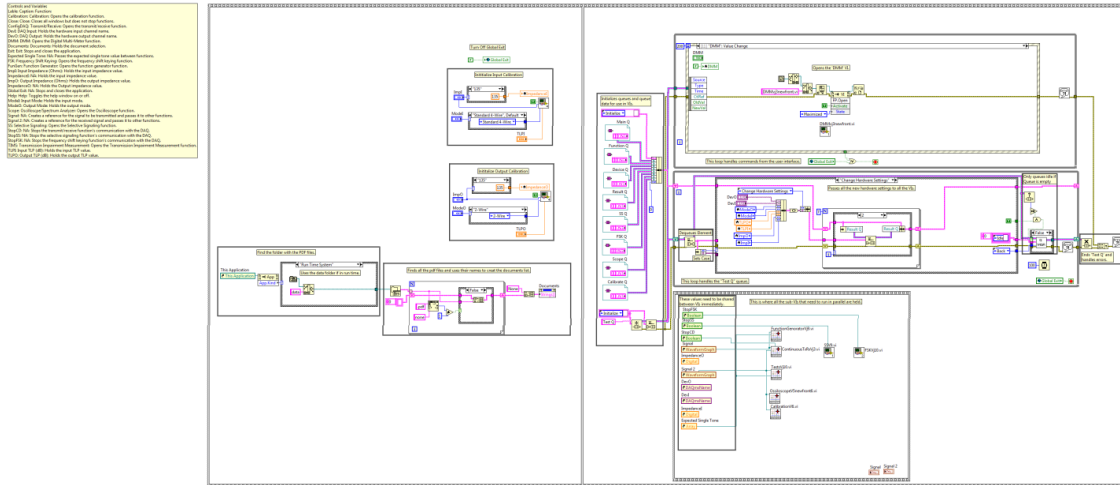The block diagram of the VMDT VI can be broken into four main sections, **Figure 11-23**. The initialize, event case, queue, and function sections.

Figure 11-23: VMDT Block Diagram

11.3.1 VMDT Initialize

The initialize section of the VMDT performs all the functions necessary to use the virtual test set before the user receives control of the virtual test set, **Figure 11-24**. These functions include setting the calibration settings for the hardware based on the values entered during the setup process, obtaining the list of documents the user can open from the virtual test set, and Initializing the queue used to communicate between VIs.
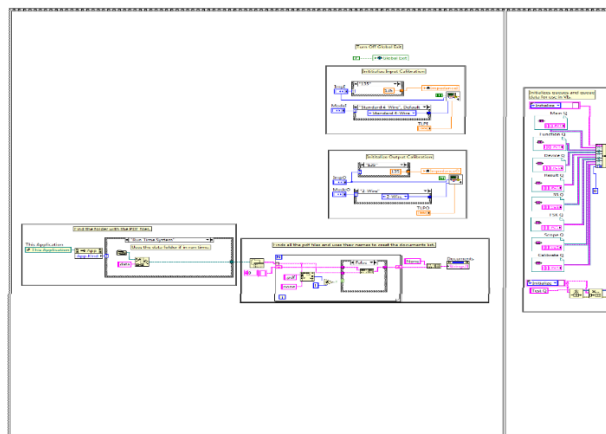

Figure 11-24: VMDT Block Diagram, Initialize

Initialize Calibration: Before any measurements can be made using the virtual test set it must be configured and calibrated for the desired hardware settings to ensure accurate results. When the

VMDT VI is first called it is configured for the settings used during the initial setup by passing the values to the Input and Output Calibrator VIs, **Figure 11-25**.
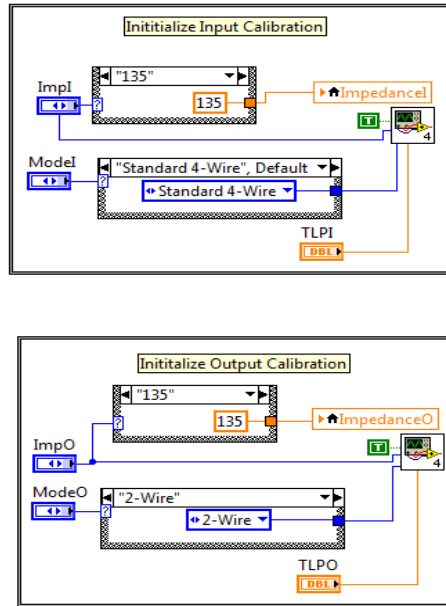

Figure 11-25: Initial Calibration

Obtaining the Document List: When the VMDT VI is first called it determines the files it has access to and builds a list for the user to select from, **Figure 11-26**. The virtual test set first locates the location of the folder containing the desired documents, this must be done differently depending on whether the virtual test set is running in development or run-time mode. The virtual test set then reads the name of every file at that location and builds a list from every file ending in .pdf. The option to select none is always added to the top of the list.
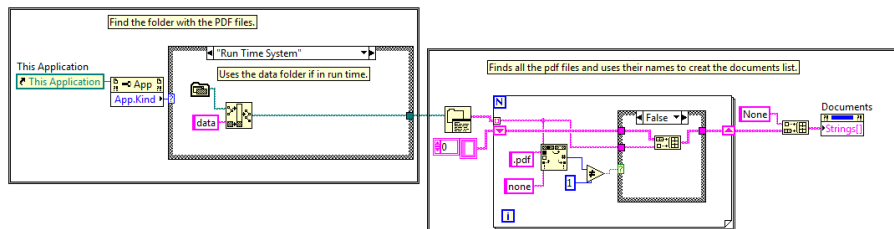

Figure 11-26: VMDT Block Diagram, Creating the Document list.

Initialize Queue References: Queue references are vital to the Queued State Machine that the virtual test set uses. Queue references allow multiple VIs to communicate using shared queue events. Two conditions must be met for VIs to share a queue. First, the VI where the queue is initiated must create the queue with a unique name to be referenced, **Figure 11-27**. Second, all VI that communicate with that queue must have a reference to the queue. The VMDT VI sets up its own 'Test Q' with the

Initialize command queued and passes all the references to the queues it communicates to with its queue loop to begin sharing commands with its sub-VIs.

Figure 11-27: VMDT Block Diagram, 'Test Q' and Queue Reference Initialization.

11.3.2 VMDT Event Case

The VMDT VI's event case handles the commands from the user. The primary function of the VMDT VI's event cases are opening the virtual test set's functions, resetting the virtual test set's hardware settings, and changing the information displayed in the help VI. The VMDT VI has the following unique event cases:

1. Exit
2. Open the selected document.
3. Close the virtual test set.
4. Open the Function Generator function.
5. Open the Transmission Impairment Measurement function.
6. Open the Selective Signaling function.
7. Toggle the help window.
8. Open the Frequency Shift Keying function.
9. Opens the Digital Multi-Meter function.
10. Opens the Oscilloscope/Spectrum Analyzer function.
11. Opens the Calibration function.
12. Reset the hardware settings.

Exit the virtual test set: This event case is triggered when the Exit button on the front panel is pressed, **Figure 11-28**. First, the Global Exit is set to true and the 'Exit' command is queue into the 'Test Que.' Second, a stacked sequence structure is used to close the VMDT VI, the Setup VI, and the Help Window in order. In each frame of the sequence structure the location of the VI to be closed is found using the path to the current VI. A reference to the VI is then called to check if the front panel is closed and, if it is not, it is then closed. This event closes all of the application windows and commands the virtual test set to stop all functions.
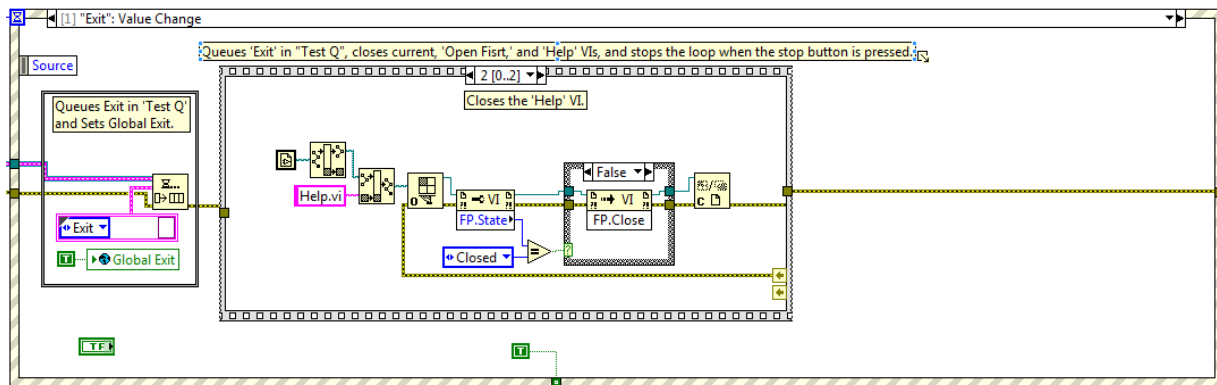
Figure 11-28: VMDT Block Diagram, Exit Event Case.

Open the selected document: This event is triggered whenever the user makes a selection from the 'Documents' menu, **Figure 11-29**. If the user does not select 'None' then the selected document is found and opened using the 'Open PDF' VI. If the application is in the Run Time mode the document is located in the 'data' folder. If the application is in development mode the document is located in the documents folder. This event case opens the selected document.
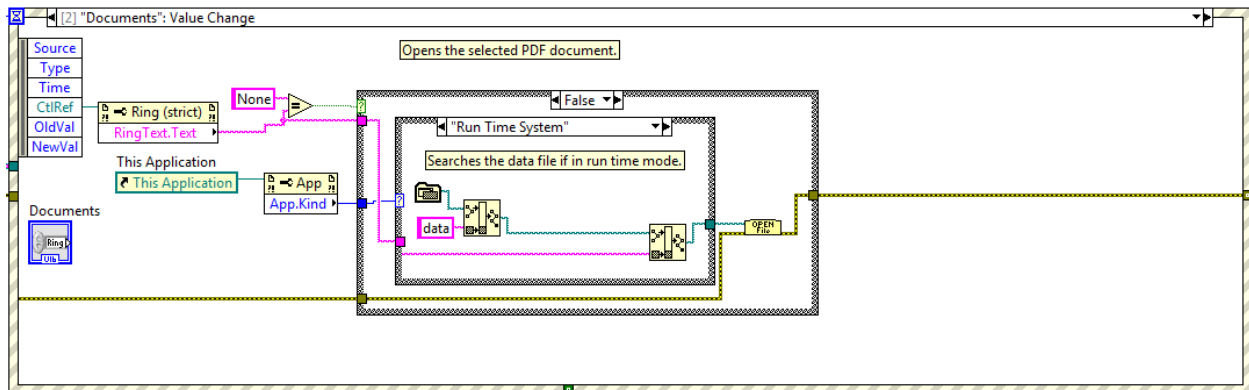
Figure 11-29: VMDT Block Diagram, 'Open the selected document' Event Case.

Close the virtual test set: This event case is triggered when the Close button on the front panel is pressed, **Figure 11-30**. First, the 'Close Test Suite' command is queue into the 'Test Que.' Second, a stacked sequence structure is used to close the VMDT VI and the Help Window in order. In each frame

of the sequence structure the location of the VI to be closed is found using the path to the current VI. A reference to the VI is then called to check if the front panel is closed and, if it is not, it is then closed. This event case closes all of the virtual test set's windows but does not command any functions to stop.
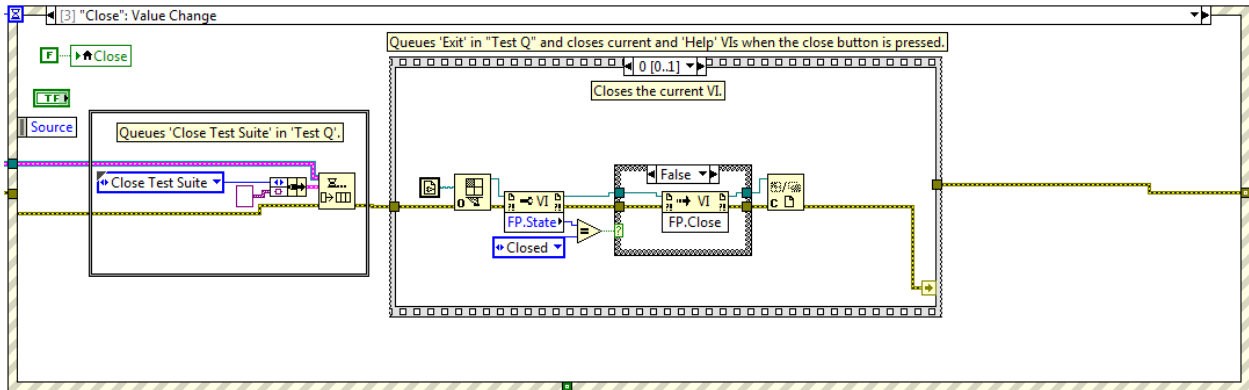

Figure 11-30: VMDT Block Diagram, 'Close the Virtual Test Set' Event Case.

Open a parallel function event: These events are triggered when the user selects a function from the main virtual test set menu that is capable of running at the same time as other functions. The corresponding VI is located from the same folder the VMDT VI is located in and a reference to the VI is opened to open and maximize the front panel of the VI, **Figure 11-31**. The event cases for the Function Generator, Transmission Impairment Measurement, Oscilloscope/Spectrum Analyzer, and Digital Multi-Meter functions are all functions that can run in parallel.
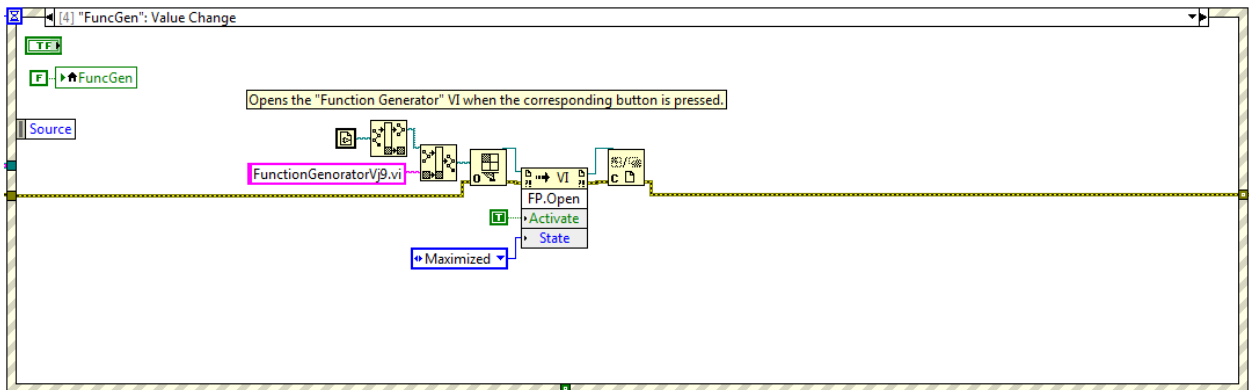
Figure 11-31: VMDT Block Diagram, 'Open the Function Generator function' Event Case.

Open non-parallel function: These events are triggered when the user selects a non-parallel from the main virtual test set menu, **Figure 11-32**. These functions cannot be used in parallel with other functions and gives the user a warning before proceeding. If the user chooses to proceed the conflicting VI's will be stopped. Then the corresponding VI is opened from the same folder the VMDT VI is located in, a reference to the VI is opened to open and maximize the front panel of the VI. Finally, the 'MasterTxRx' command is queued to stop all other communication with the DAQ. An example of the

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process is shown in **Figure 11-33**. If the user chooses not to continue, nothing is done. Selective Signaling and Frequency Shift Keying are non-parallel functions.

Figure 11-32: VMDT Block Diagram, 'Open the Selective Signaling function' Event Case.

Figure 11-33: 'Open the Selective Signaling function' case sequence structure.

Toggle the help window: This event is triggered whenever the help button is pressed. A path to the VI is used to create a control reference for help.vi, **Figure 11-34**. If the help window needs opened, the reference is used to open the help window. If the help window needs closed the reference checks if the help window is already closed and closes it if it is not.
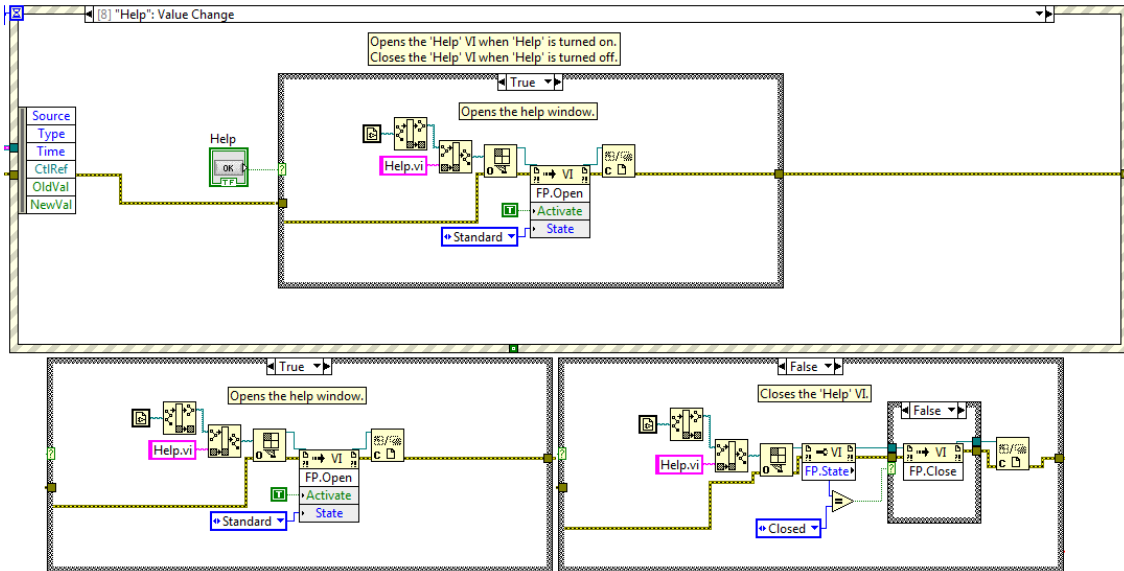
Figure 11-34: VMDT Block Diagram, 'Toggle help window' Event Case, with case structure true and false cases.

Open the Calibration function: This event is triggered when the user selects 'Calibration' from the main virtual test set menu, **Figure 11-35**. This function cannot be used in parallel with other functions and gives the user a warning before proceeding. If the user chooses to proceed the 'MasterTxRx' command is queued to stop all other communication with the DAQ and all other functions except function Generator are stopped, the 'Change Hardware Settings' command is called to set the transmission level points to zero, the Calibration VI is opened using an Invoke Node, and sets the Input Calibrator VI to calibration mode, **Figure 11-36**. If the user chooses not to continue, nothing is done. This event case opens the Calibration Function.
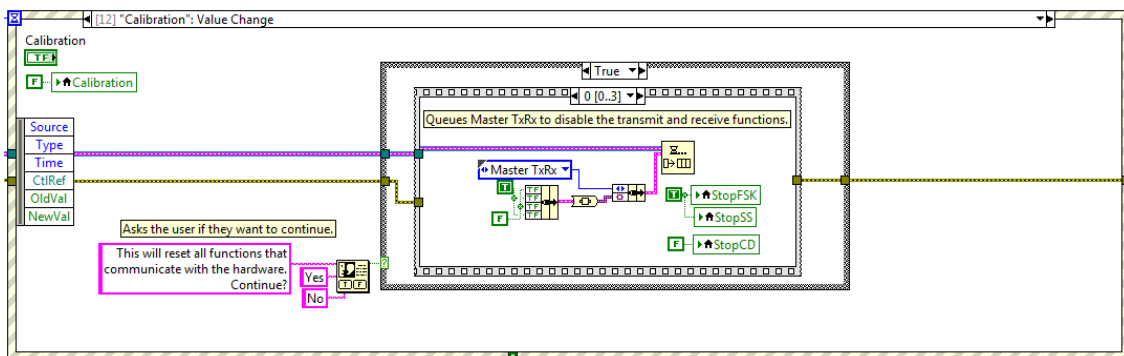
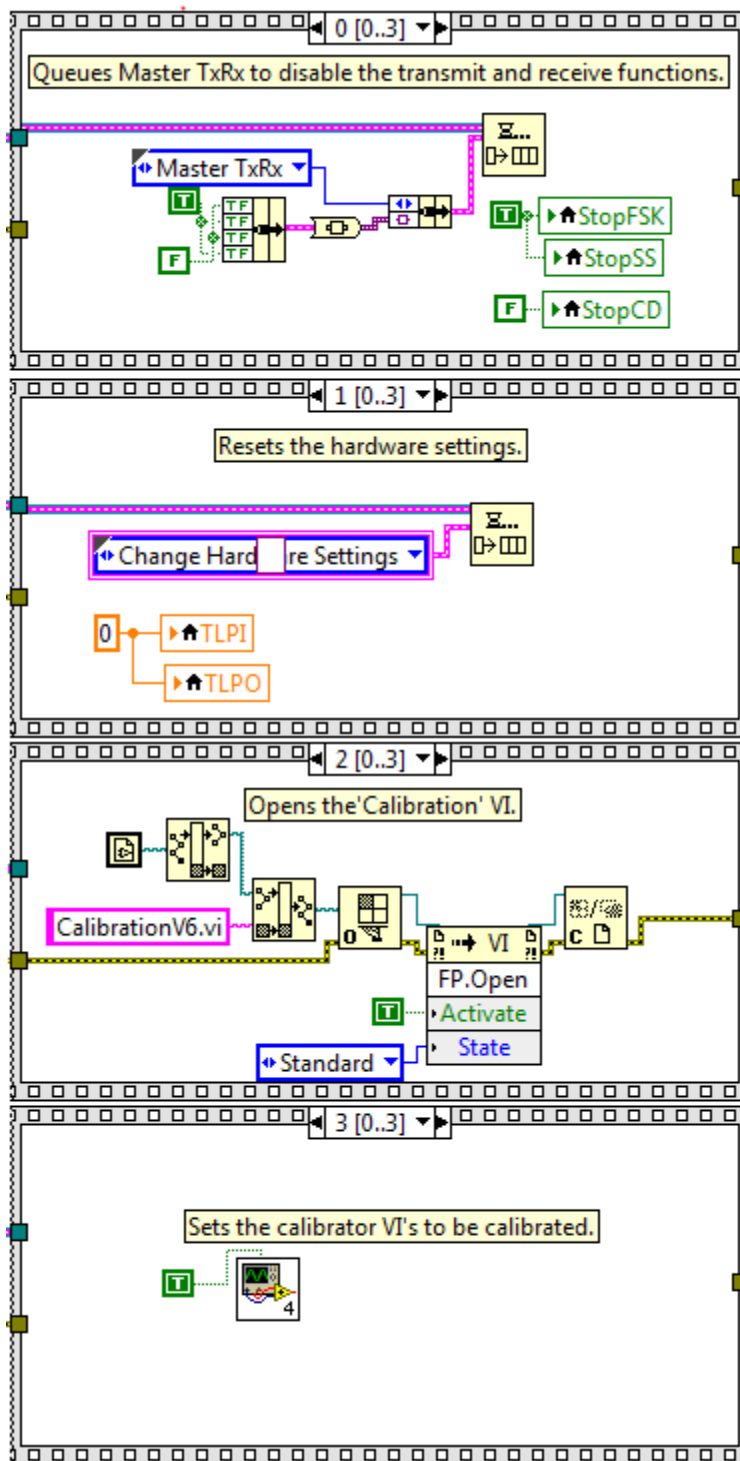**Figure 11-35: VMDT Block Diagram 'Open the Calibration Function' event case.**

Figure 11-36: 'Open the Calibration function' case sequence structure.

Reset Hardware Settings: This event is triggered when any of the hardware options are changed, **Figure 11-37**. The user is warned that changing the hardware settings will reset all communication with the hardware. If the user chooses to continue, a sequence structure is used to perform the multistep

process of resetting all hardware settings. If the user chooses not to continue, the changed setting is returned to its previous value. Both cases are shown in **Figure 11-38**. This event changes the hardware settings.
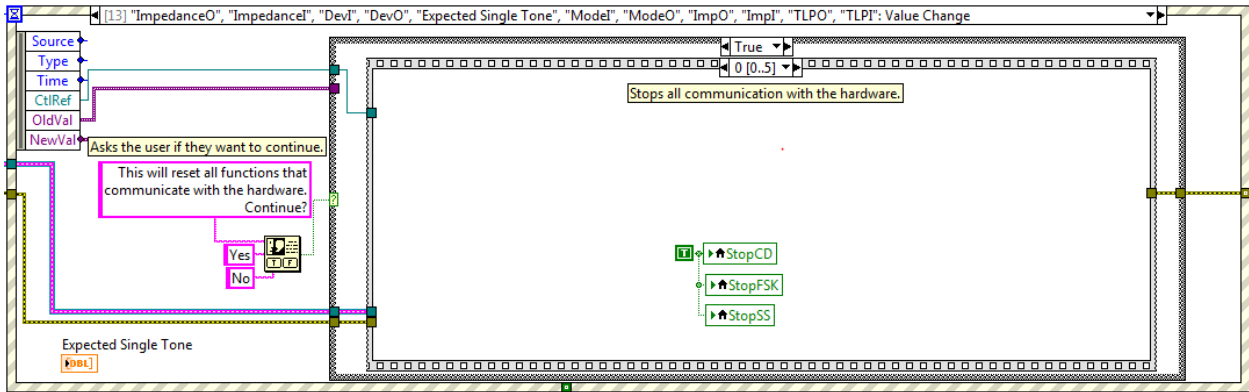
Figure 11-37: VMDT Block Diagram, 'Change Hardware Settings' event case.
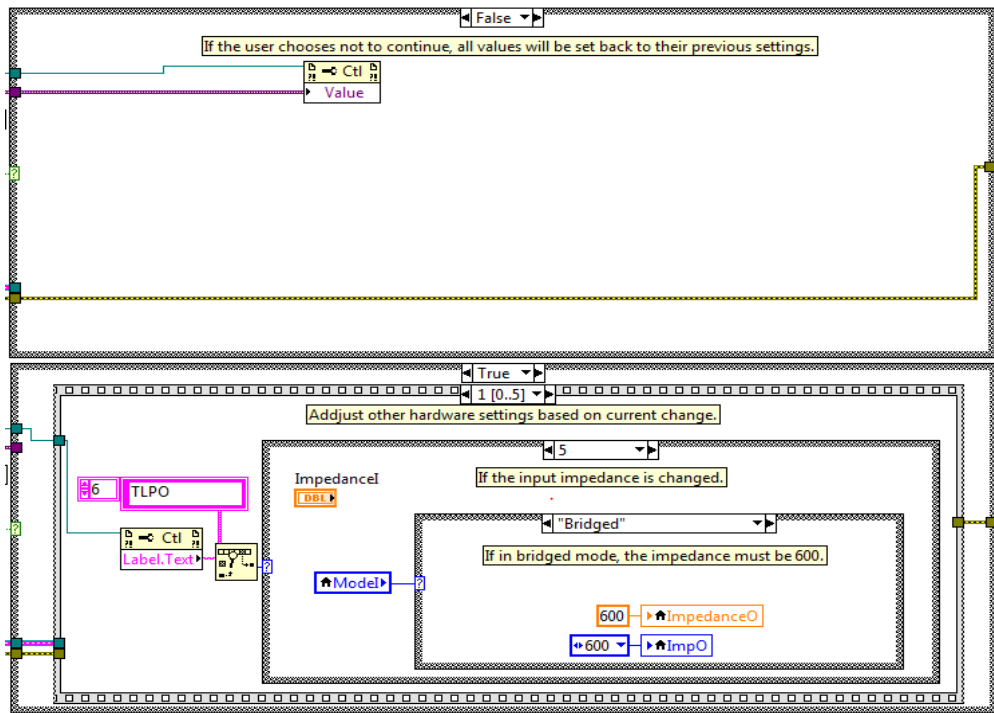
Figure 11-38: 'Change Hardware Settings function' case's case structure.

If the user chooses not to change the hardware settings the case structure of the event is set to false and a reference to the changed setting is return the setting to its previous value. However, it the user

chooses to change the hardware settings, the case is set to true and a sequence structure is used to execute the following steps:

1. All communication with the hardware is stopped, **Figure 11-39**.
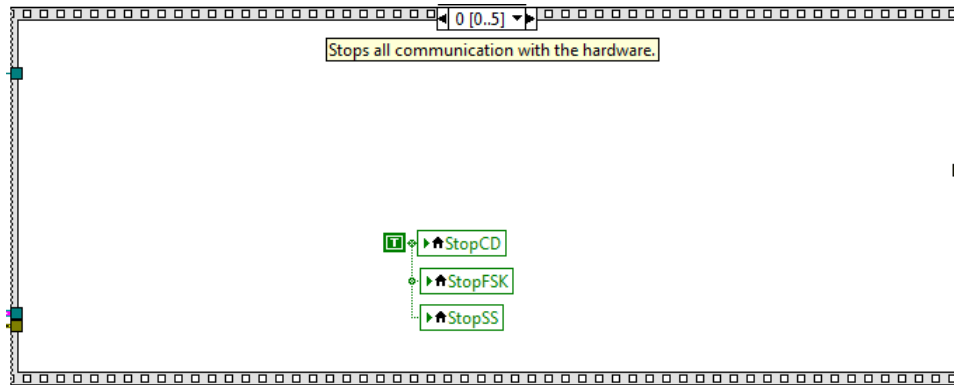
Figure 11-39: 'Change Hardware Setting Function' sequence structure frame 0.

2. Ensures that in 2-wire mode all settings are the same for the input and output, **Figure 11-40**. This is done using a series of cases selected based on the setting that was changed. Nothing is done if the input or output device is changed. If the input or output mode is changed to 2-Wire mode, the input and output settings are made the same. If the input is changed to Bridged or the output is changed to bypass the corresponding impedance is set to 600 Ohms. If the input or output were changed from 2-wire mode to standard 4-wire mode both settings are set to standard 4-wire mode. If the input or output impedance is changed in 2-wire mode both impedances are set to the same setting while in bridged or bypass modes the impedance change is rejected. If either TLP is changed, they are both changed in 2-wire mode.
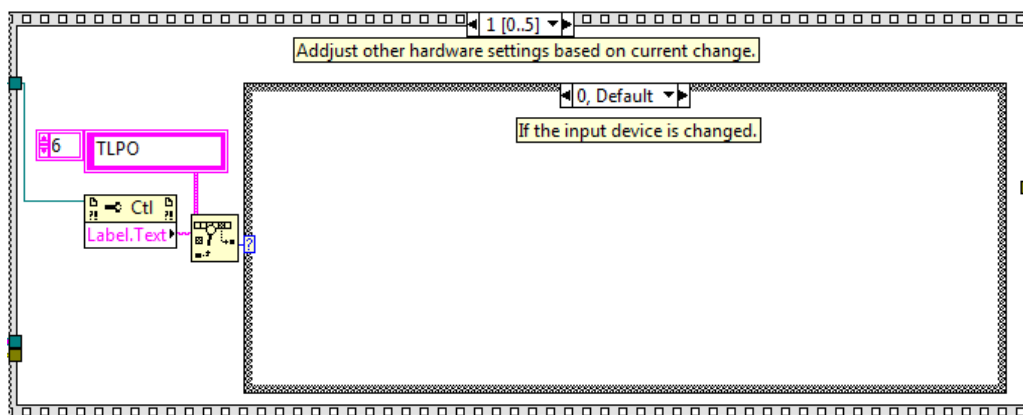

Figure 11-40: 'Change Hardware Setting Function' sequence structure frame 1.

3. The output calibration settings are changed using the Output Calibration VI, **Figure 11-41**.
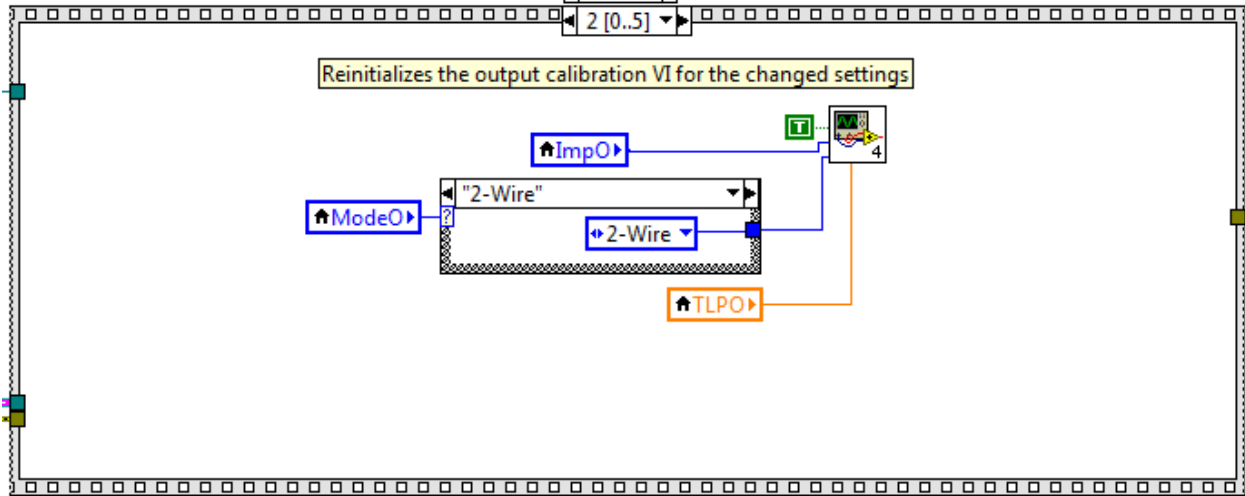

Figure 11-41: 'Change Hardware Setting Function' sequence structure frame 2.

4. The input calibration settings are changed using the Input Calibration VI, **Figure 11-42**.
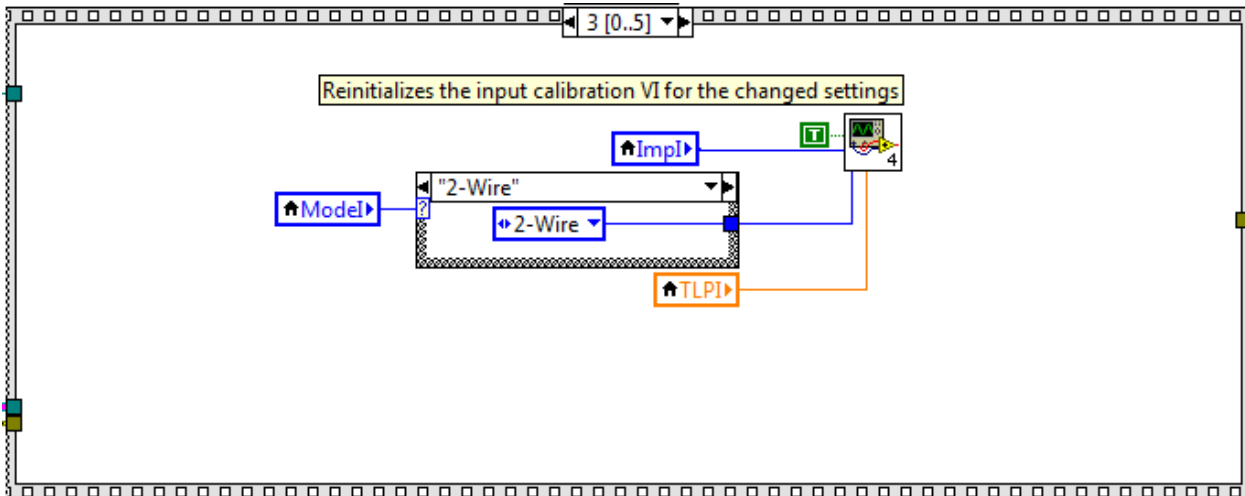

Figure 11-42: 'Change Hardware Setting Function' sequence structure frame 3.

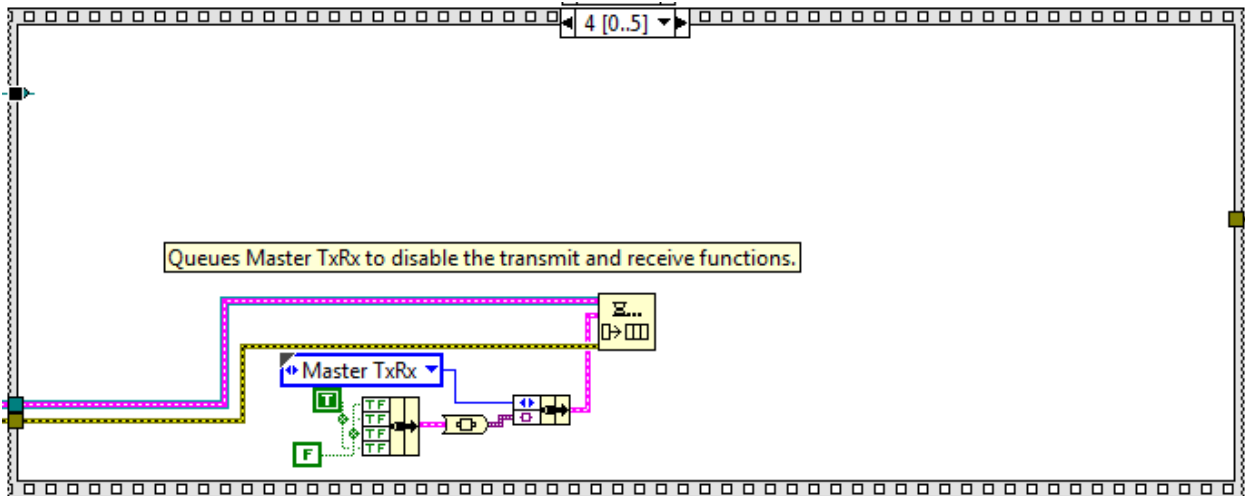5. The command 'Master TxRx' is queued to stop all communication with the hardware, **Figure 11-43**.

Figure 11-43: 'Change Hardware Setting Function' sequence structure frame 4.

6. The command 'Change Hardware Settings' is queued to inform the other functions of the hardware change, **Figure 11-44**.
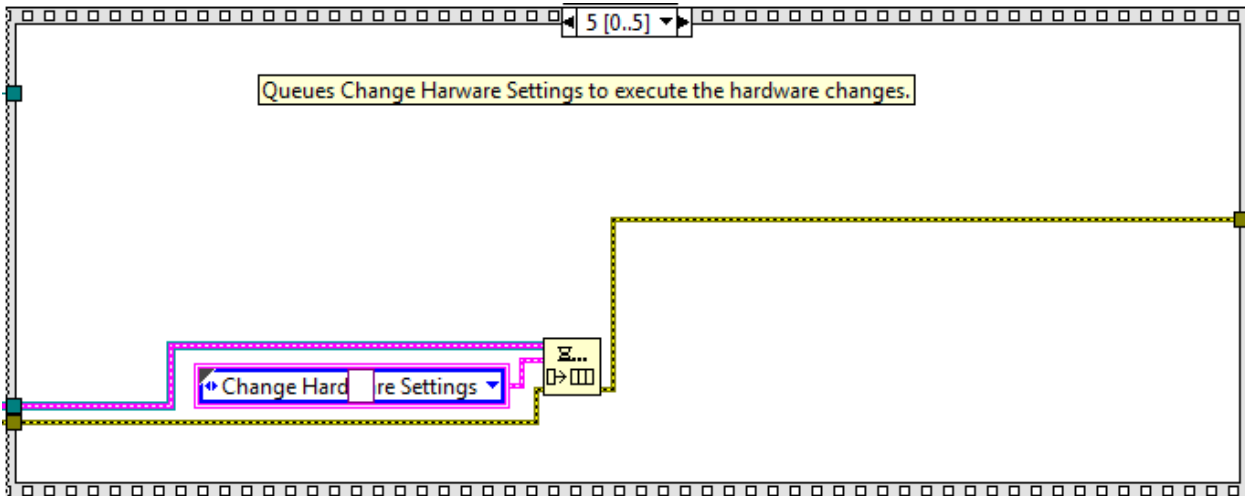

Figure 11-44: 'Change Hardware Setting Function' sequence structure frame 4.

11.3.3 VMDT Queue Loop

The primary function of the VMDT VI's Queue Loop is to send commands to and execute commands from other functions through the use of shared Queues. Like the event case structure, different actions are taken based on the different commands that are programed into the case structure. The only difference with the Queue Loop is the commands are initiated by other functions instead of the user. The VMDT VI's Queue Loop processes the following unique commands:

1. Impulse Testing
2. Exit

Impulse Testing: This command is queued when an impulse test is being performed. This command only passes the command to the 'Device Q' so it can be executed in the Transmit and Receive VI, **Figure 11-45**.
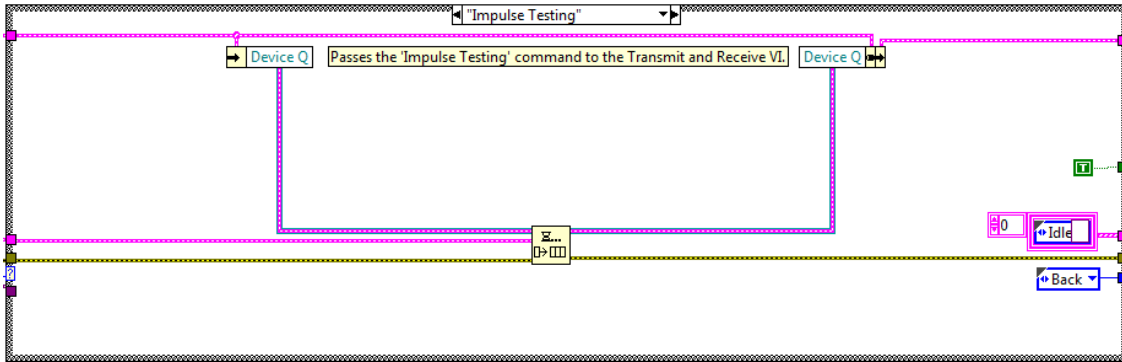

Figure 11-45: VMDT Block Diagram, 'Impulse Testing' Queue Command.

Exit: This command is queued when the virtual test set is shut down. This event queues the exit command, **Figure 11-46**.
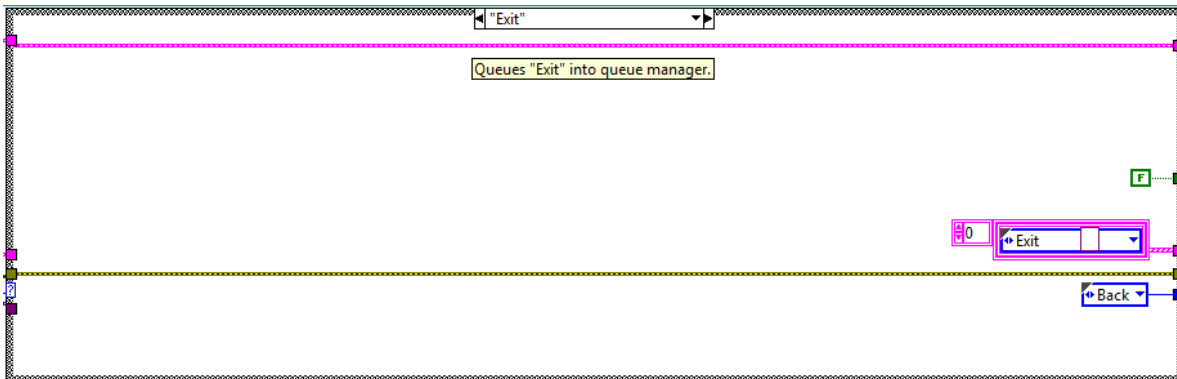

Figure 11-46: VMDT Block Diagram, 'Exit' Queue Command.

11.3.4 VMDT Functions

The VMDT VI's functions area holds all of the VIs that continuously run in parallel with the VMDT VI, **Figure 11-47**. The functions are located in a sequence structure that contains a single frame to ensure the functions only run once and prevent the virtual test set from closing until all of the functions have stopped. When the virtual test set is first activated control references are shared between the necessary sub-VIs so control values can be immediately passed between VIs. For example, the Signal 2 reference ensures the received signal graph is immediately shared with the Continuous TxRx, Test, Oscilloscope, and Calibration VIs to display the received signal in real time. Once the references have been passed to the sub-VIs they continue to run in the background until the virtual test set is shut down.
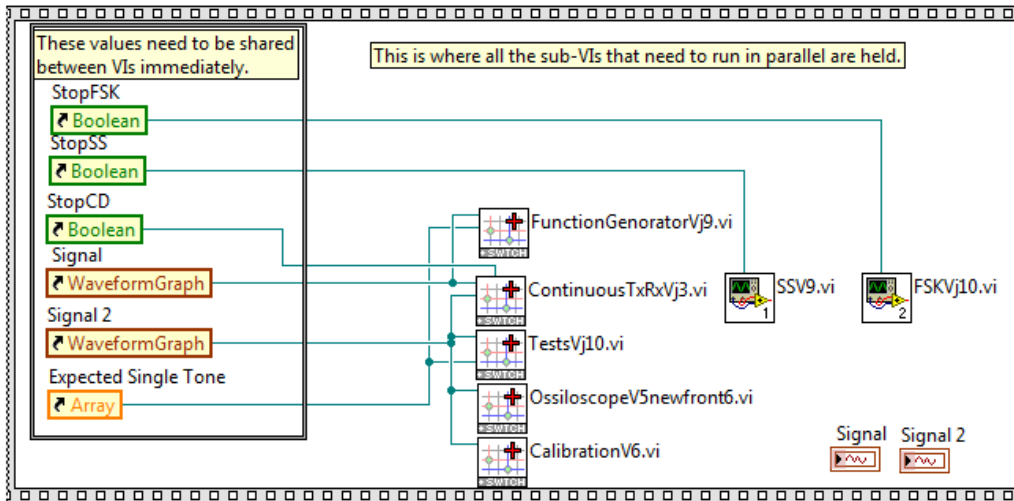
Figure 11-47: Virtual Test Set Functions.

11.4 Continuous Transmit and Receive

Continuous Transmit and Receive is responsible for the communication with the MyDAQ. This VI receives the transmission signal from the Function Generator VI and communicates with the MyDAQ to transmit the signal. Similarly, This VI communicates with the MyDAQ to receive a signal that is then passed to other functions for processing. This VI runs entirely in the background and the Front Panel is not used. Like the VMDT VI, the block diagram of Continuous Transmit and Receive can be broken into different sections: the initialize, event case, queue, transmit, and receive sections, **Figure 11-48**.
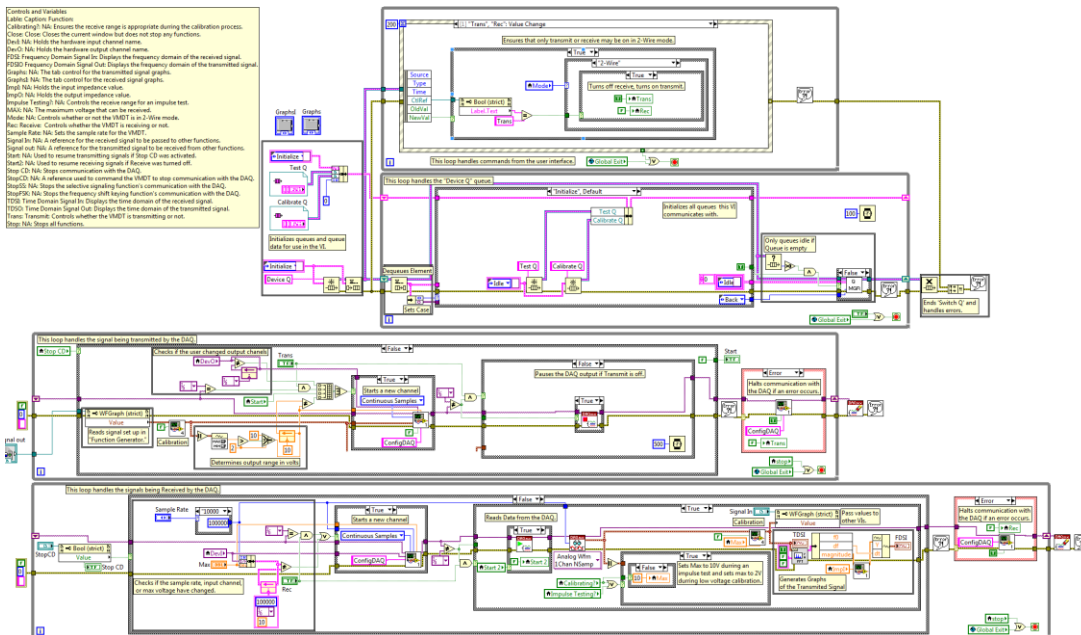

Figure 11-48: Continuous Transmit/Receive Block Diagram.

11.4.1 Continuous Transmit/Receive Initialize Section.

The initialize section of Continuous Transmit/Receive only initializes the queues that it communicates with, **Figure 11-49**. This is done by creating a cluster with Test Q and Calibrate Q references and creating its own Device Q. Test Q allows this sub-VI to send commands to the VMDT VI while the Calibrate Q allows it to send commands to the Calibration function. Device Q is the queue that holds the commands executed by Continuous Transmit/Receive and other VIs will need a queue reference to it in order to pass commands to this sub-VI.
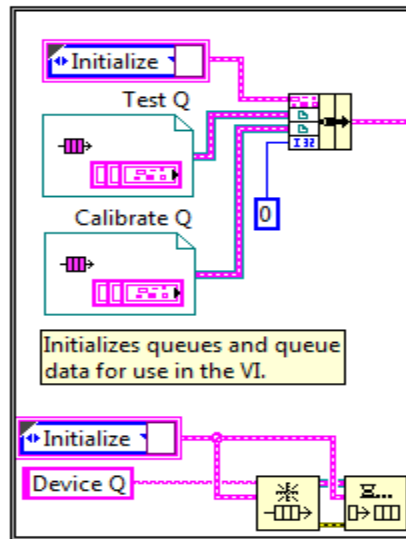

Figure 11-49: Continuous Transmit/Receive Initialize Section.

11.4.2 Continuous Transmit/Receive Event Case Section.

This section is only used for debugging and is not used while the virtual test set is running.

11.4.3 Continuous Transmit/Receive Queue Section.

The Continuous Transmit/Receive's Queue Loop handles all of the commands passed to the VI. The primary function of the Continuous Transmit/Receive's Queue Loop is to execute commands from other functions through the use of shared Queues. The Transmit/Receive's Queue Loop can process the following unique commands:

1. Impulse Testing
2. Calibration

Impulse Testing: This command turns 'Impulse Testing Mode' on or off, **Figure 11-50**. This allows the virtual test set to more accurately capture sharp impulses in the signal at the cost of less accurately reading low power signals. This command is called whenever the virtual test set is performing an Impulse Test with a high power threshold.
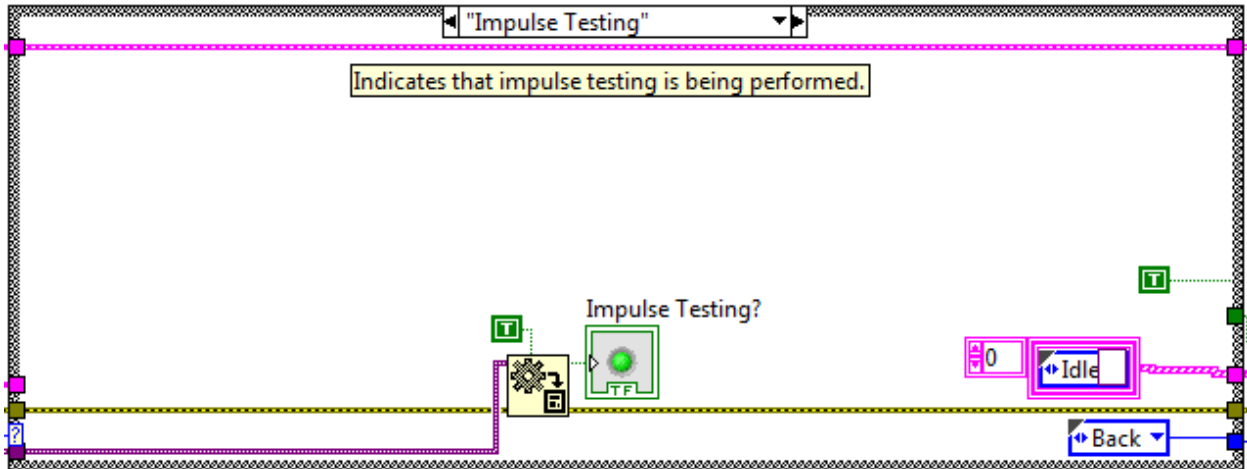
Figure 11-50: Continuous Transmit/Receive Queue, Impulse Testing Case.

Calibration: This command turns 'Calibration Mode' on or off, **Figure 11-51**. This allows the virtual test set to ensure that the 2 volt input range is used when calibrating low voltage input. This command is called whenever low voltage input calibration is performed.
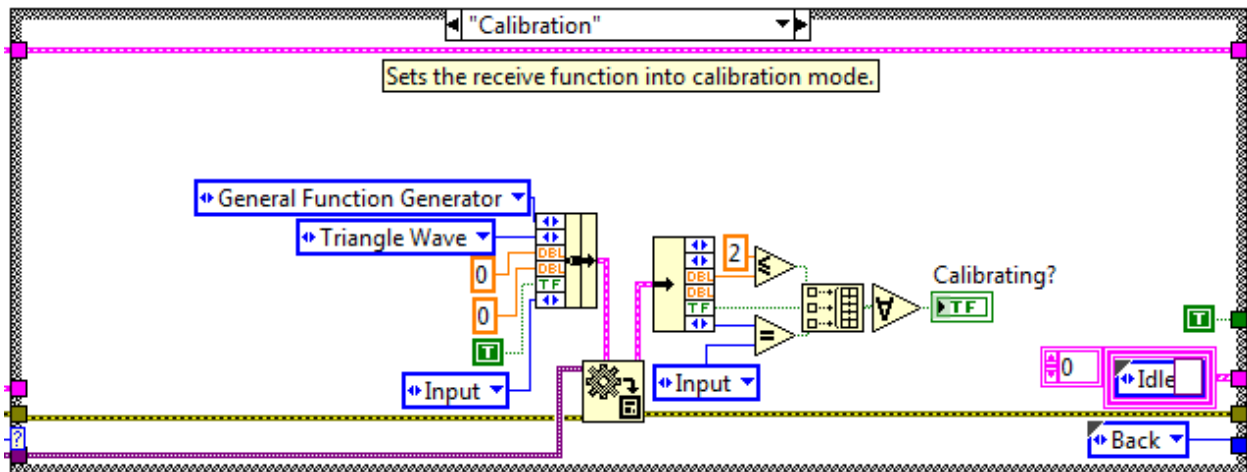

Figure 11-51: Continuous Transmit/Receive Queue, Calibration Case.

11.4.4 Continuous Transmit/Receive Transmit Section.

This section of Continuous Transmit/Receive transmits the signals passed to it through the Signal Out reference, **Figure 11-52**. The Transmit section performs its functions in four basic steps:

1. The signal is calibrated.
2. The hardware task is initialized.
3. The signal is transmitted.
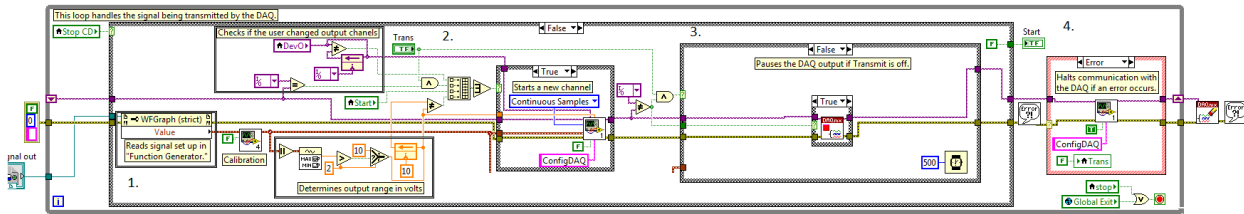4. Errors are handled.

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Figure 11-52: Transmit Receive Block Diagram, Transmit.

Signal Calibration: When the output signal is received, it is passed to the 'Output Calibration' VI to be calibrated for transmission, **Figure 11-53**. To calibrate the signal the transmission level multiplier is applied to the signal, the appropriate calibration range is chosen by approximating the maximum value after calibration, and the selected calibration values are applied sequentially. Once the signal is calibrated, the maximum value is determined to set the appropriate hardware setting.
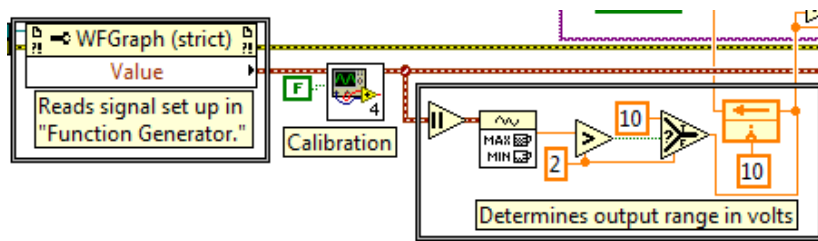

Figure 11-53: Transmit Receive Block Diagram, Transmit Calibration.

Initializing the hardware task: In order for the virtual test set to communicate with the hardware, a task must be assigned to the MyDAQ utilized by the virtual test set. A new task is generated if any of the following occur: there is no task, the output device is changed, another function was communicating with the MyDAQ, or the voltage range needs changed, **Figure 11-54**. If any of these conditions are met a new task is assigned to the MyDAQ using the 'Analog Output DAQ Manager' VI. However, if another function utilizes the DAQ the 'Analog Output DAQ Manager' VI is called to stop communication with the MyDAQ.
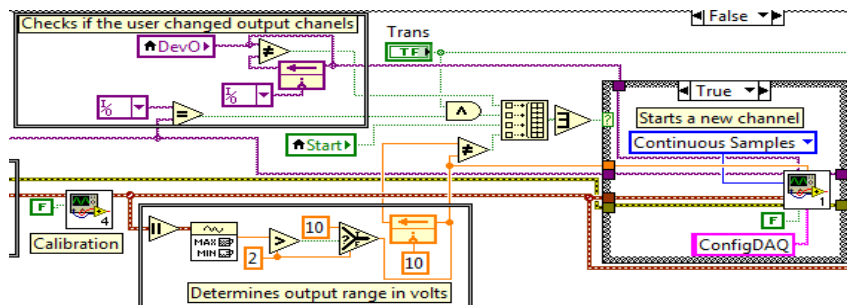

Figure 11-54: Transmit Receive Block Diagram, Transmit Initialize Hardware Task.

Transmit the signal: Once a task is created and the virtual test set is set to transmit, the calibrated signal is transmitted over the assigned channel, **Figure 11-55**. If the virtual test set's transmit is turned off the task is paused and waits for the virtual test set to be ready to transmit again. Signal graphs of the transmitted signal are generated but these are used for debugging purposes only.
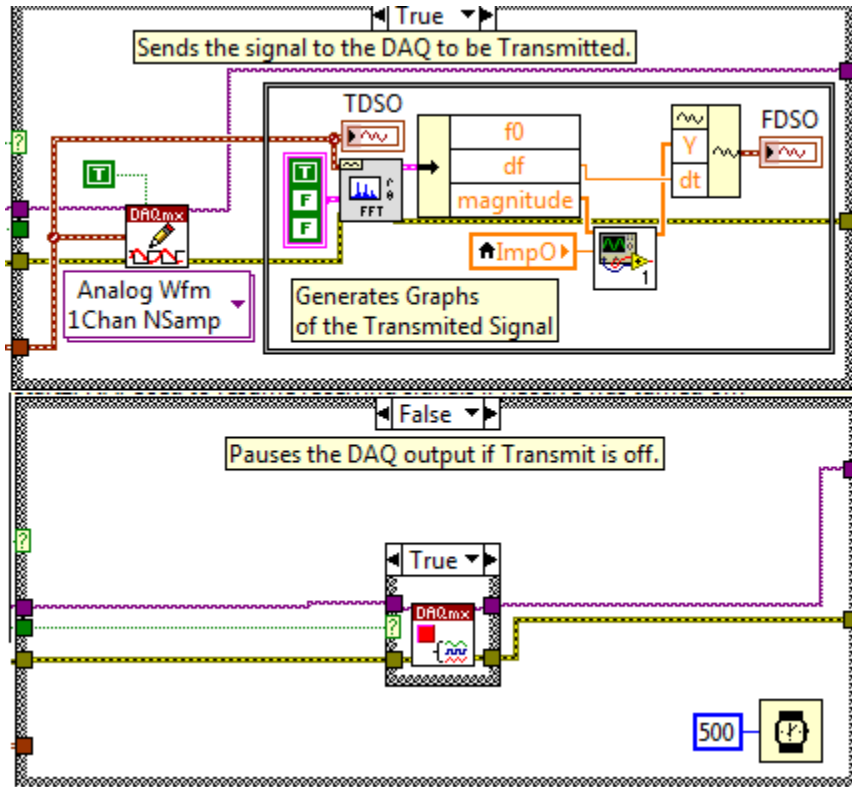

Figure 11-55: Transmit Receive Block Diagram, Signal Transmission.

Handle Errors: If any errors occur during transmission, a message will be displayed to the user and the 'Analog Output DAQ Manager' VI is called to clear the hardware channel, **Figure 11-56**.
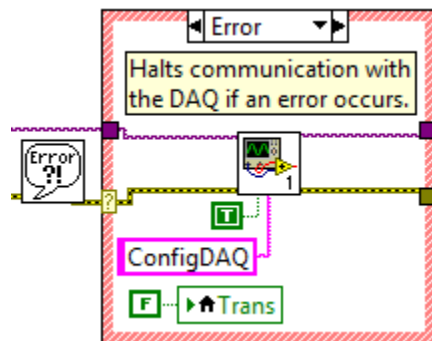

Figure 11-56: Transmit Receive Block Diagram, Transmission Error Handler.

11.4.5 Continuous Transmit/Receive Section.

This section of Continuous Transmit/Receive receives the signals and passes them to other VIs through the Signal In reference, **Figure 11-57**. The Receive section performs its functions in four basic steps:

1. The hardware task is initialized.
2. The signal is received and calibrated.
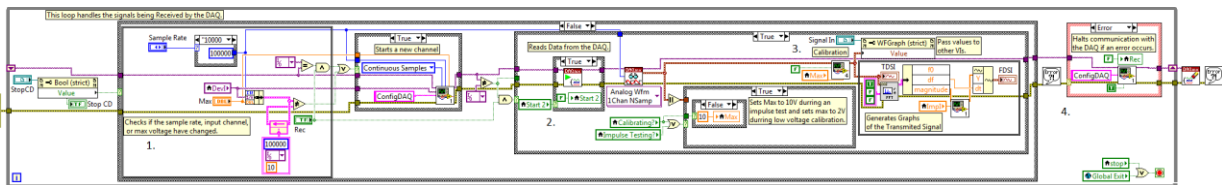3. Errors are handled.

Figure 11-57: Continuous Transmit/Receive Block Diagram, Receive.

Initializing the hardware task: In order for the virtual test set to communicate with the hardware, a task must be assigned to the MyDAQ utilized by the virtual test set. A new task is generated if any of the following occur: there is no task, the output device is changed, the sample rate is changed, or the voltage range needs changed, **Figure 11-58**. If any of these conditions are met a new task is assigned to the MyDAQ using the 'Analog Input DAQ Manager' VI. However, if another function utilizes the DAQ the 'Analog Input DAQ Manager' VI is called to stop communication with the MyDAQ.
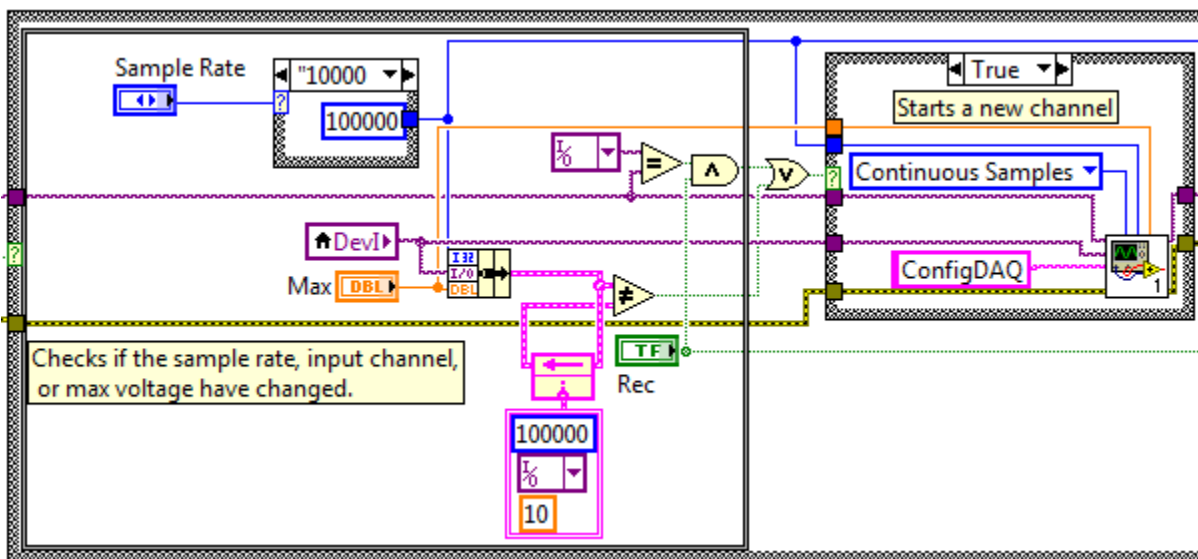

Figure 11-58: Transmit Receive Block Diagram, Transmit Initialize Hardware Task.

Receive and calibrate the signal: Once a task is created and the virtual test set is set to receive, the signal is read from the hardware and passed to the 'Input Calibrator' VI to be calibrated, **Figure 11-59**. Once the signal is calibrated, it is passed to other VIs to be processed. If the virtual test set's receive is turned off the task is paused and waits for the virtual test set to be ready to receive again. Signal graphs of the received signal are generated but these are used for debugging purposes only.
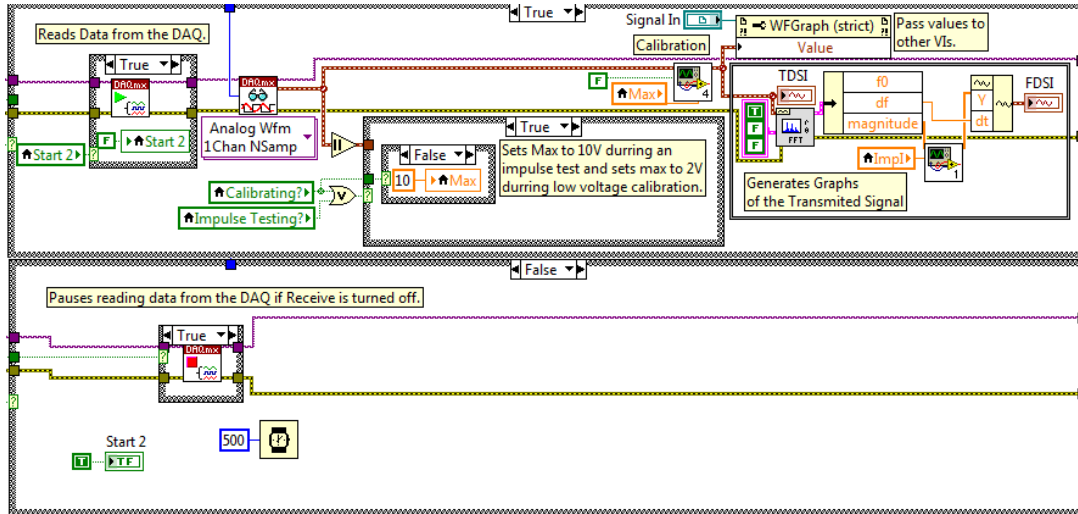

Figure 11-59: Continuous Transmit/Receive Block Diagram, Receive Signal.

Handle Errors: If any errors occur during transmission, a message will be displayed to the user and the 'Analog Input DAQ Manager' VI is called to clear the hardware channel, **Figure 11-60**.
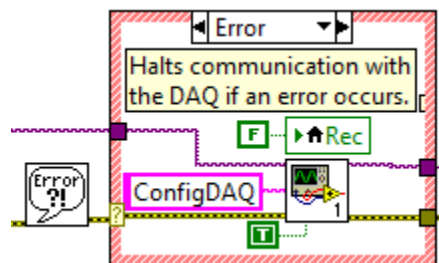
Figure 11-60: Continuous Transmit\Receive Block Diagram, Receive Error Handler.

11.5 Function Generator Block Diagram

The Function Generator VI is used to configure the continuous signal to be transmitted by the virtual test set. Once transmit is selected, a queue command is sent to the VMDT VI. The VMDT VI then sends a command to all other functions to indicate that a continuous signal is being transmitted and the Continuous Transmit and Receive VI begins transmitting the signal using a control reference for the signal graph. Like the VMDT VI, the block diagram of the Function generator can be broken into different sections: the initialize, event case, queue, and signal generation sections, **Figure 11-61**.
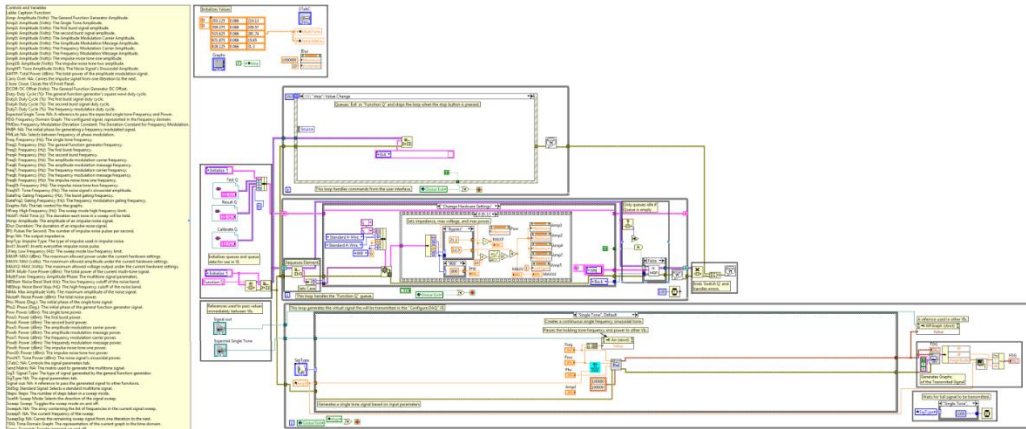
Figure 11-61: Function Generator Block Diagram

11.5.1 Function Generator Initialize section.

The initialize section of Function Generator initializes the queues and references it uses to communicate with other VIs. This is done by creating a cluster with Test Q, Result Q, and Calibrate Q references and creating its own Function Q, **Figure 11-62**. The Test Q, Result Q, and Calibrate Q allows this VI to send commands to other VIs while the Function Q is the queue that holds the commands executed by the Function Generator. Other VIs will need a queue reference to Function Q in order to pass commands to this sub-VI.
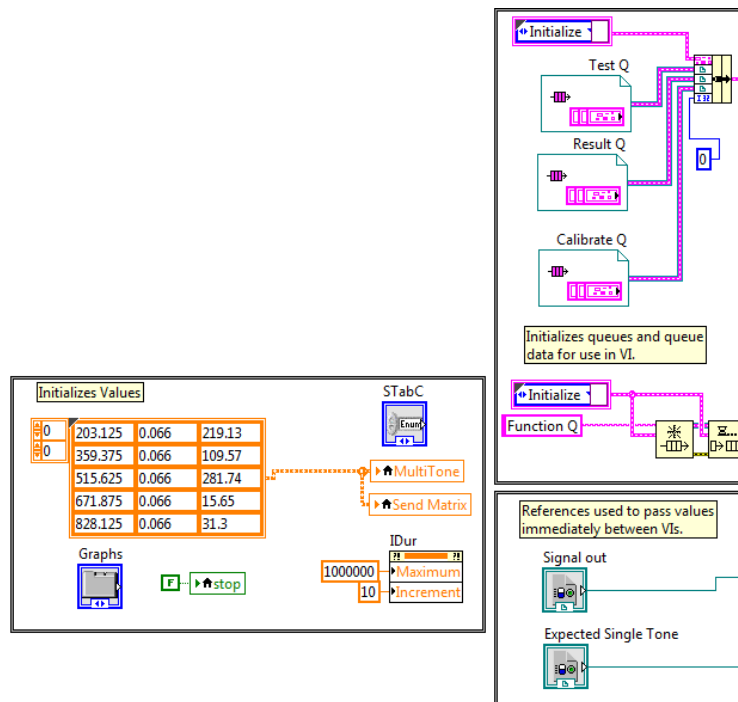

Figure 11-62: Function Generator, Initialize Sections

11.5.2 Function Generator Event Case Section.

The Function Generator's event case handles the commands from the user. The primary functions of the Function Generator's event cases are creating preset and custom signals while ensuring the signals do not exceed the limits of the virtual test set. The Function Generator has the following unique event cases:

1. Change General Function Generator Amplitude and DC offset.
2. Change Single Tone Generator Amplitude and Power.
3. Set Multi-Tone Signal to a Standard Signal.
4. Change Multi-Tone Signal.
5. Change Sweep Settings.
6. Tab Control.
7. Change Burst Amplitude and Power.
8. Change Amplitude Modulation Amplitude and Power.
9. Toggle Frequency and Phase Modulation.
10. Change Frequency Modulation Amplitude and Power.
11. Change Impulse Amplitude and Power.
12. Change Impulse Type.
13. Change Noise Amplitude and Power.

Change General Function Generator Amplitude and DC offset: This event activates whenever the amplitude or DC offset for the general function generator is changed. This event auto adjusts the amplitude or DC offset to prevent the maximum output amplitude from being exceeded, **Figure 11-63**.
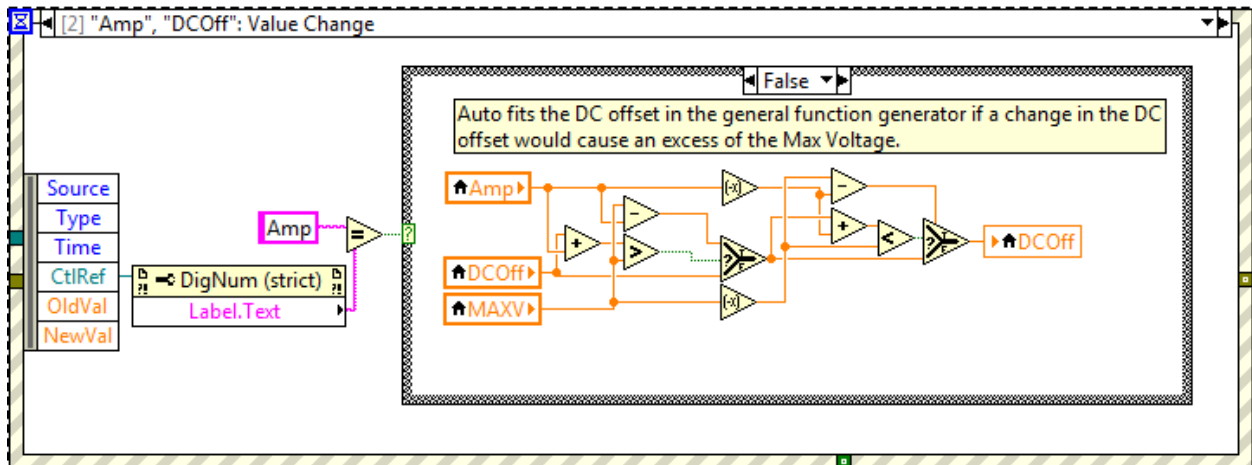
Figure 11-63: Function Generator Event Case, Change the General Function Generator Amplitude or DC Offset Event.

Change Single Tone Generator Amplitude and Power: This event activates whenever the amplitude or power of the Single Tone Generator is changed. This event converts amplitude to power and auto adjusts the amplitude or power to prevent the maximum output amplitude from being exceeded, **Figure 11-64**.

Figure 11-64: Function Generator Event Case, Change Single Tone Generator Amplitude and Power Event

Set Multi-Tone Signal to a Standard Signal: This event activates whenever a standard multi-tone signal is selected from the pull down menu. This event sets the multi-tone signal to a preset signal, **Figure 11-65**.
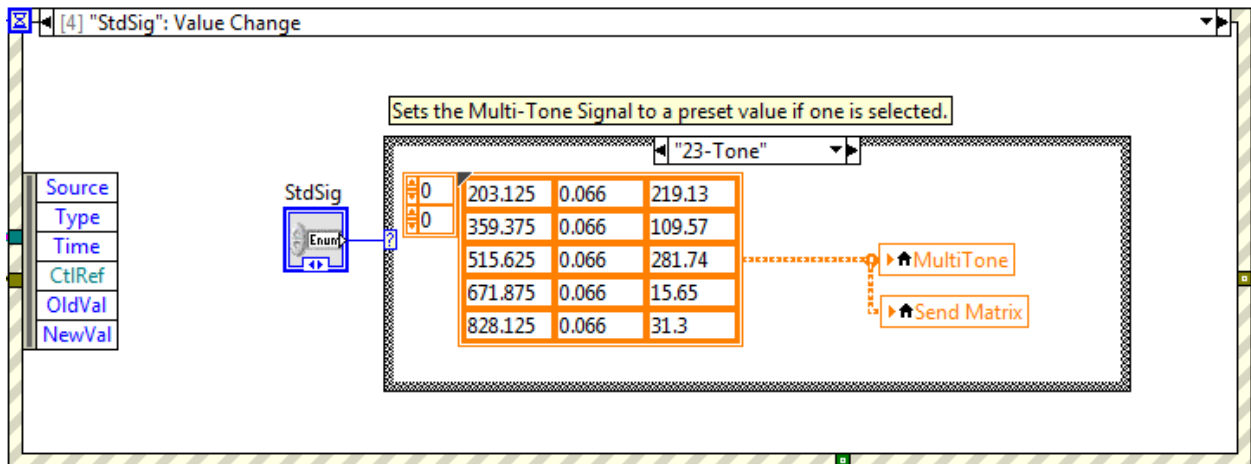
Figure 11-65: Function Generator Event Case, Set Multi-Tone Signal to a Standard Signal Event.

Change Multi-Tone Signal: This event activates whenever the multi-tone signal is changed. This event calculates the maximum amplitude and auto scales the tones to prevent the maximum output amplitude from being exceeded, **Figure 11-66**.
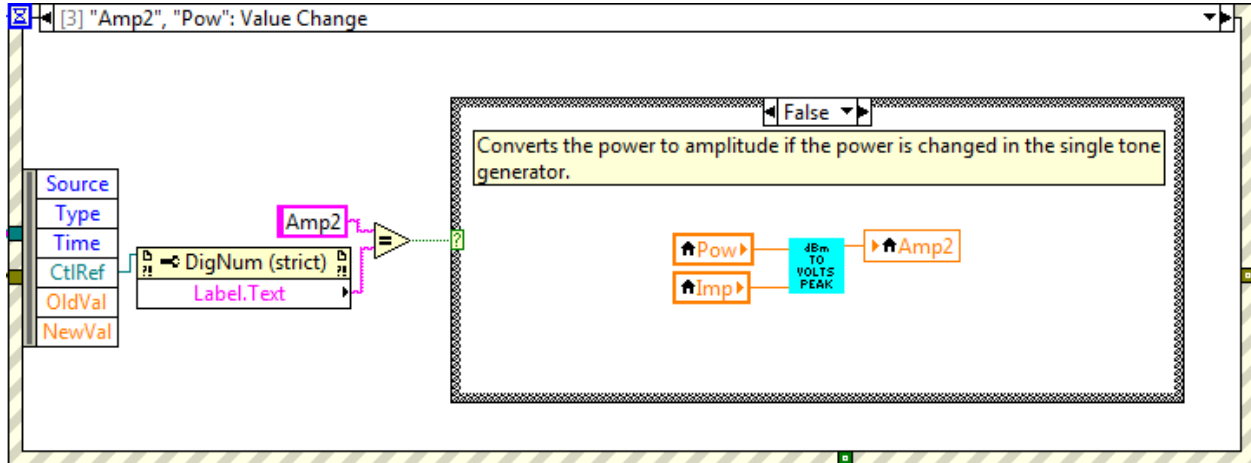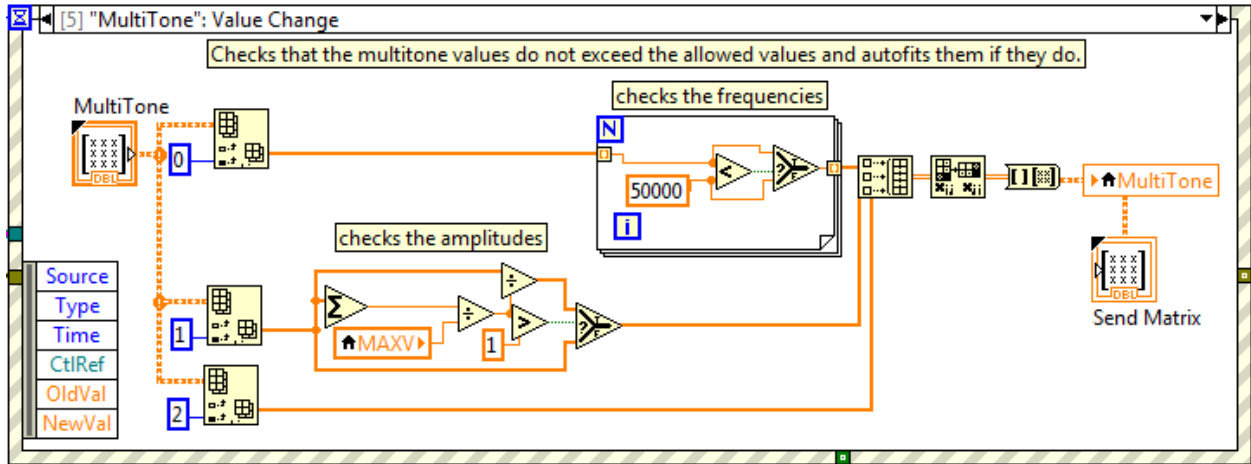

Figure 11-66: Function Generator Event Case, Change Multi-Tone Signal Event.

Change Sweep Settings: This event activates when any of the general function generator's sweep settings are changed. This event creates a list of frequencies the sweep will cycle through based on the new settings, **Figure 11-67**.
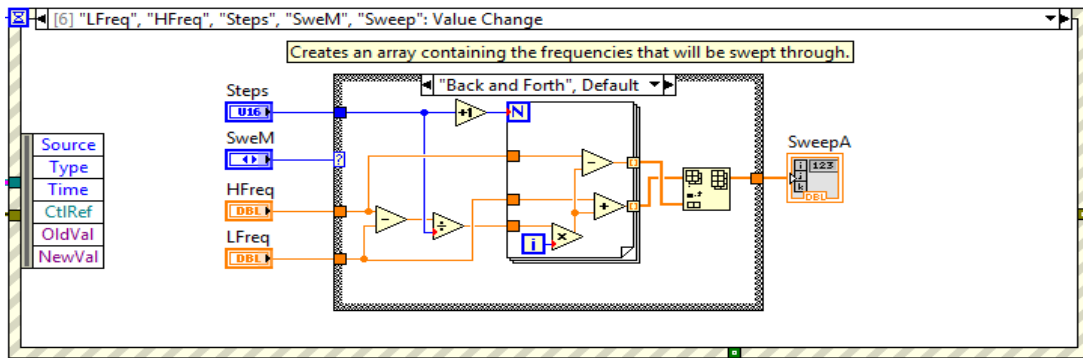
Figure 11-67: Function Generator Event Case, Change Sweep Settings Event.

Tab Control: This event is activated whenever a tab is selected from the pull down menu and activates the selected tab page, **Figure 11-68**.
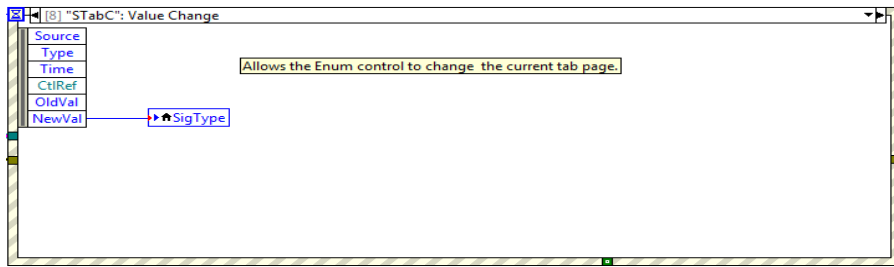
Figure 11-68: Function Generator Event Case, Tab Control Event.

Change Burst Amplitude and Power: This event activates when either of the burst signal's Amplitude, Power, or Duty Cycles are changed. This event auto adjusts the power and amplitude settings to ensure the maximum amplitude is not exceeded as well as forcing the two duty cycles to total to 100%, **Figure 11-69**.
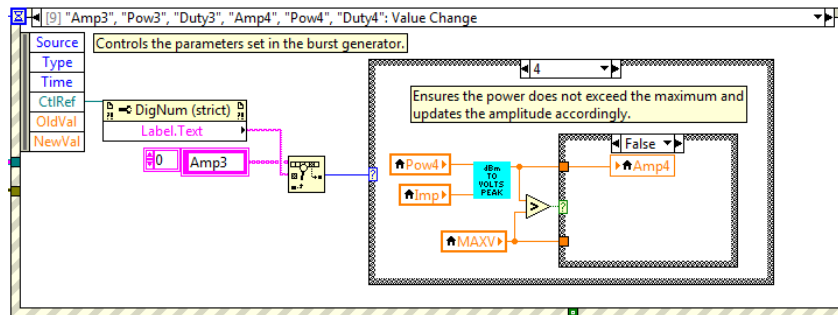
Figure 11-69: Function Generator Event Case, Change Burst Amplitude and Power Event.

Change Amplitude Modulation Amplitude and Power: This event activates when either of the amplitude modulation Amplitudes or Powers are changed. This event auto adjusts the power and amplitude settings to ensure the maximum amplitude is not exceeded while also ensuring the modulating amplitude does not exceed the carrier amplitude, **Figure 11-70**.
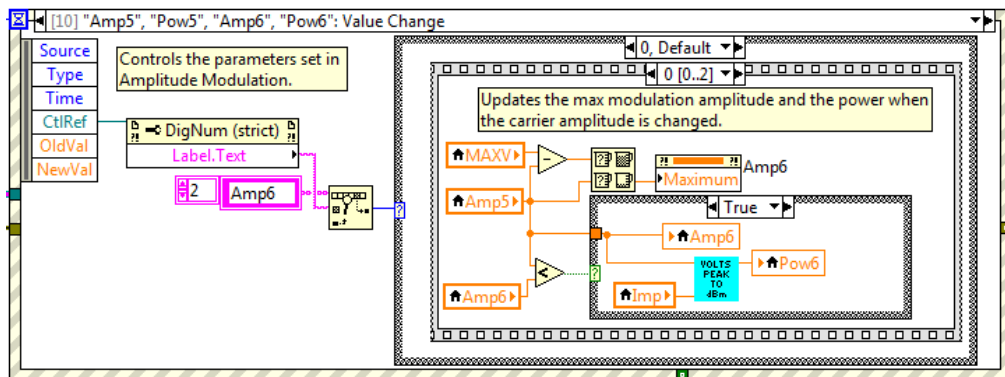
Figure 11-70: Function Generator Event Case, Change Amplitude Modulation Amplitude and Power Event

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Toggle Frequency and Phase Modulation: This event activates when frequency modulation is switched to phase modulation or vice-versa. This event updates the deviation constant text and maximum value, **Figure 11-71**.
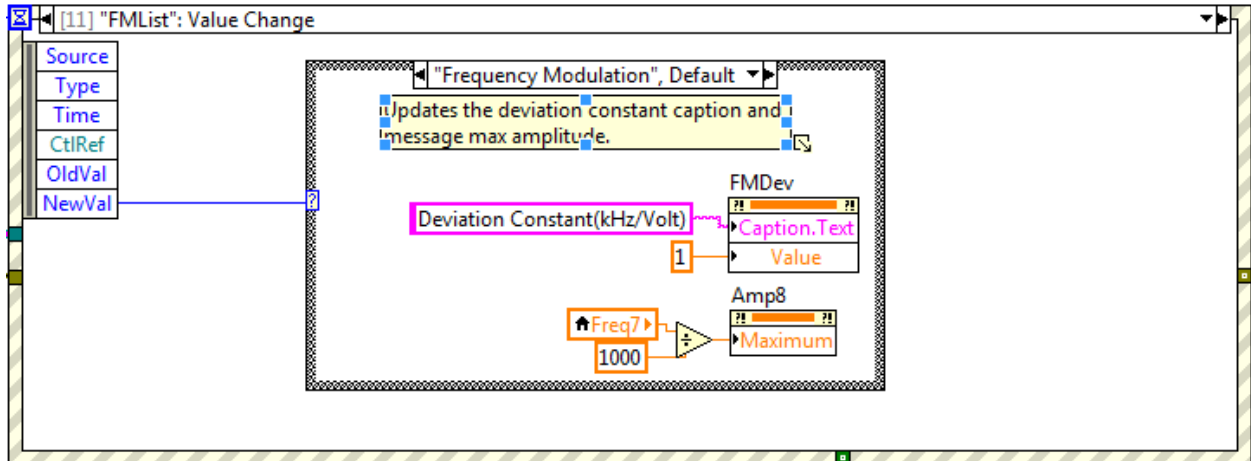
Figure 11-71: Function Generator Event Case, Toggle Frequency and Phase Modulation Event.

Change Frequency Modulation Amplitude and Power: This event activates when the frequency modulation's amplitude or power is changed. This event auto adjusts the power and amplitude settings to ensure the maximum amplitude is not exceeded while also ensuring the modulating amplitude and deviation constant never causes the carrier frequency to reach zero or the phase to swing by more than 360 degrees, **Figure 11-72**.
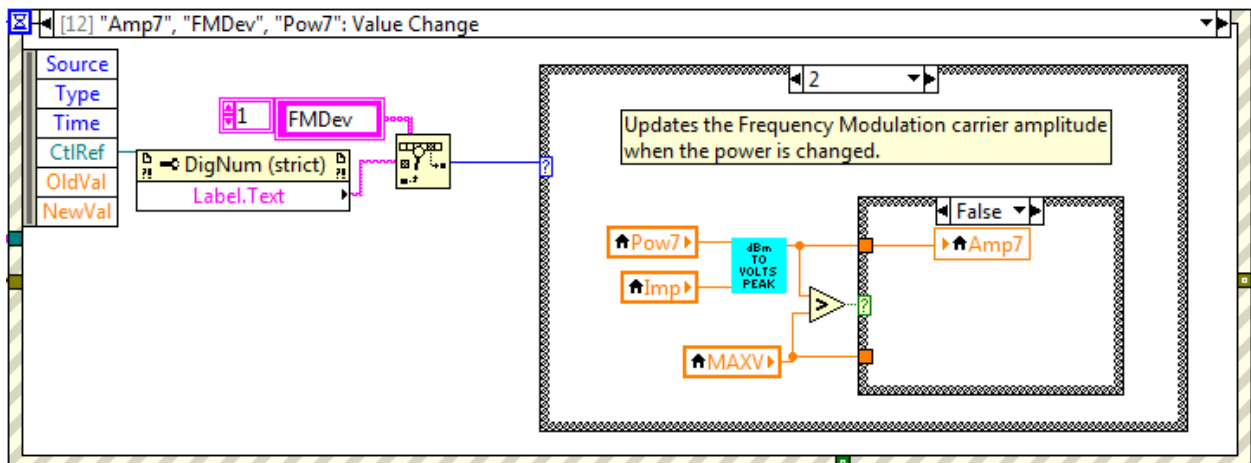
Figure 11-72: Function Generator Event Case, Change Frequency Modulation Amplitude and Power Event.

Change Impulse Amplitude and Power: This event activates when the impulse noise holding tone amplitude or power or the impulse amplitude is changed. This event auto adjusts the power and amplitude settings to ensure the maximum amplitude is not exceeded. Gain Hits and Square pulses are set to zero amplitude when the holding tone is changed, **Figure 11-73**.
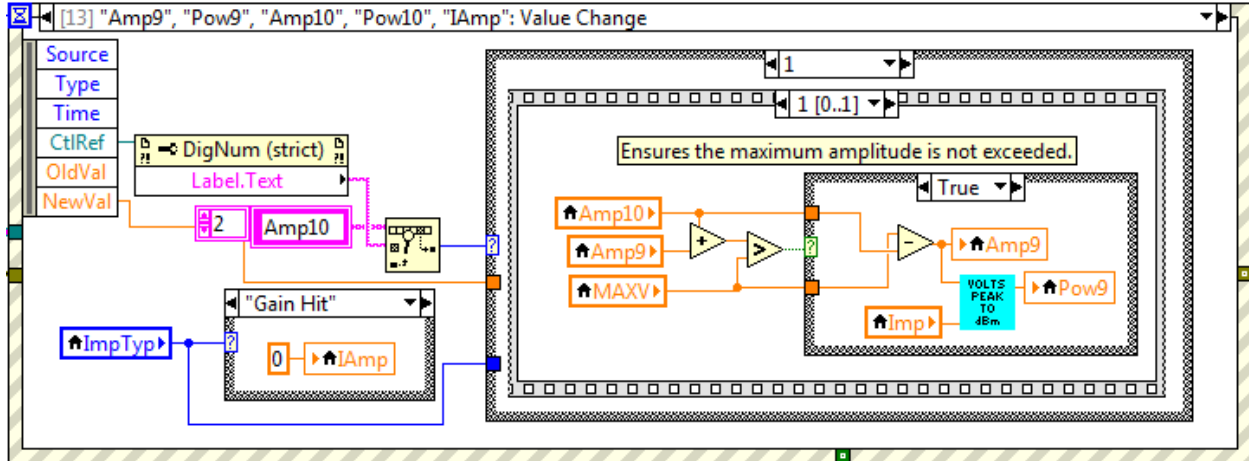
Figure 11-73: Function Generator Event Case, Change Impulse Amplitude and Power Event.

Change Impulse Type: This event is activated when the impulse type is changed. This event updates the units for the impulse duration and sets the entry limits for both the impulse amplitude and duration, **Figure 11-74**.
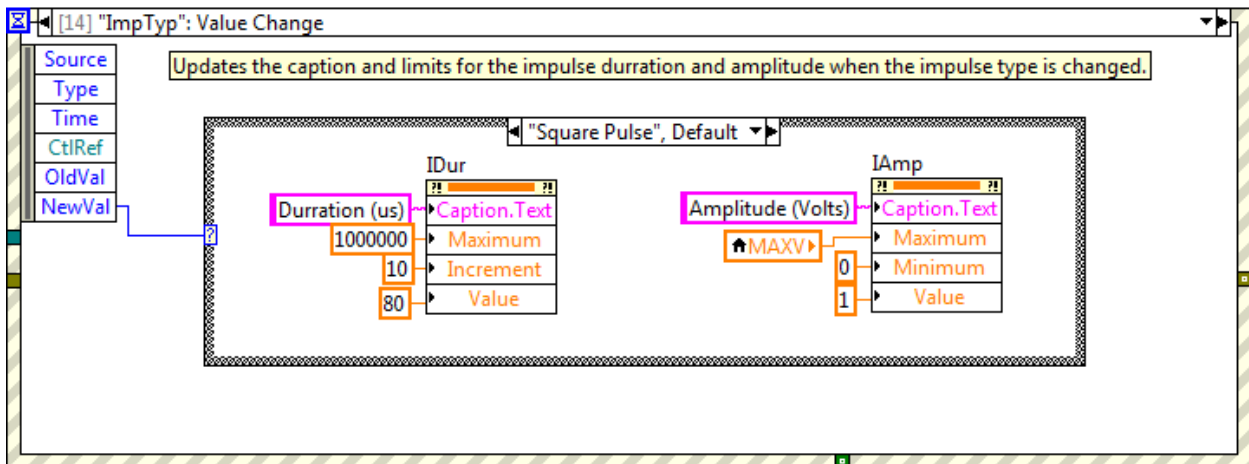
Figure 11-74: Function Generator Event Case, Change Impulse Type Event.

Change Noise Amplitude and Power: This event activates when the random noise holding tone amplitude or power or the random noise amplitude is changed. This event auto adjusts the power and amplitude settings to ensure the maximum amplitude is not exceeded, **Figure 11-75**.

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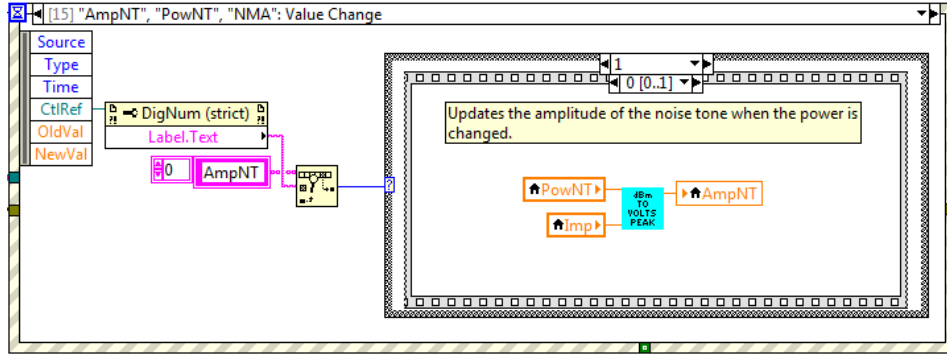
Figure 11-75: Function Generator Event Case, Change Noise Amplitude and Power Event.

11.5.3 Function Generator Queue.

The Function Generator's Queue Loop handles all of the commands passed to the VI. The primary function of the Queue Loop is to execute commands from other functions through the use of shared Queues. The Function Generator's Queue Loop can process the following unique commands:

1. New Tone
2. Calibration

New Tone: This command is queued whenever another function requires the function generator to transmit a new signal for it, **Figure 11-76**. The command contains the type of signal and all the necessary signal parameters for the function generator to generate the signal. Once the function generator receives this command it will begin to transmit the new signal.
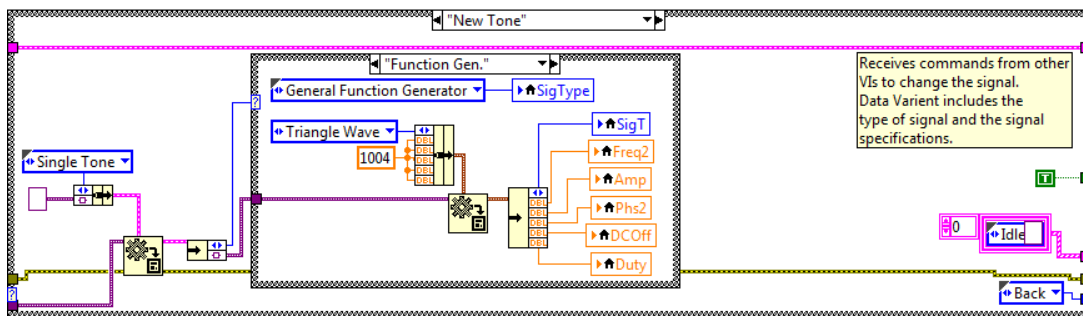
Figure 11-76: Function Generator Queue, New Tone Command.

Calibration: This command is queued when the virtual test set is calibrating. The virtual test set requires the function generator when it is calibrating its output. This command contains all of the parameters for the new signal as well as indicating the type of calibration that is being performed, **Figure 11-77**. When the function generator receives this command it will begin generating the new signal even if the signal exceeds the normal output limits of the function generator for the given hardware settings, these settings are set for the hardware output while calibration uses the DAQ output.
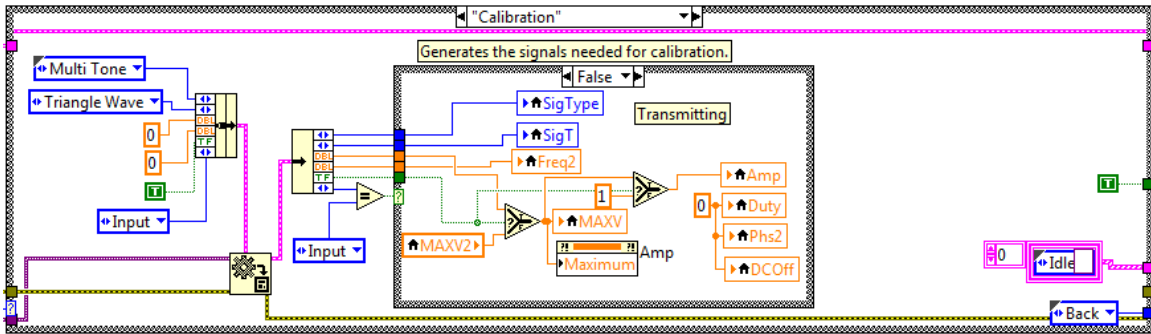
Figure 11-77: Function Generator Queue, Calibration Command.

11.5.4 Function Generator Signal Generation

The Function Generator's Signal Generation section consists of a case structure that is linked to the front panel's signal type selector. The case structure creates the specified signal and passes it to the transmit signal graph. The case structure has the following cases:

- Single Tone
- Multi-tone
- General Function Generator
- Noise
- Amplitude Modulation
- Frequency Modulation
- Impulse Noise

Single Tone: This case generates a single sinusoidal signal by passing the parameters to a basic LabVIEW function, **Figure 11-78**.
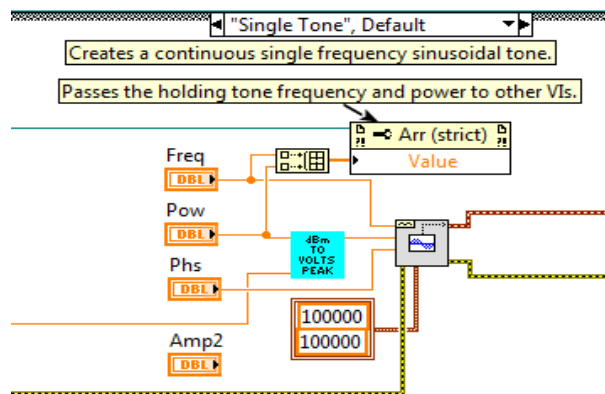
Figure 11-78: Function Generator, Single Tone Block Diagram

MultiTone: This case generates a signal consisting of multiple frequencies by passing the parameters to a basic LabVIEW function, **Figure 11-79**.
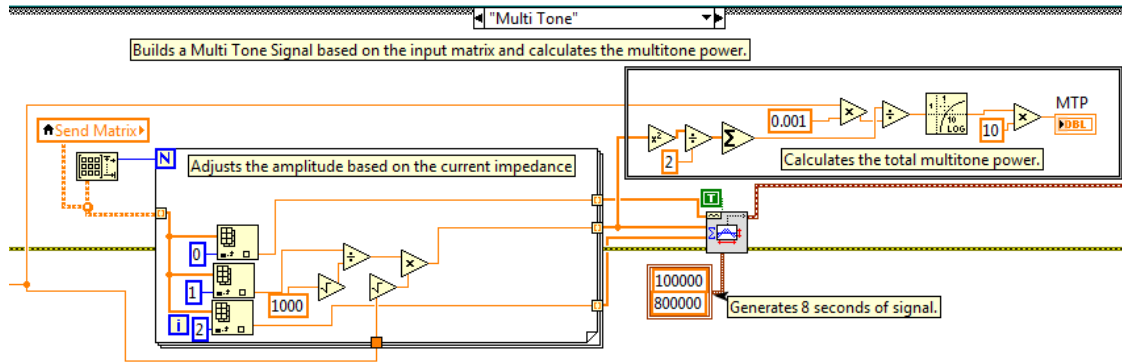
Figure 11-79: Function Generator, MultiTone Block Diagram

General Function Generator: This case generates sinusoidal, square, triangle, and sawtooth waves by passing the parameters to a basic LabVIEW function, **Figure 11-80**. This case is also capable of performing a sweep function by generating the different frequencies in segments equal to the hold time, appending the segments end to end in an array, and sending the resulting sweep waveform one second at a time, **Figure 11-81**.
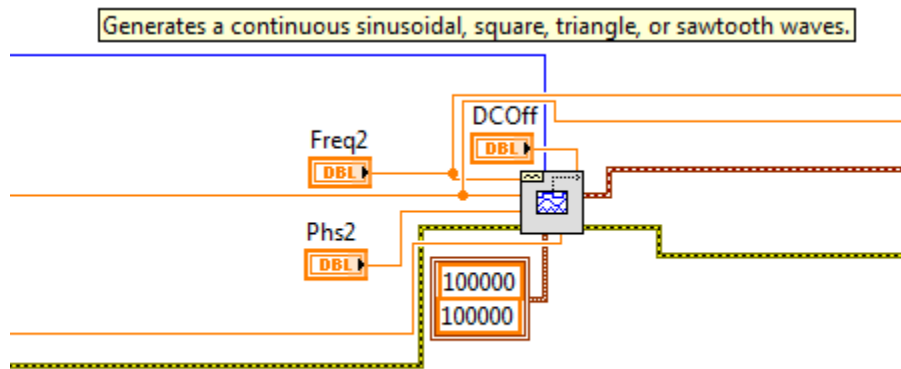
Figure 11-80: Function Generator, Continuous General Function Generator Block Diagram.
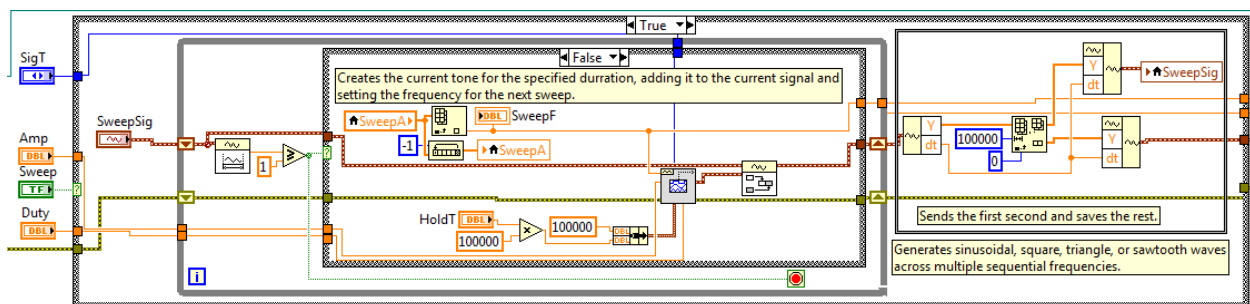
Figure 11-81: Function Generator, Sweeping General Function Generator Block Diagram.

Noise: This case generates a single sinusoidal signal and Gaussian noise by passing the parameters to basic LabVIEW functions, **Figure 11-82**. The wide band Gaussian noise is passed through a bandpass filter with cutoff frequencies determined by the user and calculates the power before the noise is added to the sinusoidal waveform.
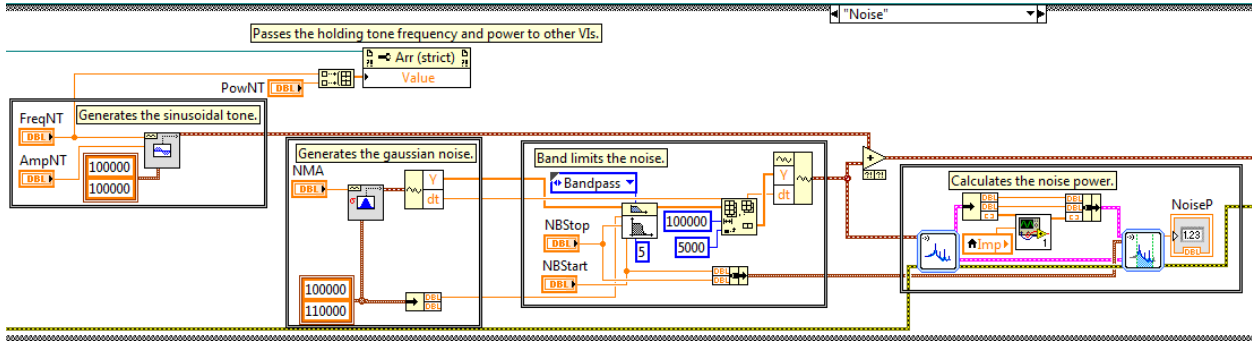
Figure 11-82: Function Generator, Noise Block Diagram.

Burst: This case generated short duration sinusoidal signals by multiplying two sinusoidal signals by a unity amplitude square wave with the desired gating frequency and duty cycle. The first sinusoidal signal is multiplied with the square wave and the second is multiplied by the logical inverse of the square wave and the two resulting signals are added together, **Figure 11-83**.
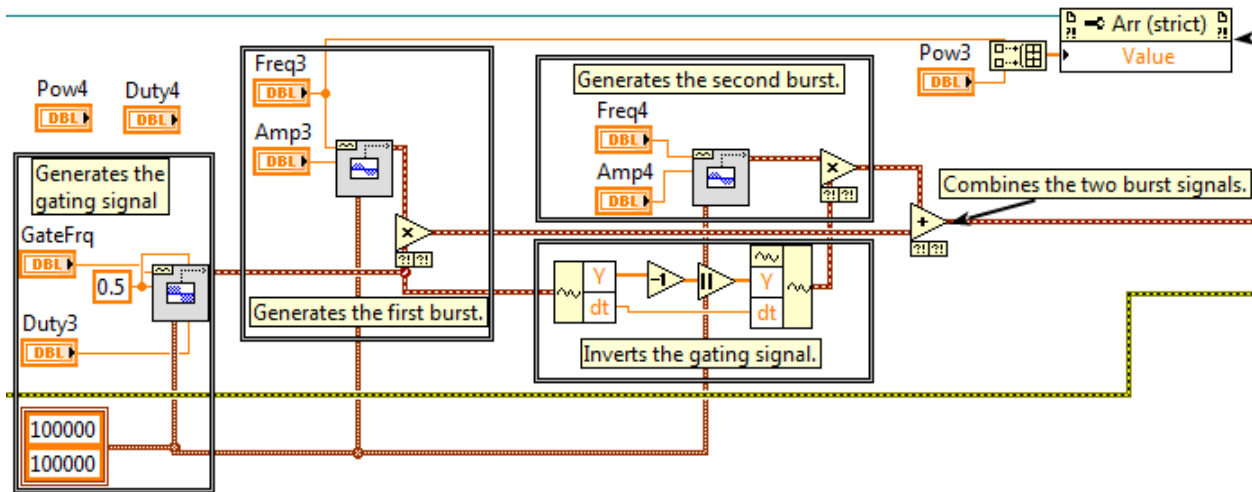
Figure 11-83: Function Generator, Burst Block Diagram.

Amplitude Modulation: This case generates an amplitude modulated signal by generating two sinusoidal signals. The message waveform is added to the amplitude of the carrier and the result is divided by the carrier amplitude. The carrier signal is multiplied by the resulting gain signal to produce the AM signal, **Figure 11-84**.
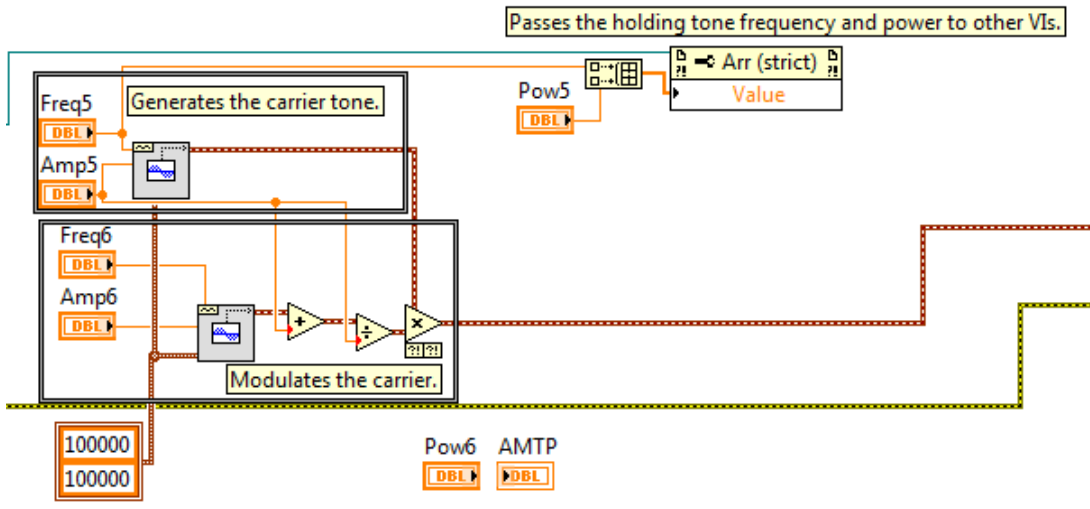
Figure 11-84: Function Generator, Amplitude Modulation Block Diagram.

Frequency Modulation: This case generates frequency and phase modulated signals. The message waveform is multiplied by the deviation constant and the unity gain gating signal. For Frequency Modulation the result is a Frequency vs Time signal. For Phase Modulation the result is a Phase Offset vs Time signal. The FM signal is generated by calculating the point by point phase of the signal and multiplying the sine of the signal by the carrier signal amplitude, **Figure 11-85**.
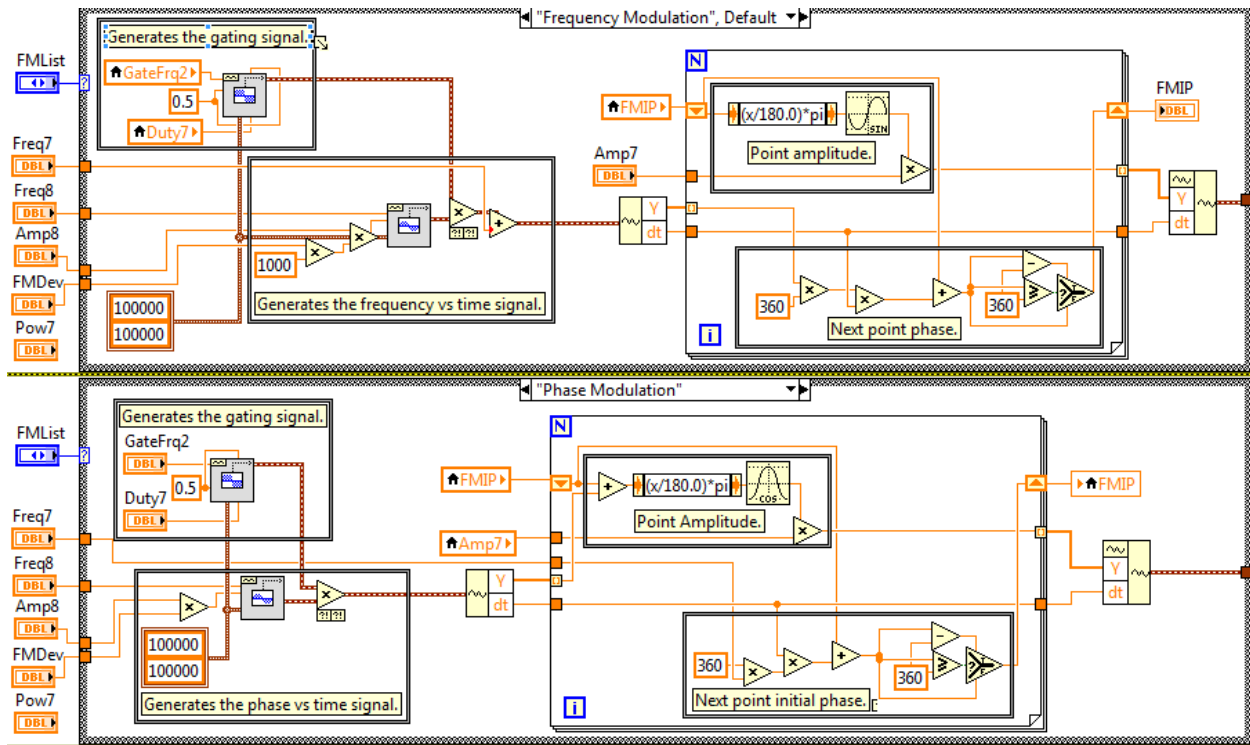
Figure 11-85: Function Generator, Frequency Modulation Block Diagram.

Impulse Noise: This case generates a sinusoidal signal and adds a second sinusoidal signal and transient noise, **Figure 11-86**. The message waveform is multiplied by the deviation constant and the unity gain gating signal. The available transient signals include square pulses, phase hits, and gain hits. For square pulses the pulse signal is simply added to the combined sinusoidal signals. For phase hits, a square wave phase modulation is applied to the designated holding tone. For gain hits, a square wave amplitude modulation is applied to the designated holding tone.
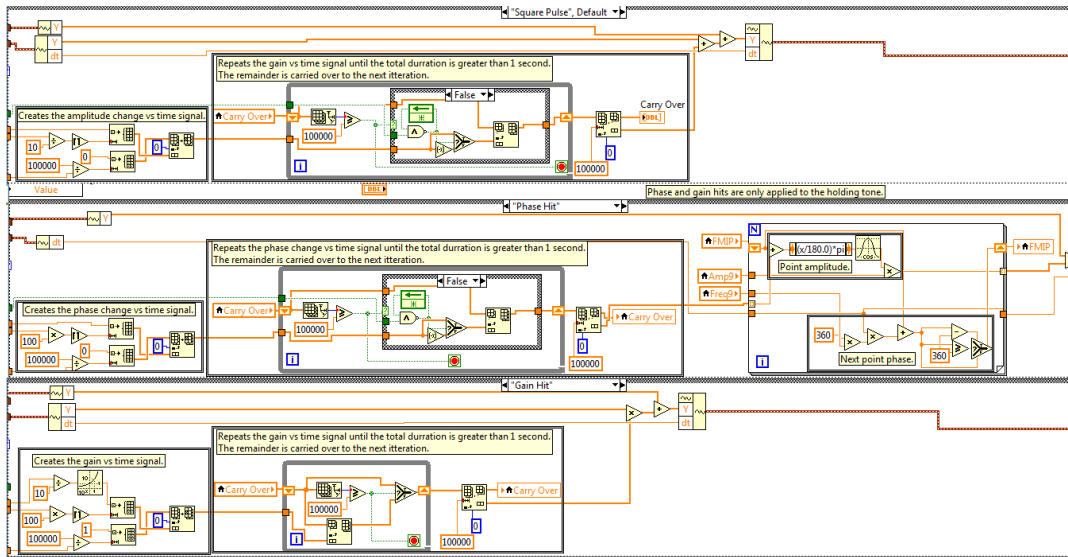
Figure 11-86: Function Generator, Impulse Block Diagram.

11.6 Transmission Impairment Measurements Block Diagram

The block diagram of the Transmission Impairment Measurement (TIM) VI can be broken into different sections: event case, queue, series tone, auto test, and measurement sections, **Figure 11-87**.
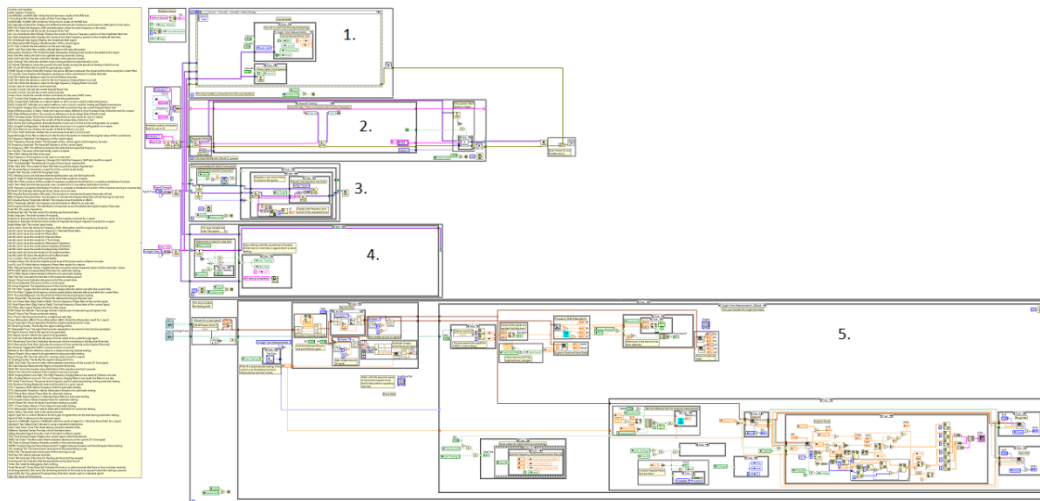
Figure 11-87: TIM Block Diagram.

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11.6.1 TIM Event Case Section.

The TIM's event case handles the commands from the user. The primary functions of the TIM's event cases are controlling the settings and flow of the Transmission Impairment tests. The TIM VI has the following unique event cases:

1. Idle
2. Reorder Impulse Noise Thresholds.
3. Calculate Remaining Time.
4. Start Impulse Noise Measurement.
5. Start a Series Tone Test.
6. Set the Series Tone Test to a Standard Series.
7. Select a Standard Testing Regiment for Automatic Testing.
8. Add Tests to the Automatic Test Queue.
9. Save the current values for a report.
10. Pause automatic testing.
11. Cancel a test.
12. Change the viewed tests for automatic testing.
13. Calibrate an Echo test.

Idle: This event is activated when 200ms pass without any user commands. The TIM idle event queues the impulse testing command to turn off the impulse testing state once the impulse test finishes, **Figure 11-88**.
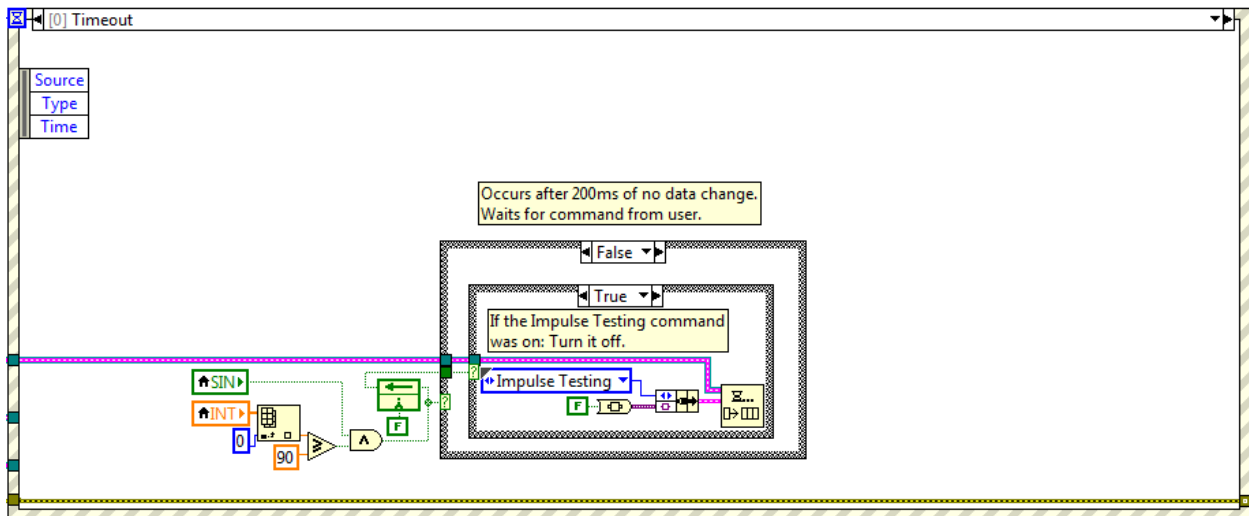
Figure 11-88: Transmission Impairment Measurement, Idle Event.

Reorder Impulse Noise Thresholds: This event is activated when the threshold settings for impulse noise are changed. This event re-orders the thresholds into descending order, **Figure 11-89**.
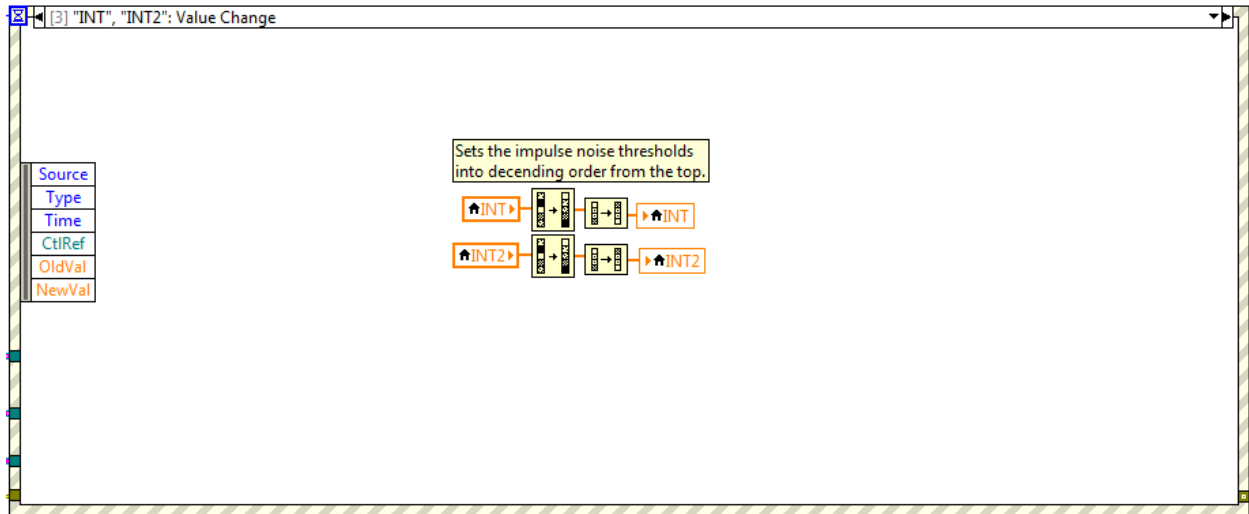
Figure 11-89: Transmission Impairment Measurement, Reorder Impulse Noise Thresholds Event.

Calculate Remaining Time: This event is activated when the duration of the impulse noise test is changed. This event converts the duration from minutes to seconds, **Figure 11-90**.
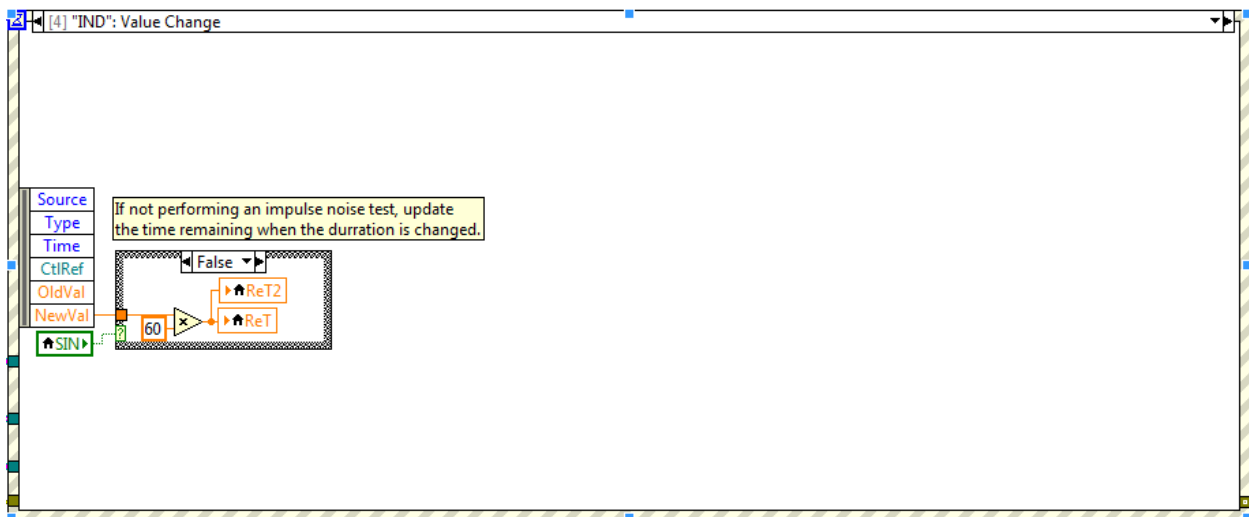
Figure 11-90: Transmission Impairment Measurement, Calculate Remaining Time Event.

Start Impulse Noise Measurement: This event is activated when an impulse noise test is activated. This event sets all the initial settings for the impulse noise test, **Figure 11-91**.
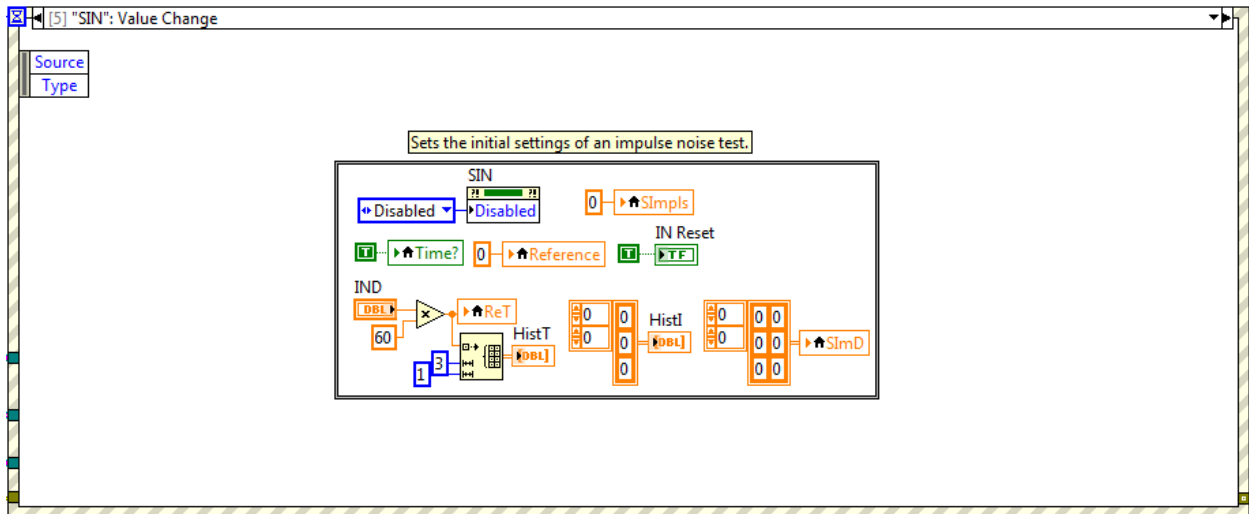
Figure 11-91: Transmission Impairment Measurement, Start Impulse Noise Measurement Event.

Start a Series Tone Test: This event is activated when a series tone test is begun. This event sets all of the initial settings of the test, queues the series of tones to be cycled through during the test, and calculates the lossless values for an echo test, **Figure 11-92**. The lossless values are calculated by sending the ideal tones through the echo loss filters for later comparison, **Figure 11-93**.
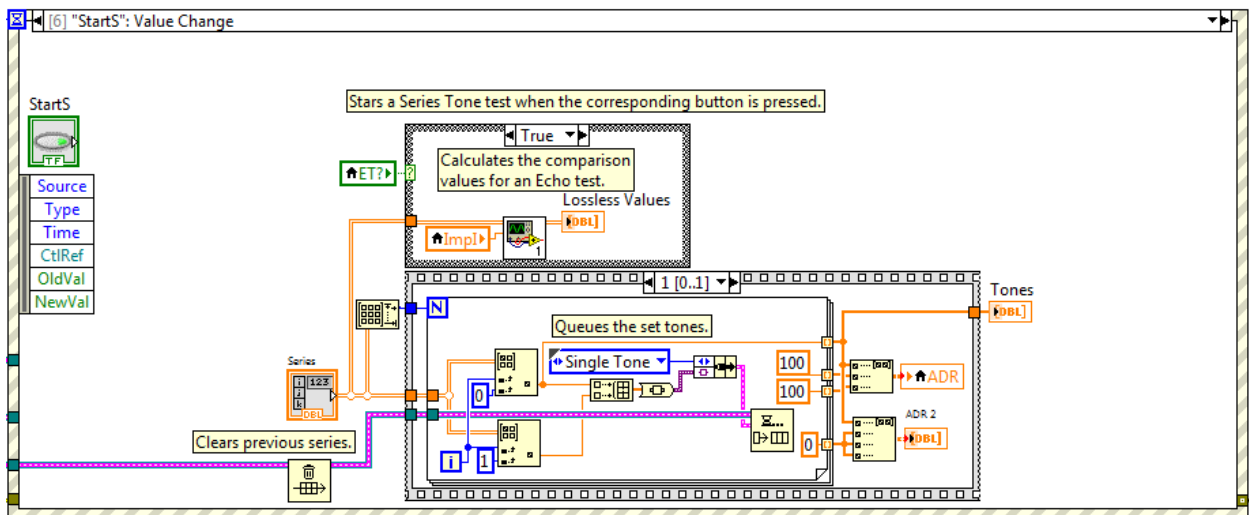
Figure 11-92: Transmission Impairment Measurement, Start a Series Tone Test Event.
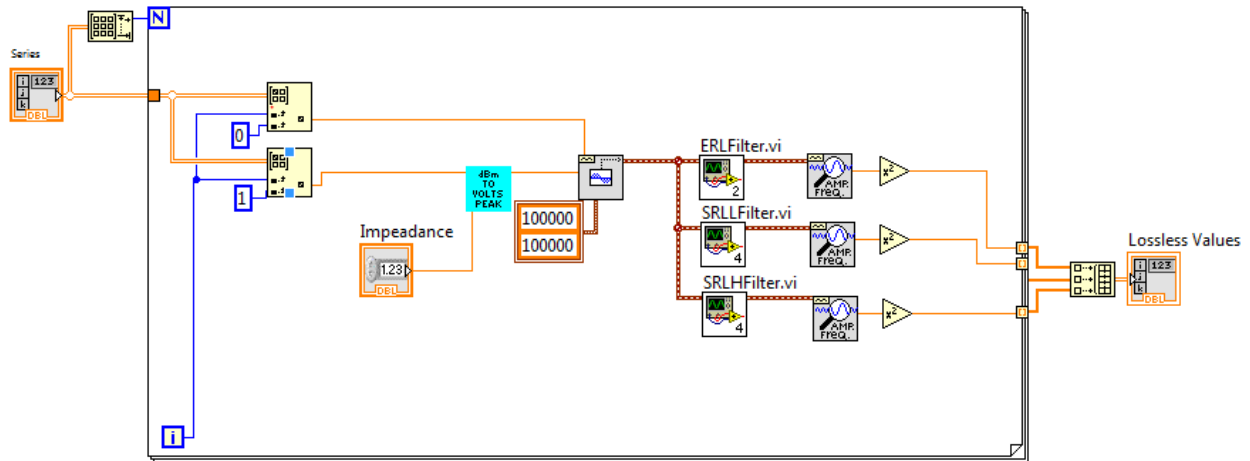
Figure 11-93: Lossless Echo Loss Sub-VI.

Set the Series Tone Test to a Standard Series: This event is activated whenever the user selects a standard series from the dropdown menu. This event sets the series to a pre-set series, **Figure 11-94**.
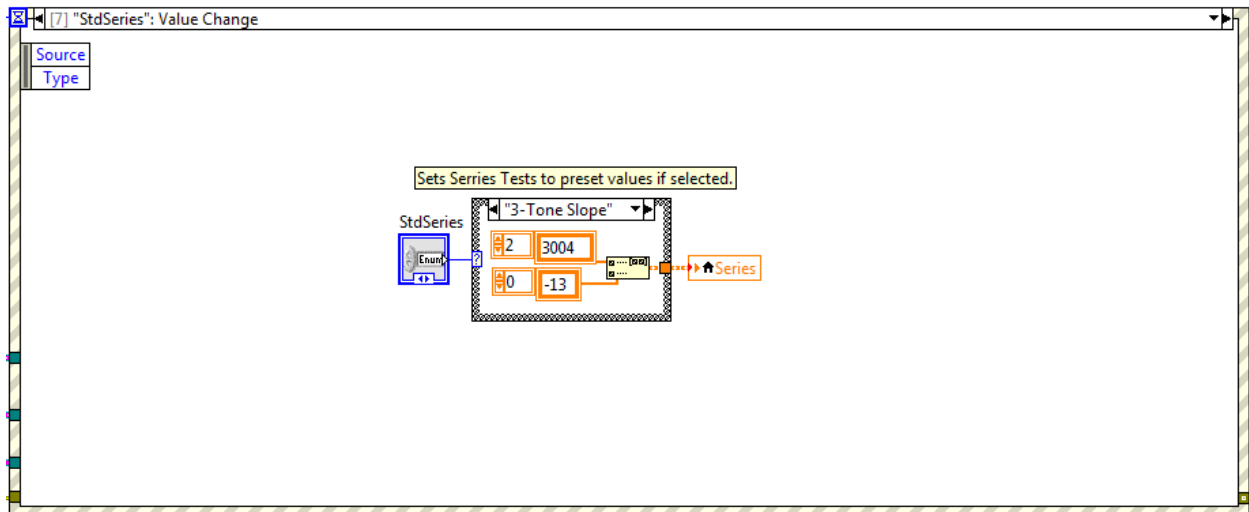
Figure 11-94: Transmission Impairment Measurement, Set the Series Tone Test to a Standard Series Event.

Select a Standard Testing Regiment for Automatic Testing: This event is activated whenever the user selects a standard testing regiment for automatic testing. This event sets all of the automatic testing settings to those of the selected testing regiment, **Figure 11-95**.
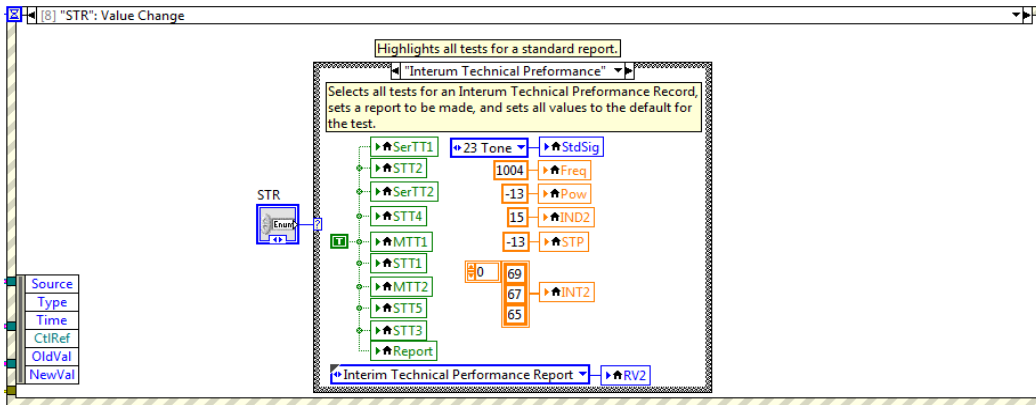
Figure 11-95: Transmission Impairment Measurement, Select a Standard Testing Regiment for Automatic Testing Event.

Add Tests to the Automatic Test Queue: This event is activated when the user chooses to add the selected tests to the automatic test queue. This event adds all of the selected tests, their settings, and a report if selected to the automatic test queue before resetting all of the test selection buttons, **Figure 11-96**.
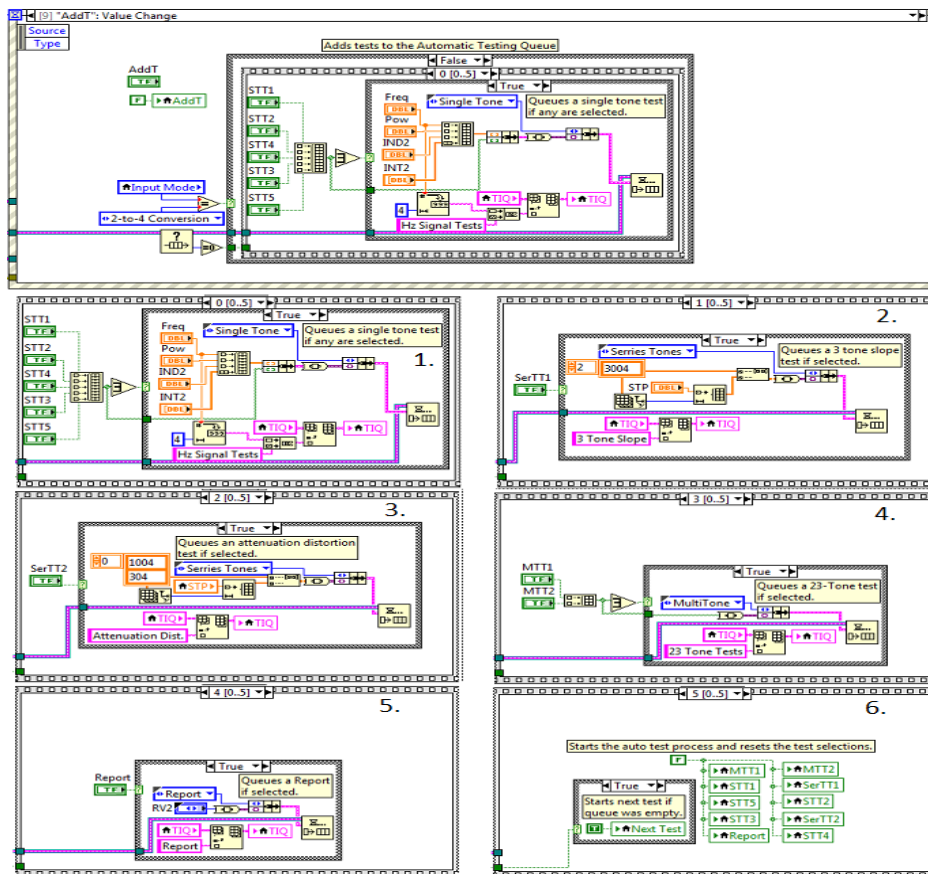
Figure 11-96: Transmission Impairment Measurement, Add Tests to the Automatic Test Queue Event Plus Stacked Sequence Frames.

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Save the current values for a report: This event is activated whenever the user presses a 'latch' button. This even stores the most recent test results for the selected test or tests to be later used in a report, **Figure 11-97**.
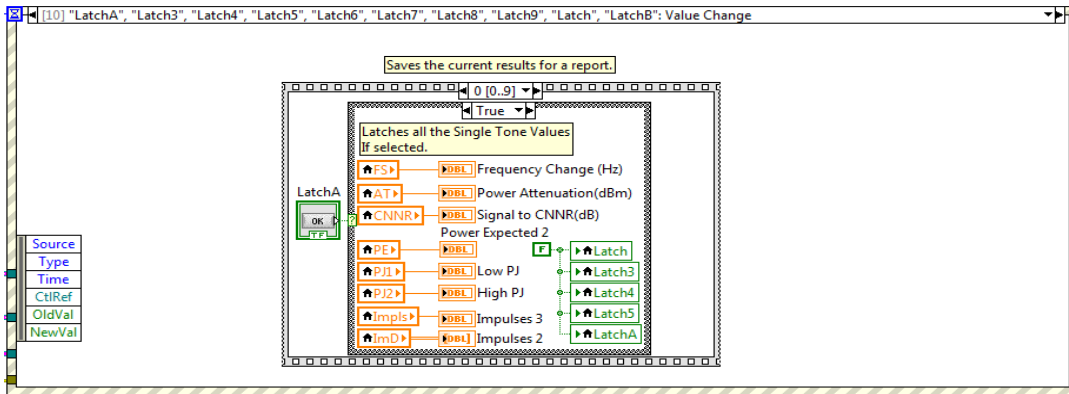
Figure 11-97: Transmission Impairment Measurement, Save the current values for a report Event.

Pause automatic testing: This event is activated when the user pauses or resumes automatic testing. When automatic testing is paused, this event saves all of the current VI values and stops all testing, **Figure 11-98**. When automatic testing is resumed this event restores all of the saved values and resumes testing.
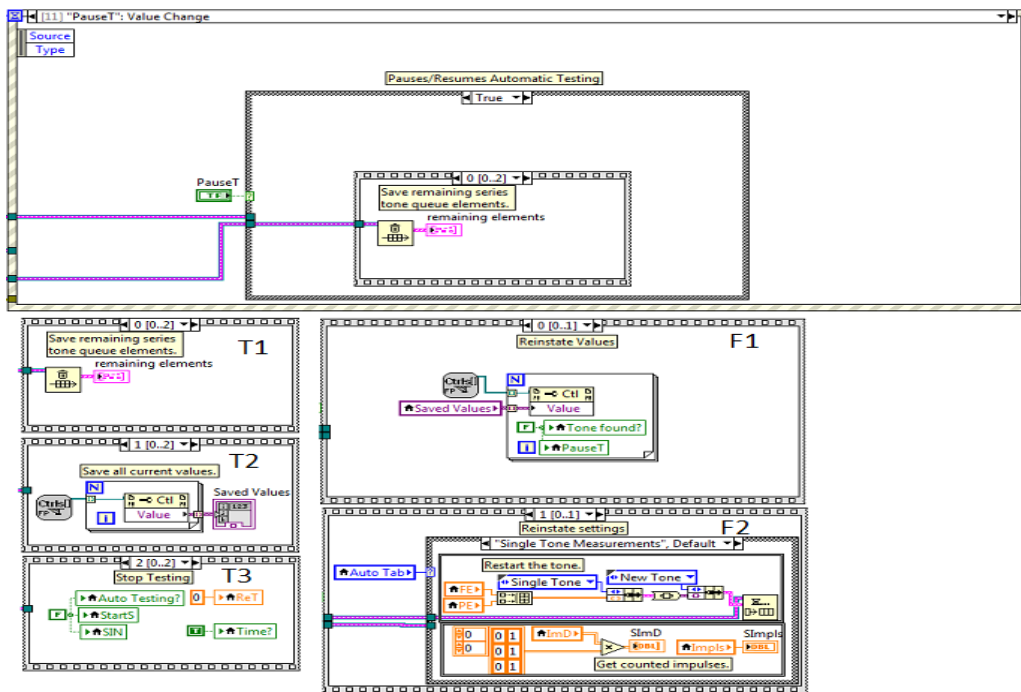
Figure 11-98: Transmission Impairment Measurement, Pause automatic testing Event plus Stacked Sequence Frames for true and false cases.

Cancel a test: This event is activated whenever a test is canceled. For an impulse test, the remaining time is set to zero to immediately finish the test. For a series test the signal series queue is flushed. An automatic test will perform either of the previous actions depending on the current test, **Figure 11-99**.
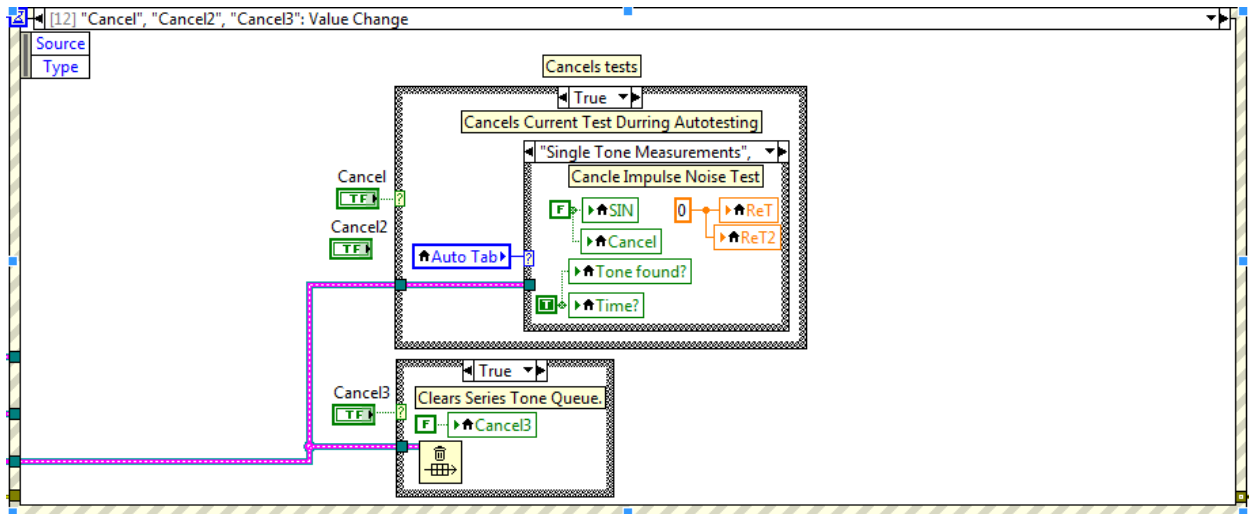
Figure 11-99: Transmission Impairment Measurement, Cancel a test Event.

Change the viewed tests for automatic testing: This event is activated whenever the user selects a test to view from the pull down menu on the automatic testing page. This event simply passes the selected value to the value to the tab control for automatic test selection.

Calibrate an Echo Test: This event is called when the user chooses to save the most recent echo test results as a reference for a future echo test. The test results can be saved temporarily or permanently. If the results are to be saved temporarily the current results simply overwrite the current calibration constants. If the results are to be saved permanently the current calibration constants and the values in the echo calibration file are overwritten, **Figure 11-100**.
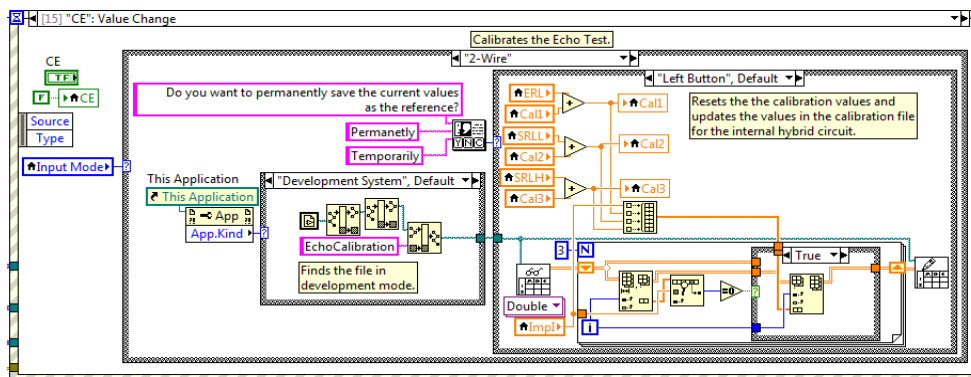
Figure 11-100: Transmission Impairment Measurement, Calibrate an Echo Test Event.

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11.6.2 TIM Queue

The TIM's Queue Loop handles all of the commands passed to the VI and passes commands to other VI's. The primary function of the TIM Queue Loop is to request new tones from the function generator through the use of shared Queues. The TIM's Queue Loop can process the following unique commands:

1. New Tone
2. Impulse Testing

New Tone: This command is queued during a series tone or automatic test to request a new signal to be transmitted from the function generator, **Figure 11-101**. The command contains the type of signal and all the necessary signal parameters for the function generator to generate the signal.
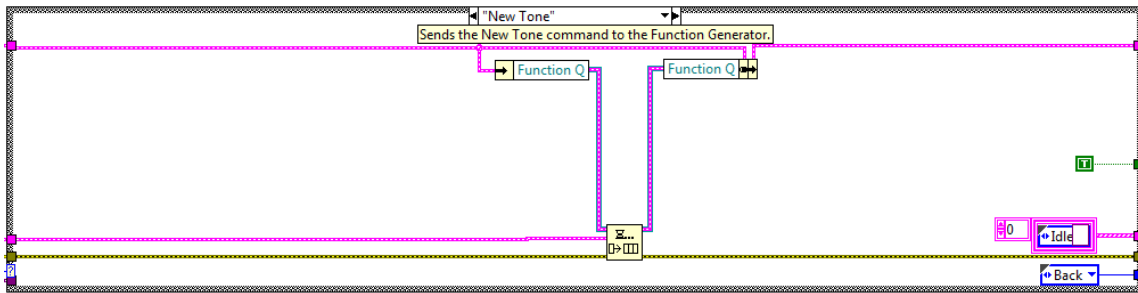
Figure 11-101: Transmission Impairment Measurement Queue, New Tone Command.

Impulse Testing: This command is queued during impulse testing to have the range and accuracy to the virtual test set's receive function be dependent on the threshold level of the impulse test rather than the max voltage of the received signal. This command is queued at the start of an impulse test to activate the receive functions impulse testing mode and once the test is finished to deactivate it, **Figure 11-102**.
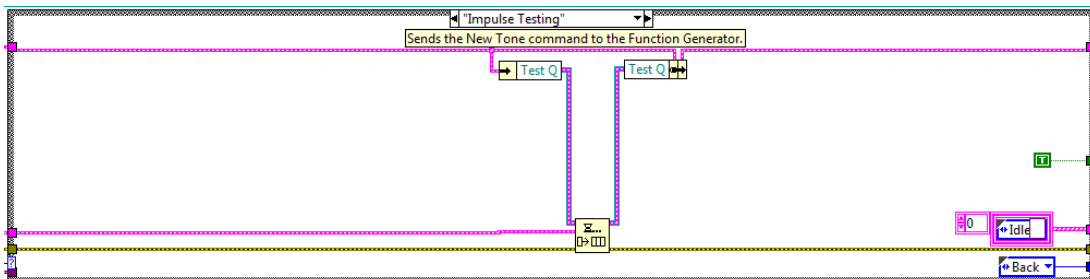
Figure 11-102: Transmission Impairment Measurement Queue, Impulse Testing Command.

11.6.3 TIM Filtering and Graph Generation

Filtering is necessary to limit the range of frequencies that will be measured by any given test. Some tests, like attenuation distortion and envelope delay distortion, cannot use filter because of the distortions they introduce. Other tests, like signal to noise ratio, must use filters to measure the noise in the frequency band of interest. The virtual test set is able to provide C, Psophometric, D, 3.4k Hz Flat, and 15k Hz Flat filters or none at all. For audio range measurements the C-Message filter is the most commonly used filter because it mimics the attenuation of the human ear, **Figure 11-103**, allowing tests to only measure noise that will be audible. The D-Message, 3.4k Flat, and 15k Flat filters are simple filters that pass different bands of frequencies **[1]**. The D-Message filter being a bandpass filter from 300 Hz to 3.4k Hz, the 3.4k and 15k Flat filters being low pass filters for frequencies below 3.4k and 15k Hz respectively. The Psophometric filter is a filter that has attenuation similar to the human ear and follows the ITU-T O.41 standard **[15]**, it was included as an alternative to the C-Message Filter. When a filter is used, the virtual test set saves two consecutive segments of the sampled signal before passing the two segments through the filter, **Figure 11-104**. Measurements must be made from the middle segment of the filtered signal to eliminate the transient effects of the filter. When a new segment of signal is sampled, the older of the two saved segments is removed and the new segment is placed at the end. Finally, a truncated and un-truncated version of the filtered signal is passed on to be measured for impairments. The un-truncated signal is used when the measurement must apply further filtering. This section of the code also creates both the time and frequency domain signal graphs. When the frequency domain graph is first generated the dB values are referenced to an assumed 1000 ohm impedance and must be scaled to the actual impedance using the dB to dB converted sub-VI. The graphs can display either the pre-filtered signal or the post-filtered signal.

Frequency in Hz

Freq. in Hz	Analog Loss in dB	Tol.	Digital Loss in dB
60	55.7	2	56.6
100	42.5	2	43.3
200	25.1	2	25.8
300	16.3	2	16.8
400	11.2	1	11.4
500	7.7	1	7.8
600	5.0	1	5.0
700	2.8	1	2.8
800	1.3	1	1.3
900	0.3	1	0.3
1000	0.0	0.0	0.0
1200	0.4	1	0.4

Freq. in Hz	Analog Loss in dB	Tol.	Digital Loss in dB
1300	0.7	1	0.7
1500	1.2	1	1.2
1800	1.3	1	1.4
2000	1.1	1	1.2
2500	1.1	1	1.2
2800	2.0	1	2.1
3000	3.0	1	3.0
3300	5.1	2	4.5
3500	7.1	2	6.0
4000	14.6	3	8.7
4500	22.3	3	6.0
5000	≥28.7		3.0

See discussion on digital filter aliasing in 9.1.4.

Attenuation shall increase at not less than 12 dB/oct until it reaches 60 dB.

Figure 11-103: C-Message Filter
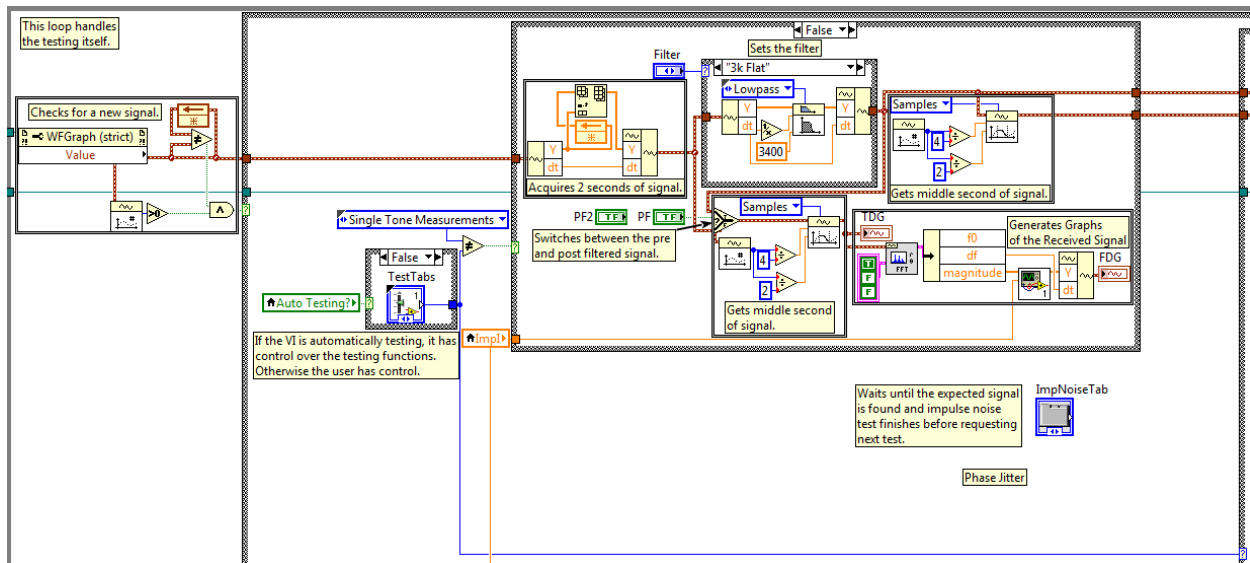
Figure 11-104: Transmission Impairment Measurement Block Diagram, Filter and Graph Generation.

11.6.4 Single Tone Measurement

This case of the measurement section of the block diagram performs the impairment measurements that are performed by transmitting a single tone signal through the system. The single tone measurement case is broken into continuous measurements and transient measurements, **Figure 11-105**.
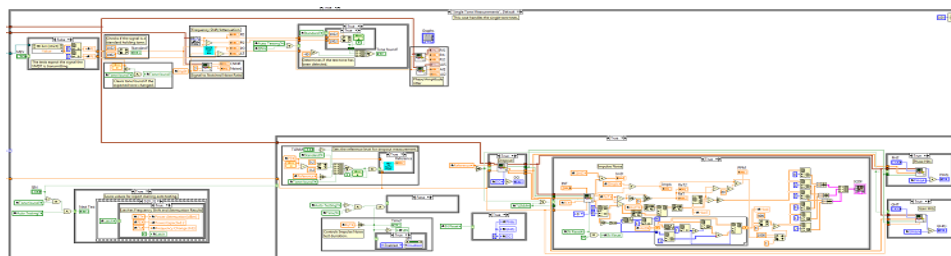
Figure 11-105: Transmission Impairment Measurement Block Diagram, Single Tone Measurements Case

11.6.4.1 Continuous Measurements

Continuous single tone measurements are made by tests that provide new results whenever a new time segment of the received signal is captured. For impulse noise and auto testing purposes, testing will not begin until the expected signal is received. The received signal must be within 1002 and 1022 Hz if the expected tone is a holding tone or within 5 Hz of expected tone if the expected tone is not a holding tone and must have a signal power greater than -40 dBm, **Figure 11-106**. Continuous measurement tests do not require the user to start them, can run in parallel with each other, and can be run indefinitely. Frequency Shift, Attenuation, Signal to Noise Ratio, Noise Level, Phase Jitter, and Amplitude Jitter are all continuous measurements.
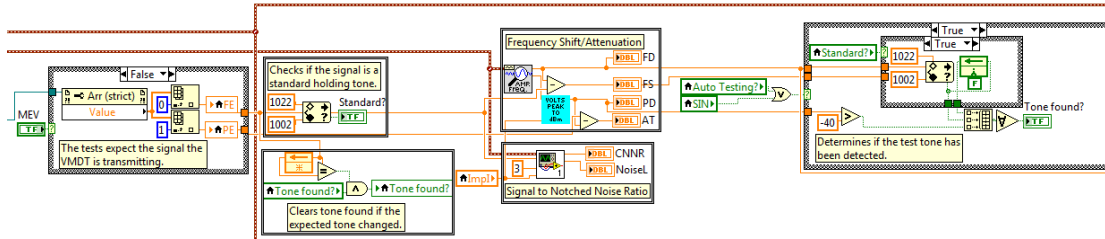
Figure 11-106: Single Tone Measurements Case, Continuous Measurement Section

11.6.4.1.1 Frequency Shift and Attenuation

Frequency Shift and Attenuation measure the changes a signal undergoes between the point it was transmitted and the point it was received. Frequency Shift is calculated by taking the difference between the received frequency and the expected frequency. Attenuation is measured by calculating the power of the received signal and comparing it with the expected power of the received signal. The received frequency, Frequency Shift, received power, and Attenuation are measured at the same time, **Figure 11-107**.
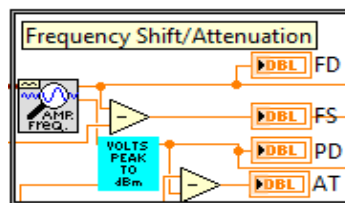
Figure 11-107: Single Tone Measurements Case, Frequency Shift Section

11.6.4.1.2 Signal to Notched Noise Ratio and Noise Level

This test determines the signal to noise ratio by comparing the total power of the signal before being passed through a notched filter to the total power of the signal after being passed through a notched filter. For noise measurement purposes, any filter that attenuates the test tone by 50 dB and falls within the exclusion zones will satisfy IEEE requirements **[1]**. The virtual test set applies a notch filter by converting the received signal to the frequency domain and omitting the frequencies around the expected frequency leaving only the noise frequencies, **Figure 11-108**. The power in the noise band is subtracted from the power of the received signal to obtain the signal to noise ratio.
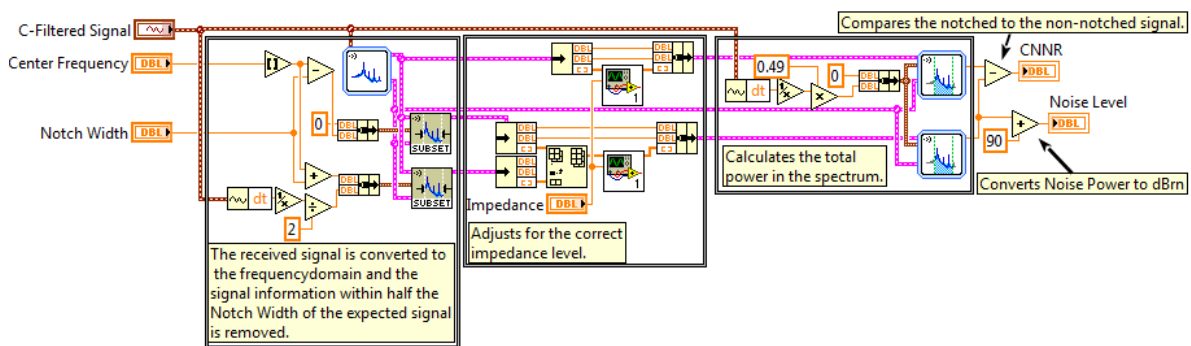
Figure 11-108: Noise Measurement Sub-VI Block Diagram

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11.6.4.1.3 Phase Jitter

Phase Jitter is measured by isolating the holding tone with a narrow bandpass filter. Once the holding tone is isolated, the signal is converted to a phase vs time signal by taking the inverse tangent of the Hilbert transformed signal divided by the original signal. The Hilbert transform of the signal shifts all of the frequency components by -90º **[16]**. Since the average phase change on the signal must be zero for a signal of constant frequency only the absolute amplitude of the Phase vs Time signal is measured and the negative can be ignored. This has the effect converting a sine into a cosine at every point in the signal, which when divided by the original signal will obtain a tangent signal with unity gain regardless of the amplitude of the original sinusoidal. The tangent signal can be converted into the Phase vs Time signal by taking the inverse tangent.

Once the Phase vs Time signal is obtained it is compared with an ideal Phase vs Time signal that is internally generated. This comparison generates a Phase Change vs Time graph which is displayed in the graphs section of the front panel. This Phase Change vs Time signal is passed through bandpass filters to isolate 4-300Hz for Low Frequency Phase Jitter and 20-300Hz for High Frequency Phase Jitter. Finally, the difference between maximum and minimum of 99% of the resulting signal is taken to calculate the final result, **Figure 11-109**.
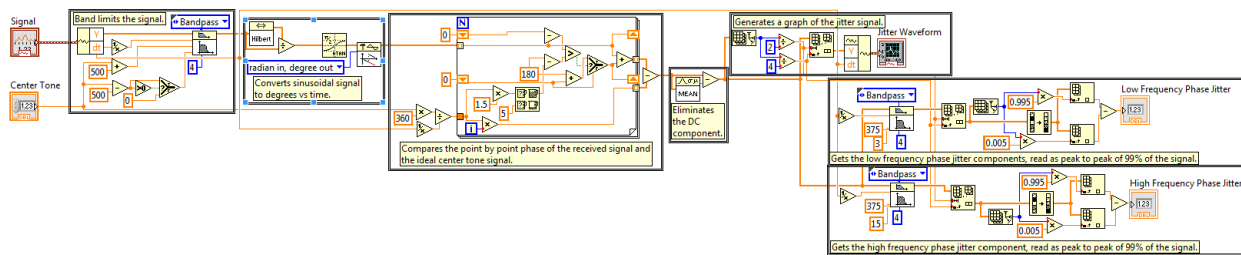
Figure 11-109: Phase Jitter Sub-VI Block Diagram

11.6.4.1.4 Amplitude Jitter

Amplitude Jitter is measured by isolating the holding tone with a narrow bandpass filter. Once the holding tone is isolated, the signal is converted to an Amplitude vs Time signal by taking the square root of the sums of the squares of the Hilbert transformed and original signals. The Hilbert transform of the signal shits all of the frequency components by -90º **[15]**. Since the result is squared the negative can be ignored. This has the effect converting a sine into a cosine at every point in the signal, which when applied to the Pythagorean Theorem gives the amplitude of the signal at every point.

Once the Amplitude vs Time signal is obtained it is subtracted by the average of the signal to get the Amplitude Change vs Time signal. This Amplitude Change vs Time signal is passed through bandpass filters to isolate 4-300Hz for Low Frequency Amplitude Jitter and 20-300Hz for High Frequency Amplitude Jitter. Finally, the resulting signal divided by the original amplitude and the final result is calculated from the maximum deviation off 99% of the signal, **Figure 11-110**.
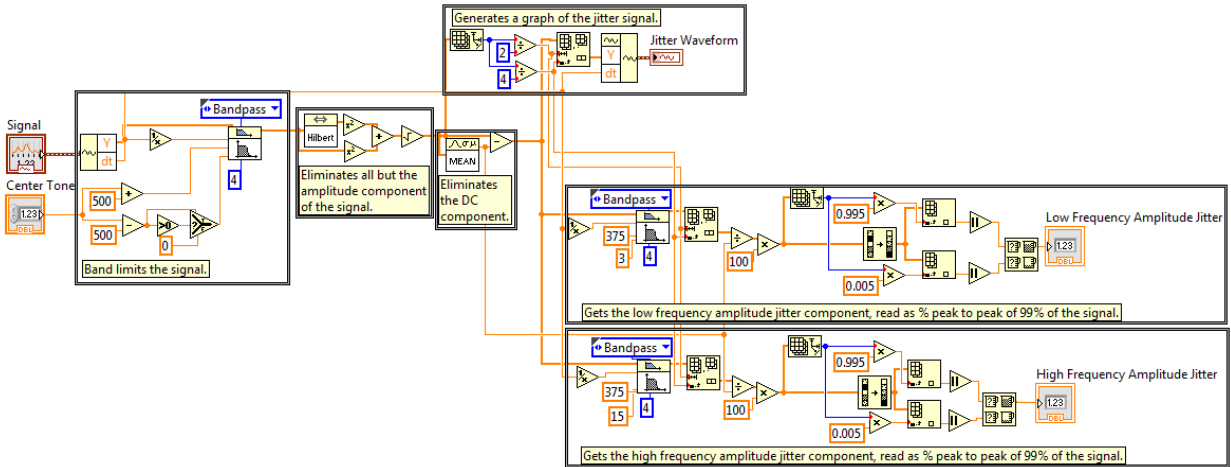
Figure 11-110: Amplitude Jitter Sub-VI Block Diagram

11.6.4.2 Transient Measurements

Transient measurements are made by tests that counts instantaneous changes in a single tone signal over a period of time. Measurement will not begin until the start button is pressed and a valid signal is received, **Figure 11-111**. The received signal must be within 1002 and 1022 Hz if the expected tone is a holding tone or within 5 Hz of expected tone if the expected tone is not a holding tone and must have a signal power greater than -40 dBm, **Figure 11-106**. Impulse Noise, Gain Hits, Phase Hits, and Drop Outs are all transient measurements.
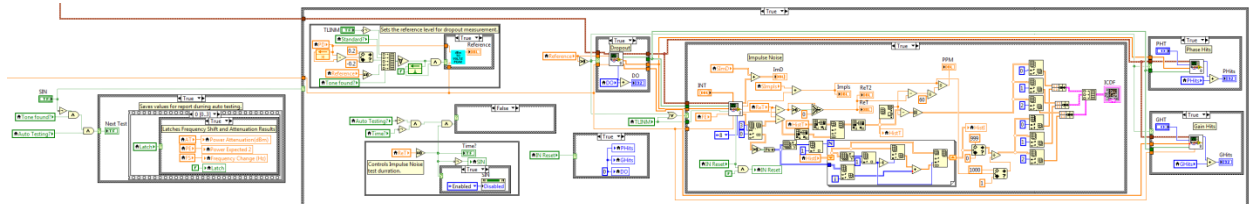
Figure 11-111: Single Tone Measurements Case, Transient Section

11.6.4.2.1 Impulse Noise

The impulse noise test counts the number of noise spikes occur over a period of time, typically 15 minutes. The measurement begins when the received signal and a valid time signal is passed to the impulse noise sub-VI. The valid time array prevents impulses from being counted during a dropout. The impulse noise sub-VI returns a count of the impulses and a time stamp of when they occurred. This information is used to build a cumulative distribution graph of the impulses that were measured as well as calculate the average number of impulses per minute, **Figure 11-112**.

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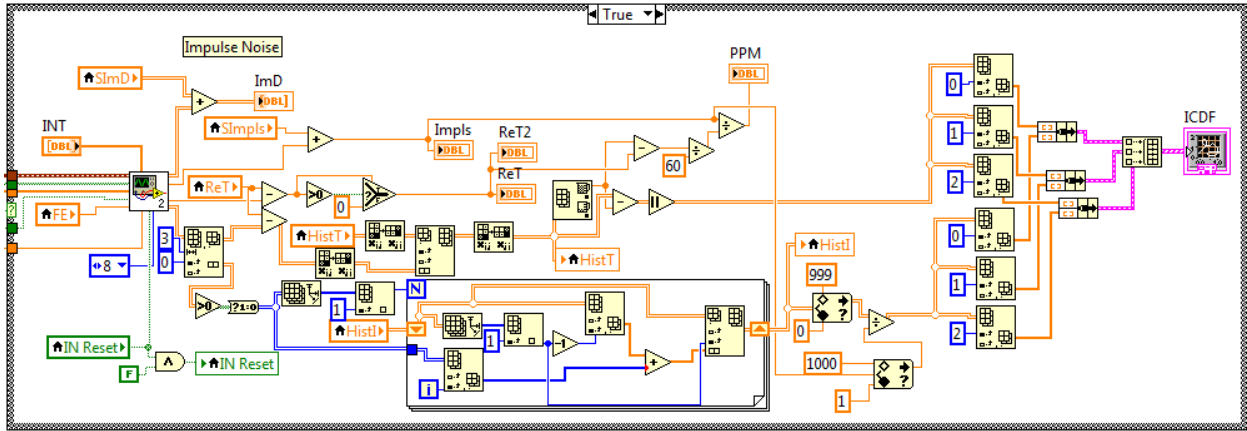
Figure 11-112: Impulse Noise Measurement Section
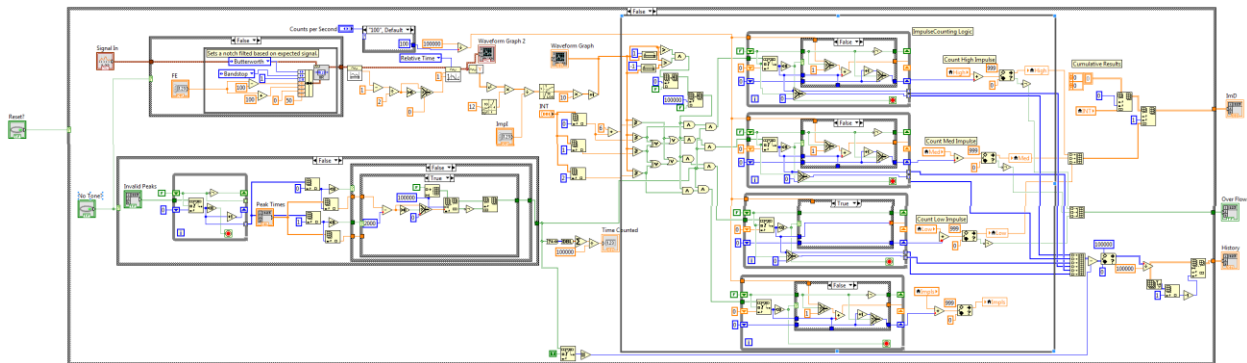
Figure 11-113: Impulse Noise Sub-VI Block Diagram.

The impulse noise sub-VI, **Figure 11-113**, calculates impulse noise in two steps. The first step is to measure the noise amplitude, **Figure 11-114**. To do this the holding tone, if one is used, must be notched out. For impulse noise measurements using a 1004 Hz holding tone the precise 1010 Hz filter defined by IEEE, **Figure 11-115**, must be used to meet the impulse requirements **[1]**. After the filtering is done the middle second of the two second signal capture is isolated and converted to dBrn at every point. The second step is to count each of the impulses that meet the conditions for a given threshold, **Figure 11-116**. An impulse must be exceed the given threshold but not a higher threshold or exceed the highest threshold by 6 dBrn to be counted at any given threshold. Only the maximum point in the pulse is counted and the noise level must drop below the threshold before another pulse can be counted. The number of pulses counted in the second is added to the total.

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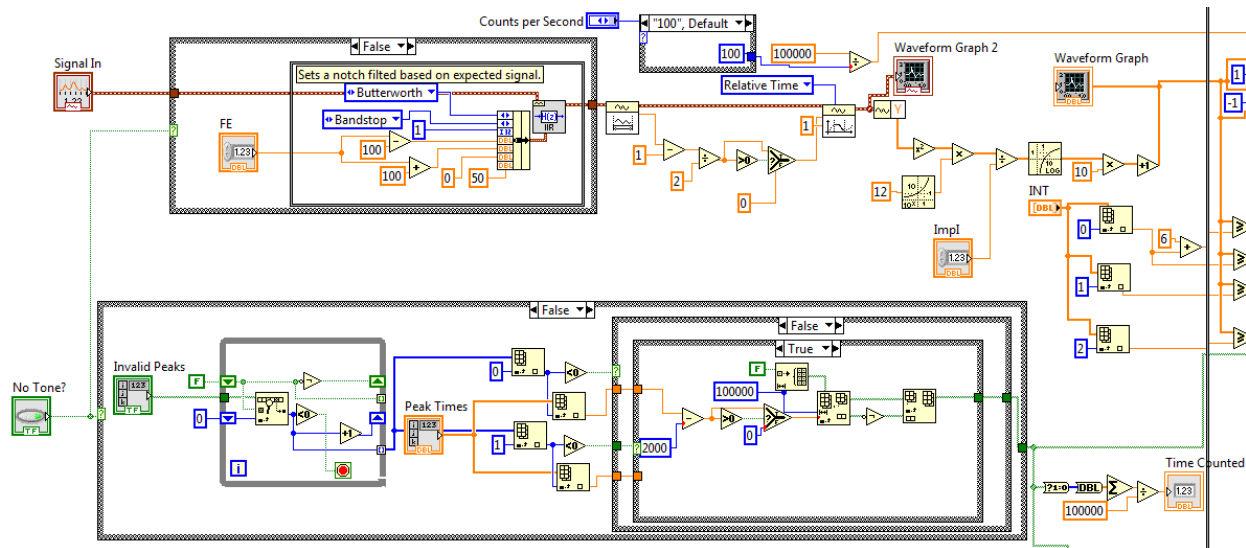
Figure 11-114: Detect Impulse Noise
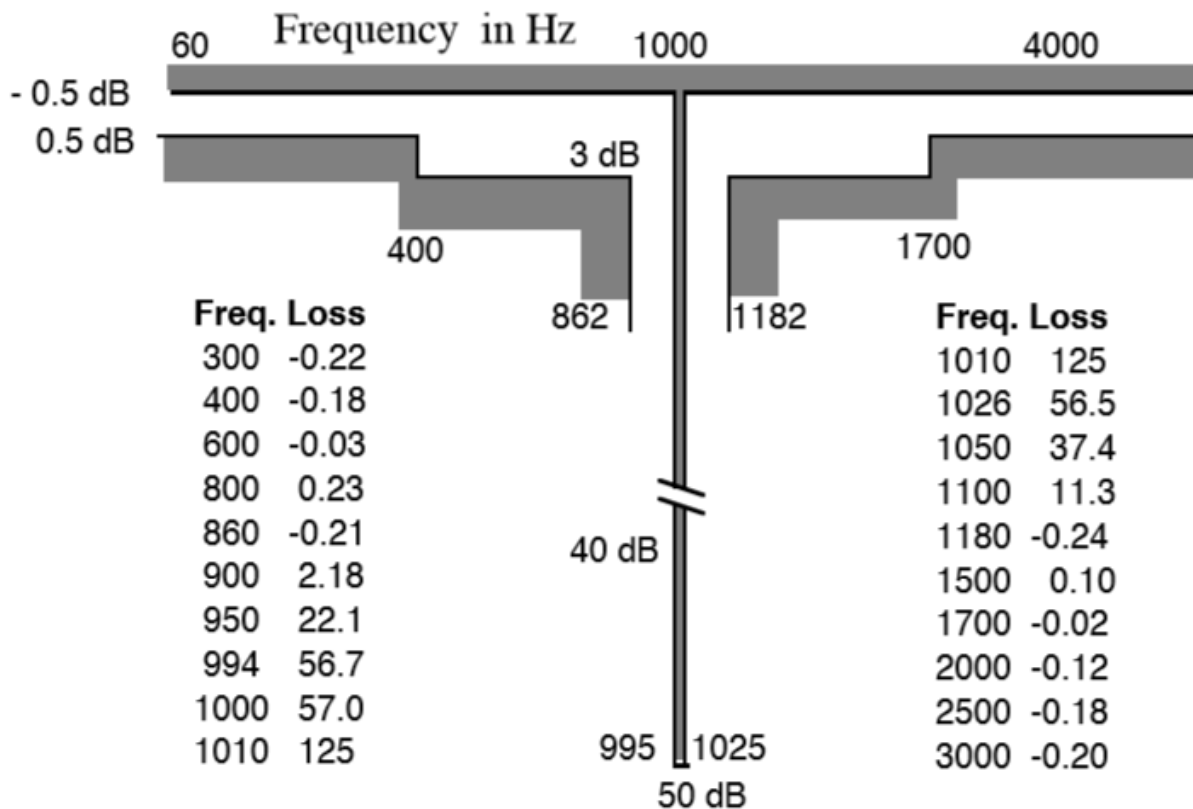
Figure 11-115: Notch Filter.
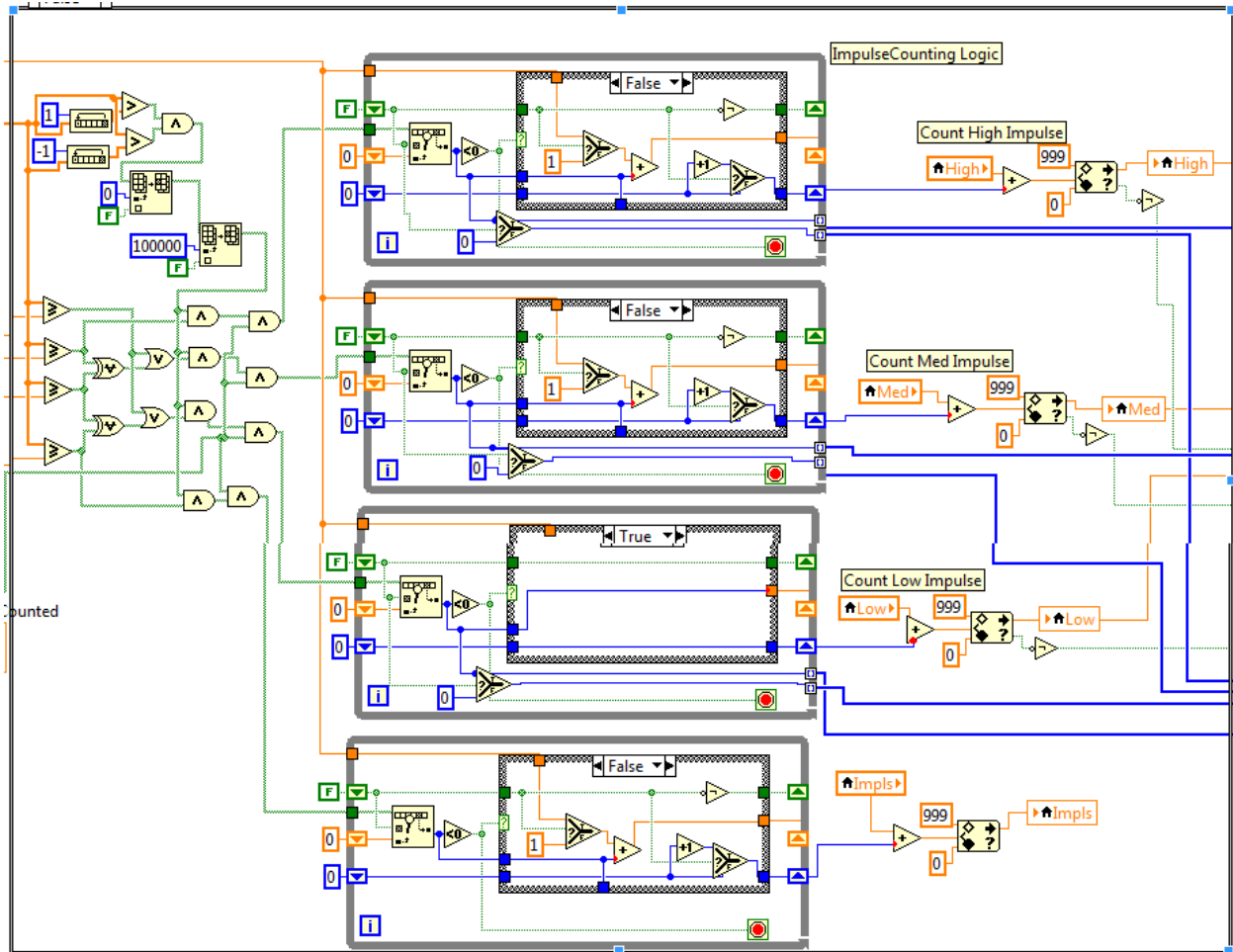
Figure 11-116: Count Impulse Noise

11.6.4.2.2 Gain Hits

Gain Hits are sudden changes to the amplitude of a signal. The gain hit sub-VI, **Figure 11-117**, is passed the signal, the amplitudes of the peaks of the signal, and the peaks that occurred during a dropout. The gain change is measured by comparing the amplitude of a peak with the amplitude of a peak earlier in the signal. The resulting gain change signal is compared to the minimum detectable change for a given threshold to determine if further calculation is needed. If a gain hit is detected the portions of the signal containing the gain hit is broken into smaller segments that all have the amplitude measured and converted into power. The resulting segmented power is compared to the power of a segment before the gain hit. The segments with differences exceeding the threshold are counted to determine is the qualification interval of 3.5 periods of the signal if exceeded before the gain hit is finally counted. The number of gain hits counted is returned to the TIM VI.
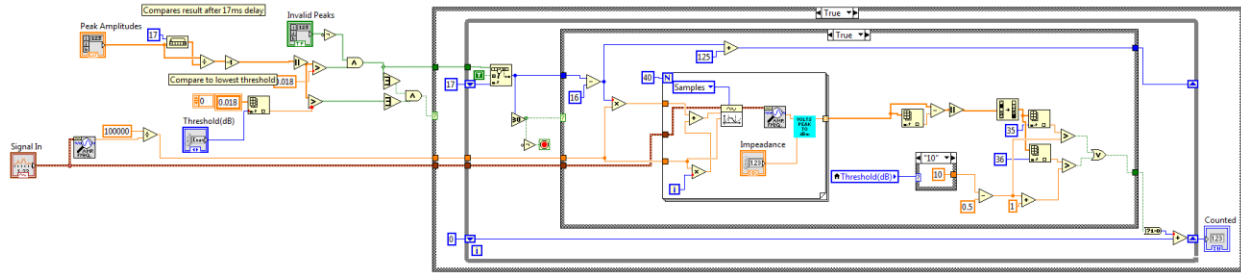
Figure 11-117: Gain Hit Sub-VI Block Diagram.

11.6.4.2.3 Phase Hits

Phase Hits are sudden changes to the phase of a signal. The phase hit sub-VI, **Figure 11-118**, is passed the signal, the times of the peaks of the signal, and the peaks that occurred during a dropout. The phase change is measured by comparing the time of a peak with the time of a peak earlier in the signal. The resulting period change signal is compared to the minimum detectable change for a given threshold to determine if further calculation is needed. If a phase hit is detected the portions of the signal containing the phase hit is broken into smaller segments that all have the phase measured. The resulting segmented phase is compared to the phase of a segment before the gain hit. The segments with differences exceeding the threshold are counted to determine if the qualification interval of 3.5 periods of the signal is exceeded before the gain hit is finally counted. The number of phase hits counted is returned to the TIM VI.
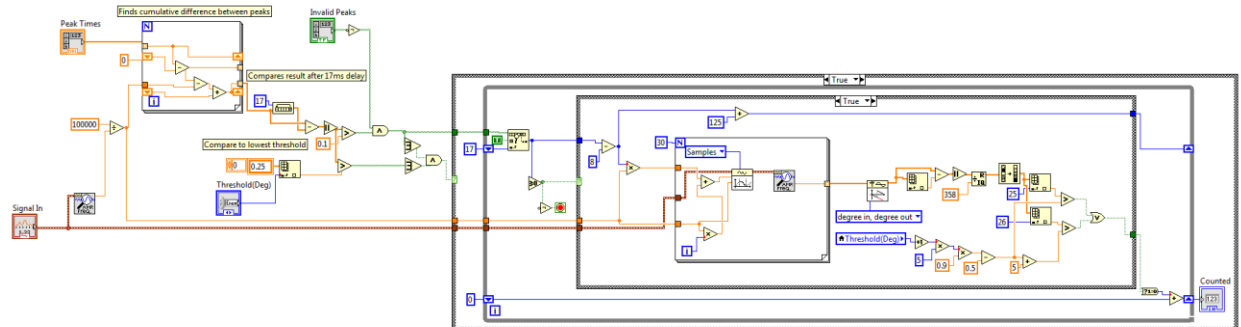
Figure 11-118: Phase Hit Sub-VI Block Diagram

11.6.4.2.4 Drop Outs

Dropout are a loss in power greater than 12 dB compared to the original power of the signal. The drop out sub-VI, **Figure 11-119**, measures the amplitude and time of every peak in the signal. If no peak is detected for 3.5 periods of the holding tone a drop out is immediately counted. If a peak is lower than the original power of the of the holding tone by more than 12 dB the section of the signal containing the drop out is broken into smaller segments that all have the amplitude measured and converted into power. The resulting segmented power is compared to the power of a segment before the drop out. The segments with differences exceeding the threshold are counted to determine if the qualification interval of 3.5 periods of the signal is exceeded before the dropout is finally counted. When

a dropout occurs one second of the signal is marked as a dropout period and no other transients can be measured during that period. Only one count can be made per second.
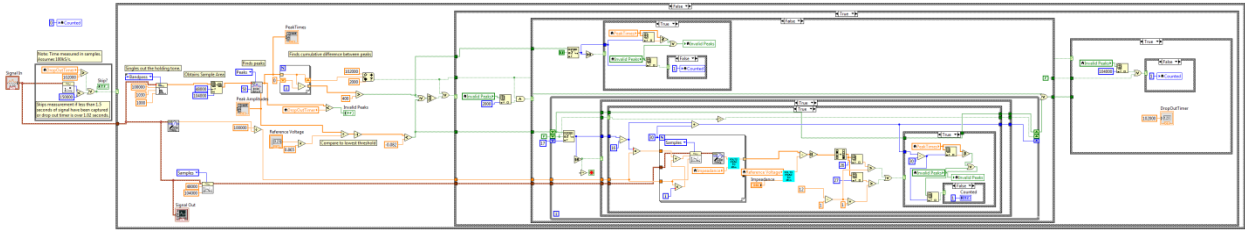
Figure 11-119: Dropout Sub-VI Block Diagram.

11.6.5 Multi-Tone Tests

Multi-tone tests are any tests that use a signal with two or more frequencies during testing. The virtual test set uses the 23-Tone signal for its multitoned testing. The 23-Tone signal uses 23 specific frequencies at specific initial phases as indicated in **Table 11-1 [1]**. The multi-tone tests the virtual test set can perform are Intermodulation Distortion (IMD) and Envelope Delay Distortion (EDD), **Figure 11-120**. When auto-testing the measurements are made when each of the 23-Tones have a power above -50 dBm.

Table 11-1: 23-Tone Test Components

Tone	Δf Mult.	Frequency in Hz	ΔØ Mult.	Phase in deg.	Tone	Δf Mult.	Frequency in Hz	ΔØ Mult.	Phase in deg.
1	13	203.125	14	219.13	13	133	2078.125	13	203.48
2	23	359.375	7	109.57	14	143	2234.375	5	78.26
3	33	515.625	18	281.74	15	153	2390.625	20	313.04
4	43	671.875	1	15.65	16	163	2546.875	12	187.82
5	53	828.125	2	31.30	17	173	2703.125	15	234.78
6	63	984.375	16	250.43	18	183	2859.375	19	297.39
7	73	1140.625	10	156.52	19	193	3015.625	8	125.22
8	83	1296.875	4	62.61	20	203	3171.875	17	266.08
9	93	1453.125	6	93.91	21	213	3328.125	22	344.35
10	103	1609.375	3	46.96	22	223	3484.375	11	172.17
11	113	1765.625	9	140.87	23	233	3640.625	21	328.70
12	123	1921.875	0	0.00	—	—	—	—	—

Figure 11-120: Transmission Impairment Measurement Block Diagram, MultiTone Measurements Case

11.6.5.1 Intermodulation Distortion

IMD occurs when non-linearity in the system generates 2nd and 3rd harmonics from the signal. To perform the intermodulation distortion test the frequency spectrum is broken into 220 15.625Hz wide 'bins' numbered from 13 to 233 spanning from 203.125Hz to 3640.625Hz, **Figure 11-121 [1]**. The 2nd and 3rd IMDs will fall into predictable 'bins' that can be classified by number, **Table 11-2**. The IMD sub-VI, **Figure 11-122**, measures the power in each of the 220 bins and organizes them into 23-Tone, 2nd IMD, 3rd IMD, and noise bins. The total power in the 2nd IMD and 3rd IMD bins that is greater than the average noise power is combined and compared to the total 23-Tone power to obtain the final result.
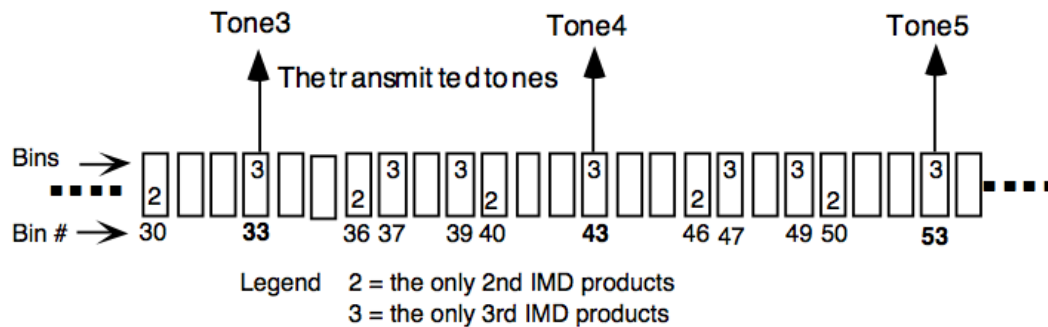
Figure 11-121: Intermodulation Distortion Bins Example

Table 11-2: Bins Containing IMDs.

IMD product	Some examples of the IMD product type	Bin number ending in	Lowest bin	Highest bin	Total bins
2nd	B–A	0	20	220	21
2nd	2A, A+B	6	26	226	21
3rd	2A–B, A+B–C	3	13	233	23
3rd	C-2A, C–A–B	7	17	207	20
3rd	3A, 2A+B, A+B+C	9	39	229	20
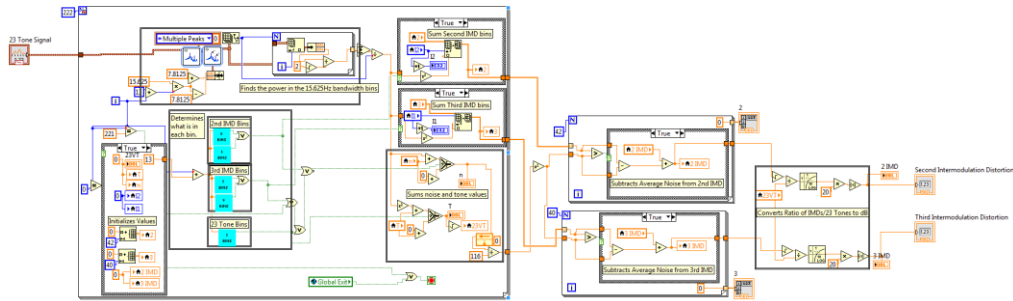
Figure 11-122: Intermodulation Distortion Sub-VI Block Diagram.

11.6.5.2 Envelope Delay Distortion

Envelope Delay Distortion is when frequency components of a signal are delayed by different amounts. To perform the envelope delay distortion test, each of the 23 tones have their phases measured in adjacent tone pairs, **Figure 11-123**. The phase difference of the tone pair is then compared to the initial phase difference of the 23 tone signal and the difference measured. The result in degrees is converted to radians and normalized by taking the cosine and arccosine or the result, this prevents phase differences of approximately 360° from being measured as abnormally high. The resulting relative phase difference is converted to delay in microseconds by dividing by the frequency difference, in radians per second. Finally, each of the measured delays are subtracted by the smallest delay with the corresponding frequency pair represented as the average between the frequencies.
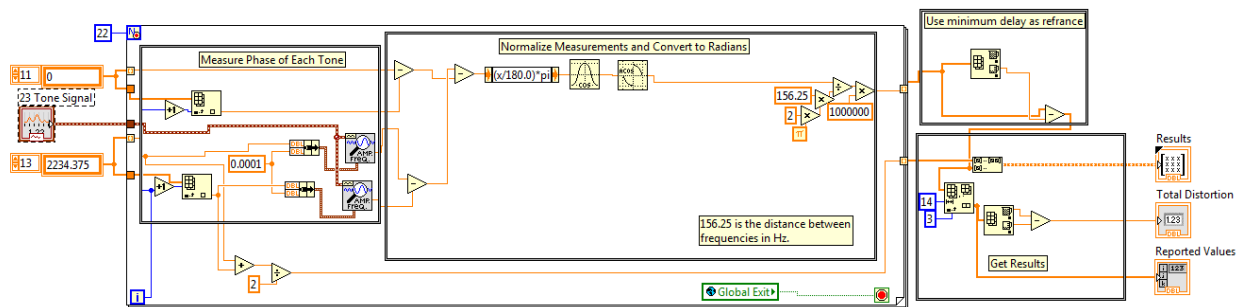
Figure 11-123: Envelope Delay Distortion Sub-VI Block Diagram

11.6.6 Series Tone Tests

Series Tone tests are tests that are performed by sending a sequence of single tone signals. The virtual test set cycles through the tones using a queue that requests a new tone from the function generator every three seconds, **Figure 11-124**. The queue will continue to re-queue the tones until every tone in the series has been received at least once. The queue will then remove the tones from the queue one by one as it cycles through the tones one more time. The series tone queue is used for 3-Tone Slope, Attenuation Distortion, and Echo Return Loss (ERL) tests.

Figure 11-124: Series Tone Test Queue.

11.6.6.1 3 Tone Slope and Attenuation Distortion

The 3 Tone Slope and Attenuation Distortion tests are both tests that measure attenuation to tones in the 304-3004 Hz range relative to each other in 3 and 27 steps respectively. The virtual test set scans the list of frequencies in the series for one that matches the received frequency. If a match is found the frequency is marked as detected and the power of the frequency is compared to the recorded power and the higher power level is kept, **Figure 11-125**.

Figure 11-125: Transmission Impairment Measurement Block Diagram, Series Tone Test Case

11.6.6.2 Echo Return Loss

Echo Return Loss is a series tone measurement that measures how much of a signal is reflected back to the virtual test set. The virtual test set performs echo return loss measurements by transmitting a series of tones covering the range of signals specified by the IEEE Echo Return Loss, Singing Return Loss Low, and Singing Return Loss High filters; **Tables 11-3, 11-4, and 11-5** respectively **[1]**. When the series tone test is indicated to be an Echo Return Loss test, the ideal series is passed through the filters in the 'Start Series Tone Test' event case and the results are saved. As the tones from the series are received they are passed through each of the filters and the results are saved alongside the attenuation slope measurements, **Figure 11-126**. Once the entire series is found the ideal results are compared to the actual results and covered to a dB result associated with each filter.

Table 11-3: Echo Return Loss Filter Specifications.

Frequency (Hz)	Relative loss (dB)	Tolerance (dB)
≤ 200	≥ 30.0	—
300	21.8	± 2.3
560	3.0	± 0.4
750	0.2	± 0.2
1000	0.0	± 0.1
1500	0.1	± 0.2
1965	3.0	± 0.4
2400	10.9	± 1.2
3000	22.9	±3.0
4000	42.6	± 5.0
≥ 5000	≥ 45.0	—

Table 11-4: Singing Return Loss Low Filter Specifications.

Frequency (Hz)	Relative loss (dB)	Tolerance (dB)
≤ 100	≥ 20	—
120	20	± 3.0
200	9.5	± 1.1
260	3	± 0.5
360	0	± 0.2
500	3	± 0.5
650	10	± 1.2
1000	20	± 3.0
≥ 1200	≥ 20	—

Table 11-5: Singing Return Loss High Filter Specifications

Frequency (Hz)	Relative Loss (dB)	Tolerance (dB)
≤ 1000	≥ 30	—
1300	30	±4.0
2000	11.5	±1.3
2200	3	± 0.5
2700	0	± 0.2
3400	3	± 0.5
3700	10.9	± 1.3
5700	30	± 4.0
≥ 6000	≥ 30	—
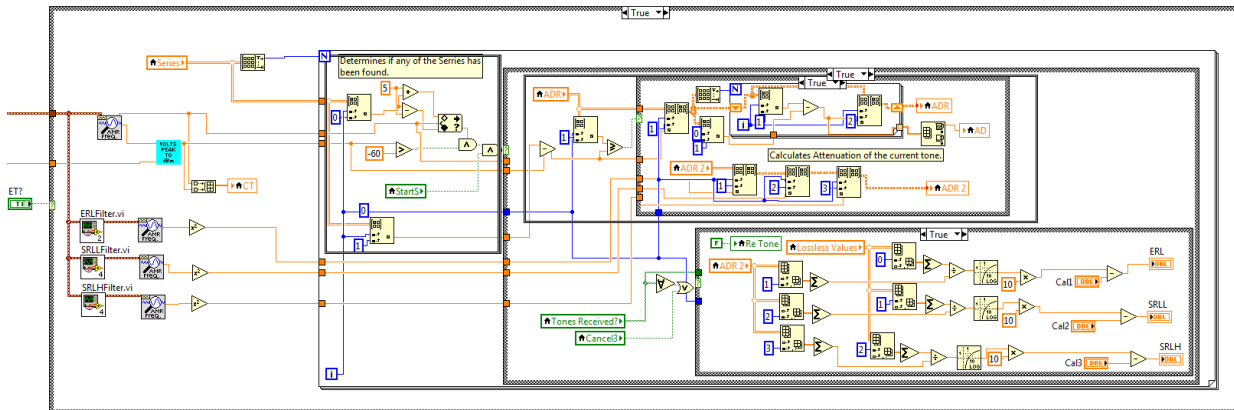
Figure 11-126: Transmission Impairment Measurement Block Diagram, Echo Return Loss Test Case

11.6.7 Auto Test

When performing a series of tests for the sake of filling out a report the process is performed according to **Figure 11-127**. This process is typically performed manually but the virtual test set allow this process to be performed automatically. This is done using a queue that switches the measurement section to the test being performed, requesting the necessary signal from the function generator, setting the necessary test parameters, and waiting for the test to indicate it has finished, **Figure 11-128**.

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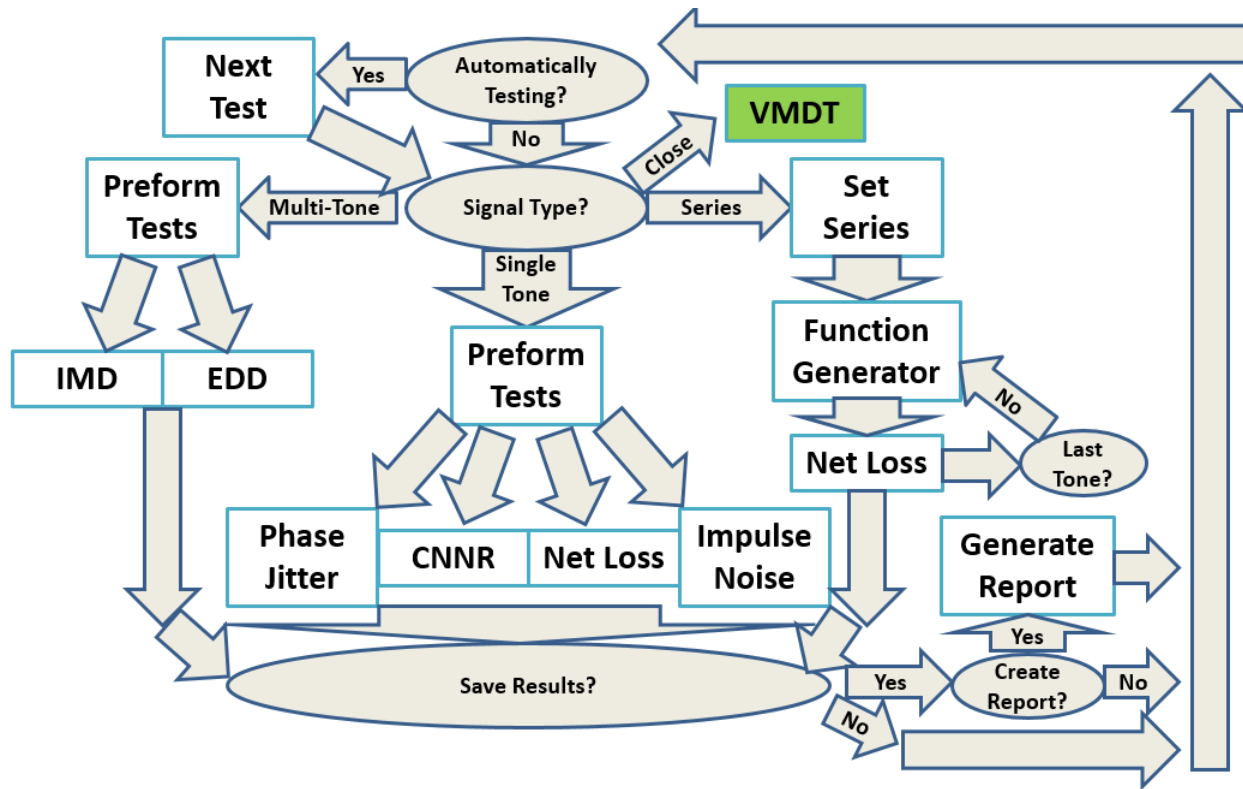
Figure 11-127: Transmission Impairment Program Flow Chart.
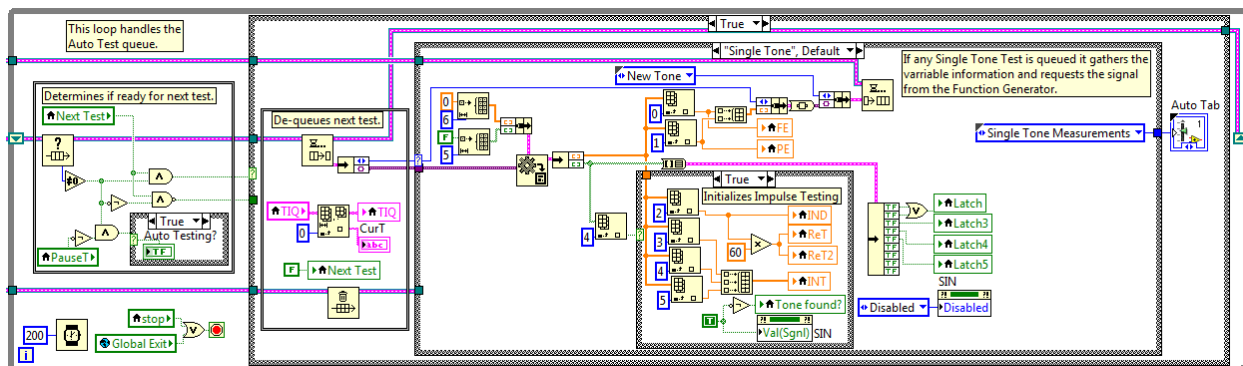
Figure 11-128: Auto Test Queue

11.6.8 Report Generation

The report generation case of the TIM measurement section functions in 4 steps, **Figure 11-129**. First, each value used in the report is compared to the tolerance associated with the report and creates an 'out of tolerance' array. Second, all of the test results are put into a 'test results' array. Third, the information on the report tab is entered into a 'report information' array. Finally the information is passed to a custom sub-VI for each the selected report type. This sub-VI uses Microsoft word bookmarks to enter information into the correct place in the report template any report value marked as 'out of tolerance' is highlighted in red.
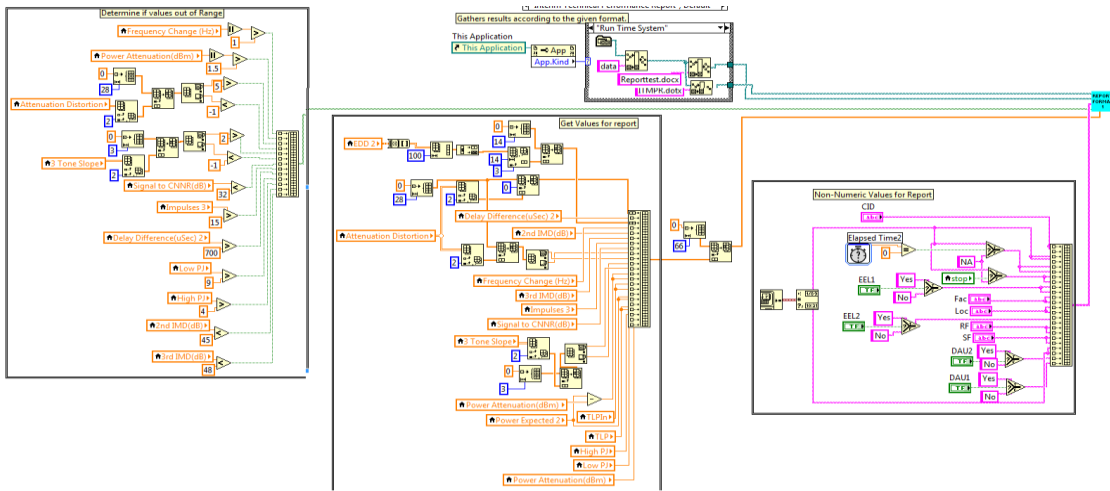
Figure 11-129: Transmission Impairment Measurement Block Diagram, Report Generation Case

11.7 Oscilloscope Block Diagram

The block diagram of the Oscilloscope VI can be broken into different sections: event case, queue, and function sections, **Figure 11-130**.
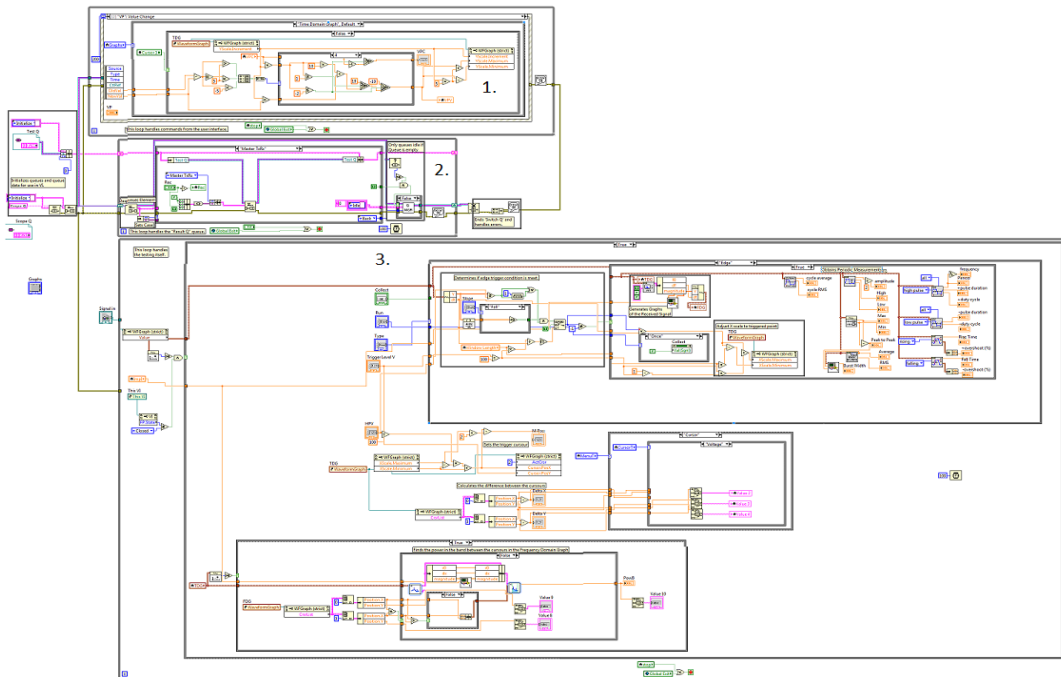
Figure 11-130: Oscilloscope Block Diagram

11.7.1 Oscilloscope Event Case

The Oscilloscope's event case handles the commands from the user. The primary functions of the Oscilloscope's event case is to cause the user controls to mimic the mechanical actions and display controls of a standalone oscilloscope. The Oscilloscope has the following unique event cases:

1. Adjust vertical position.
2. Adjust horizontal position.
3. Zero horizontal position.
4. Center Trigger Level.
5. Run continuously.
6. Run once.
7. Switch to measurement.
8. Change Volts per Division.
9. Change Seconds per Division.
10. Auto Set.
11. Switch to trigger menu.
12. Switch to cursor menu.
13. Button 1.
14. Button 2.
15. Switch to spectrum analyzer.
16. Adjust trigger level.

Adjust vertical position: This event is activated when the vertical position knob is turned. This event case has three functions it performs: mimic a physical knob, adjust the vertical position of the time domain graph, and adjust the horizontal position of cursor 2. The knob control mimics a physical knob by allowing the knob control to make a complete rotation. The knob position returns a value between 0 and 10 and an internal counter adds multiples of ten to account for multiple rotations. A check is performed with every change to ensure the limits of the control are maintained, **Figure 11-131**. The vertical position of the time domain graph is controlled by continuously having the knob control adjust the limits of the y-scale on the time domain graph. The rate of change for the scale scales with the current volts per division setting and is bound between +/- 10V. Finally, the horizontal position of cursor 2 is controlled by using the knob to adjust the relative position of the cursor. When the cursor control button is activated and the knob is turned the current graphs x-scale limits are acquired and the cursor position is determined as a percent of the difference between them.
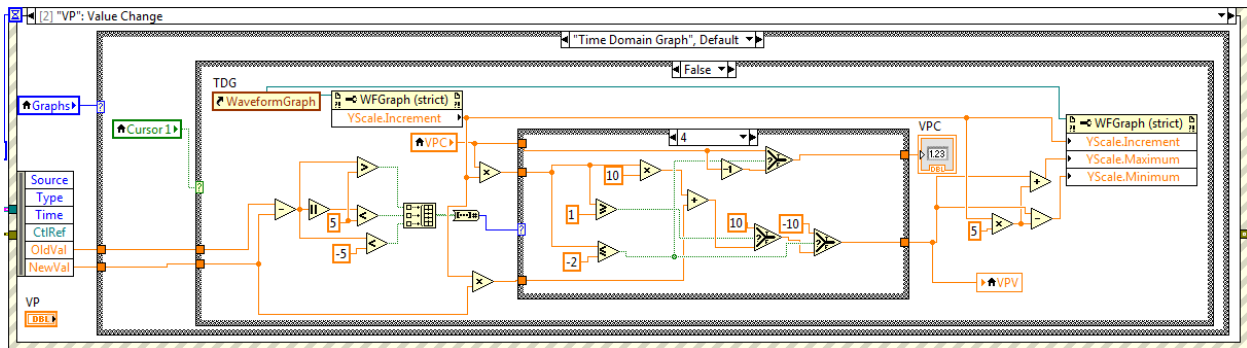
Figure 11-131: Oscilloscope Event Case, Adjust vertical position Event

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Adjust horizontal position: This event is activated when the horizontal position knob is turned. This event case has three functions it performs: mimic a physical knob, adjust the horizontal position of the trigger cursor, and adjust the horizontal position of cursor 1. The knob control mimics a physical knob by allowing the knob control to make a complete rotation. The knob position returns a value between 0 and 10 and an internal counter adds multiples of ten to account for multiple rotations. A check is performed with every change to ensure the limits of the control are maintained, **Figure 11-132**. The horizontal position of the trigger cursor is controlled by continuously having the knob control adjust the internal horizontal position value linked to the trigger level cursor. The knob controls the cursor position as a percentage within the x-scale limits of the time domain graph. Finally, the horizontal position of cursor 1 is controlled by using the knob to adjust the relative position of the cursor. When the cursor control button is activated and the knob is turned the current graphs x-scale limits are acquired and the cursor position is determined as a percent of the difference between them.
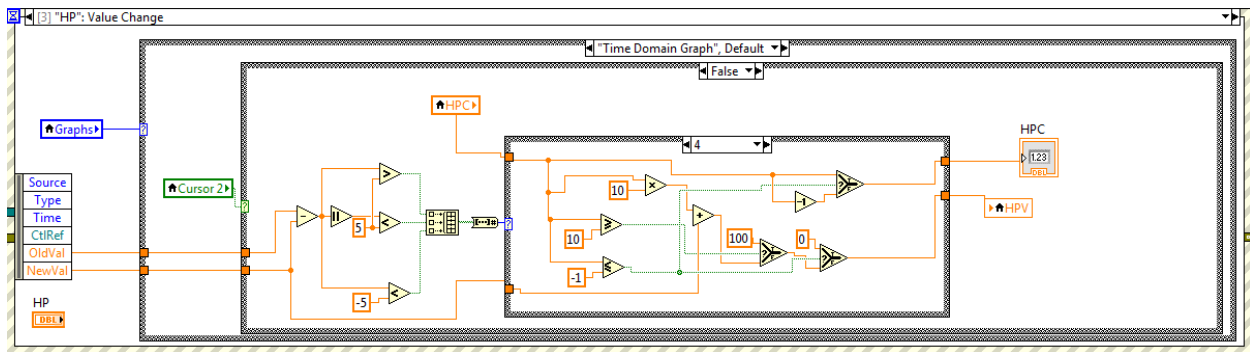
Figure 11-132: Oscilloscope Event Case, Adjust horizontal position Event

Zero horizontal position: This event is activated when the Set To Zero button is pressed. This event simply sets the horizontal position controls to their default values.

Center Trigger Level: This event is activated when the Set To 50% button is clicked. This event simply sets the trigger cursor back to its default position.

Run continuously: This event is activated when the Run/Stop button is clicked. This event requests for the virtual test set's receive function to be turned on and sets the oscilloscope to capture any signal that meets the trigger conditions.

Run once: This event is activated when the 'Single Seq' button is clicked. This event requests for the virtual test set's receive function to be turned on and sets the oscilloscope to capture only the first wave form that meets the trigger conditions.

Switch to measurement: This event activates when the Measurements button is clicked. This switches to the periodic measurements display or, if already displaying periodic measurements, returns to the time domain graph display, **Figure 11-133**.

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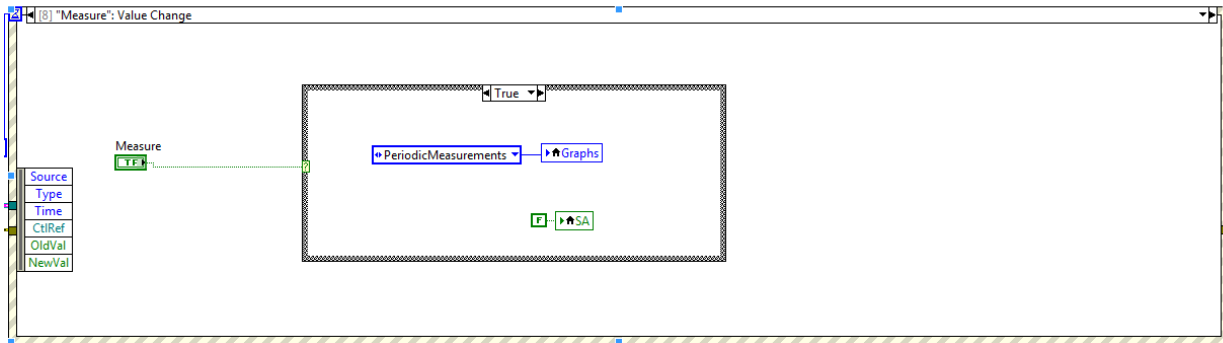
Figure 11-133: Oscilloscope Event Case, Switch to measurement Event

Change Volts per Division: This event is activated when the 'Volts Per Division' knob is turned. This event case has two functions it performs: mimic a physical knob, adjust the volts per division of the display. The knob control mimics a physical knob by jumping to, clicking to, one of four positions. Each click clockwise increments the volts per division counter, up to the maximum, while each click counter-clockwise decrements the volts per division counter, down to the minimum, each number of the volts per division counter corresponds to a volts per division setting, **Figure 11-134**. Finally, the Volts Per Division setting is applied by adjusting the time domain graph's y-scale increment, maximum, and minimum values to the appropriate levels based on the current vertical position of the graph.
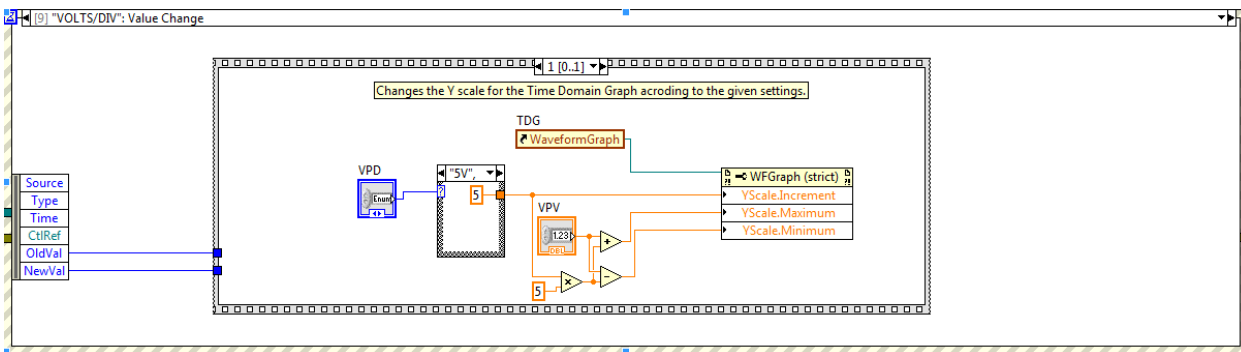
Figure 11-134: Oscilloscope Event Case, Change Volts per Division Event

Change Seconds per Division: This event is activated when the 'Seconds Per Division knob' is turned. This event case has two functions it performs: mimic a physical knob, adjust the second per division of the display. The knob control mimics a physical knob by jumping to, clicking to, one of four positions. Each click clockwise increments the volts per division counter, up to the maximum, while each click counter-clockwise decrements the volts per division counter, down to the minimum, each number of the volts per division counter corresponds to a seconds per division setting, **Figure 11-135**. Finally, the Seconds Per Division setting is applied by adjusting the time domain graph's x-scale increment, maximum, and minimum values to the appropriate levels while calculating the window length.

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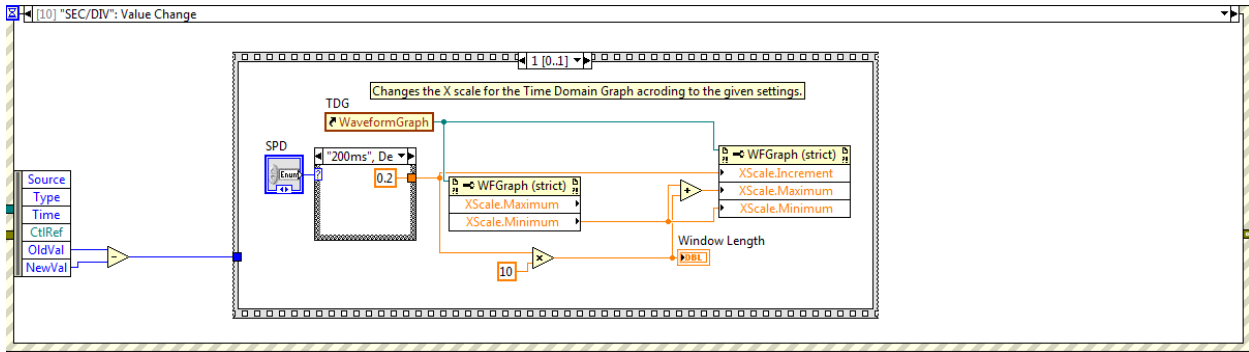
Figure 11-135: Oscilloscope Event Case, Change Seconds per Division Event

Auto Set: This event case is activated when the Auto Set button is clicked. The appropriate Volts Per Division setting is calculated by finding the smallest volts per division setting that will fit the max amplitude of the periodic function within the y-scale of the time domain graph, **Figure 11-136**. The Seconds Per Division setting is calculated by finding the smallest seconds per division setting that will fit the period of the function into the x-scale of the time domain graph. Once the appropriate settings are calculated they are applied.
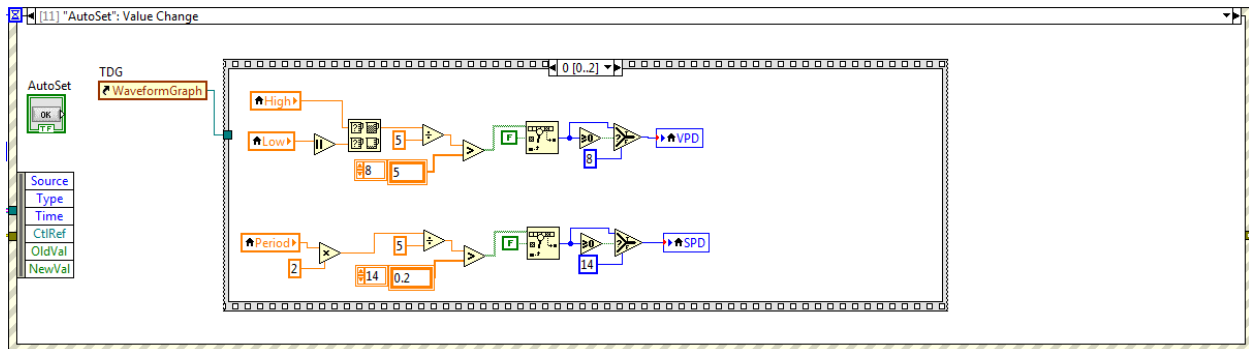
Figure 11-136: Oscilloscope Event Case, Auto Set Event

Switch to trigger menu: This event is activated when the Trig Menu button is clicked. This event changes the cursor menu displays and controls to display and control the trigger level settings, **Figure 11-137**.
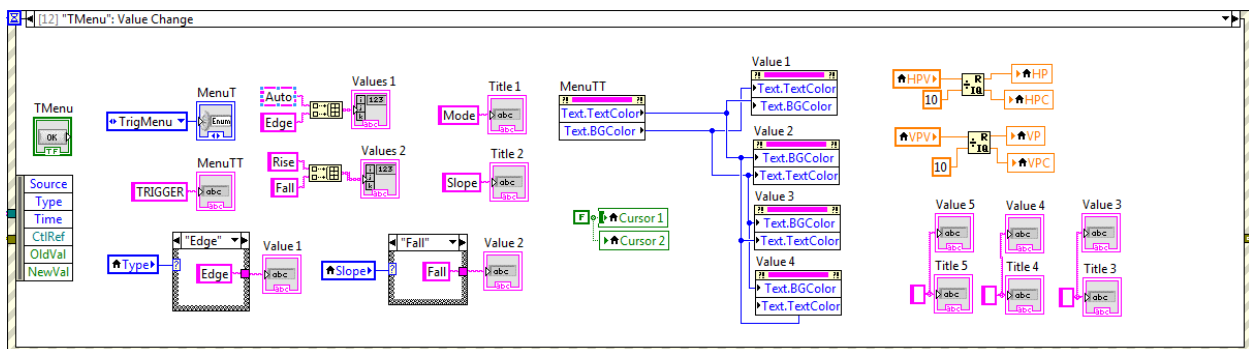
Figure 11-137: Oscilloscope Event Case, Switch to trigger menu Event

Switch to cursor menu: This event is triggered when the Cursor button is clicked. This event changes the cursor menu displays and controls to display and control the cursor settings, **Figure 11-138**. This event also sets the horizontal and vertical control knobs to cursor control mode.
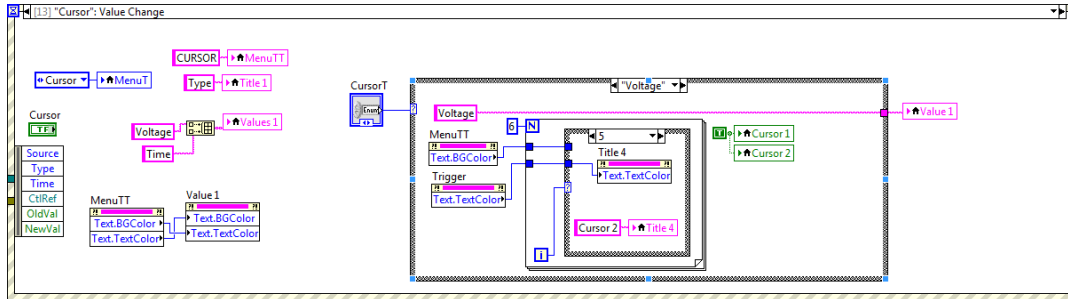
Figure 11-138: Oscilloscope Event Case, Switch to cursor menu Event

Button 1: This event is activated when the first cursor menu button is pressed. If the cursor menu is set to control the trigger level, this event toggles the trigger between edge and automatic mode, **Figure 11-139**. If the cursor menu is set to control the cursors, this event toggles the cursors between displaying the cursor time or voltage positions and measurements.
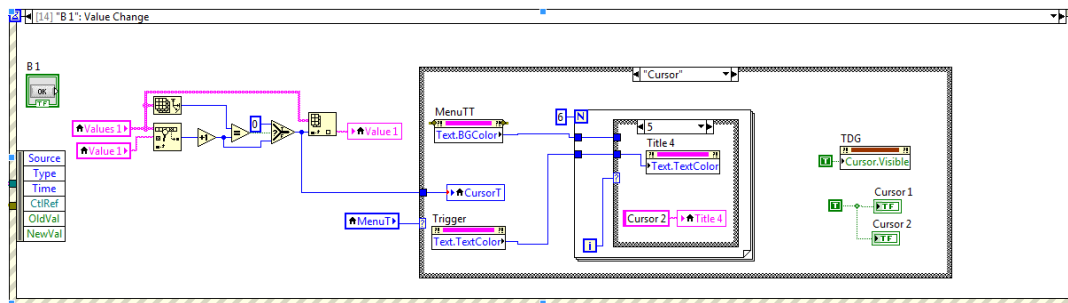
Figure 11-139: Oscilloscope Event Case, Button 1 Event

Button 2: This event is activated when the second cursor menu button is pressed. If the cursor menu is set to control the trigger level, this event toggles the trigger between activating on a rising of falling edge, **Figure 11-140**.
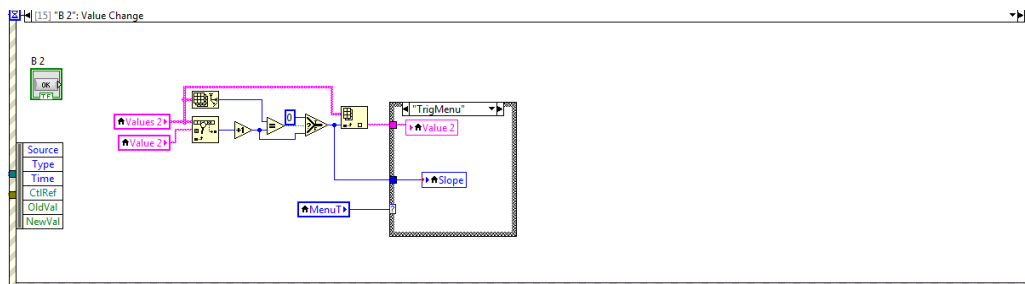
Figure 11-140: Oscilloscope Event Case, Button 2 Event

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Switch to spectrum analyzer: This event is activated when the Spectrum Analyzer button is clicked. This event switches the display to the frequency domain graph or, if the frequency domain graph is already being displayed, switches back to the time domain graph, **Figure 11-141**.
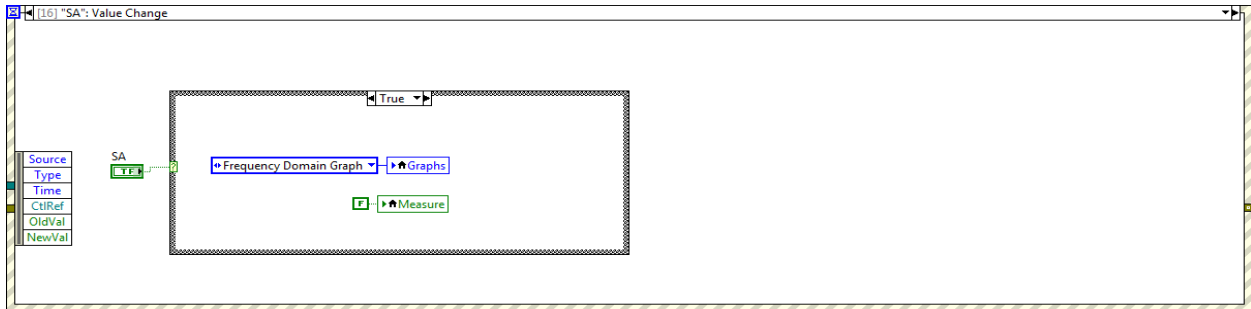
Figure 11-141: Oscilloscope Event Case, Switch to spectrum analyzer Event

Adjust trigger level: This event is activated when the Trigger Level knob is turned. This event case has two functions it performs: mimic a physical knob, adjust the trigger level voltage. The knob control mimics a physical knob by allowing the knob control to make a complete rotation. The knob position returns a value between 0 and 10 and an internal counter adds multiples of ten to account for multiple rotations, **Figure 11-142**. A check is performed with every change to ensure the limits of the control are maintained. The trigger level is bound between +/- 10V and the current trigger level as well as the falling or rising indicator is display on the trigger level indicator.
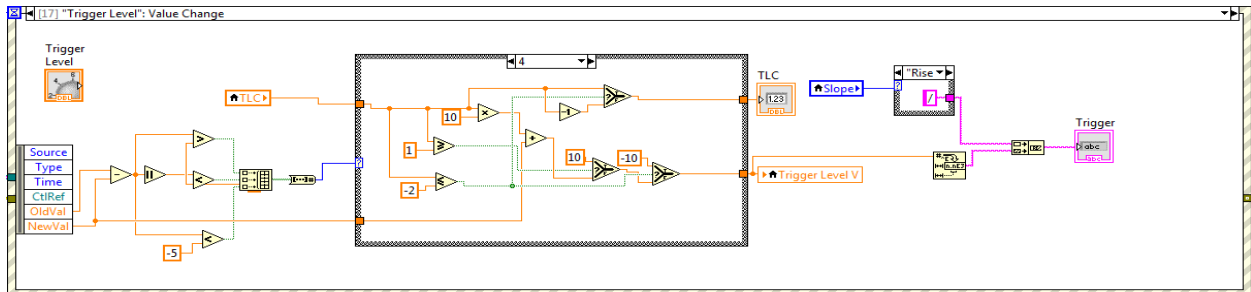
Figure 11-142: Oscilloscope Event Case, Adjust trigger level Event

11.7.2 Oscilloscope Queue.

The Oscilloscope's Queue Loop handles all of the commands passed to the VI and passes commands to other VI's. The primary function of the Oscilloscope Queue Loop is to turn on and off the receive function of the virtual test set. The Oscilloscope's Queue Loop does not process any unique commands.

11.7.3 Oscilloscope Function Section

The Oscilloscope's function section can be broken into three sub sections. The first section generates the time domain graph and performs periodic measurements, **Figure 11-143**. If the oscilloscope is set for automatic triggering it will capture any signal it detects. If the oscilloscope is set for edge triggering it will only capture signals that meet the trigger conditions. Regardless of the method

of the capture, the signal is displayed and a series of periodic measurements are made and displayed when the user presses the 'measure' button, **Table 11-6**. The second section uses references to control the graph's cursors as well as display the cursor values and the values between them when oscilloscope is in cursor mode, **Figure 11-144**. The third and final section generates a Frequency Domain Graph when the user selects 'Spectrum Analyzer' and calculates the power total power located in the band between the cursors, **Figure 11-145**.
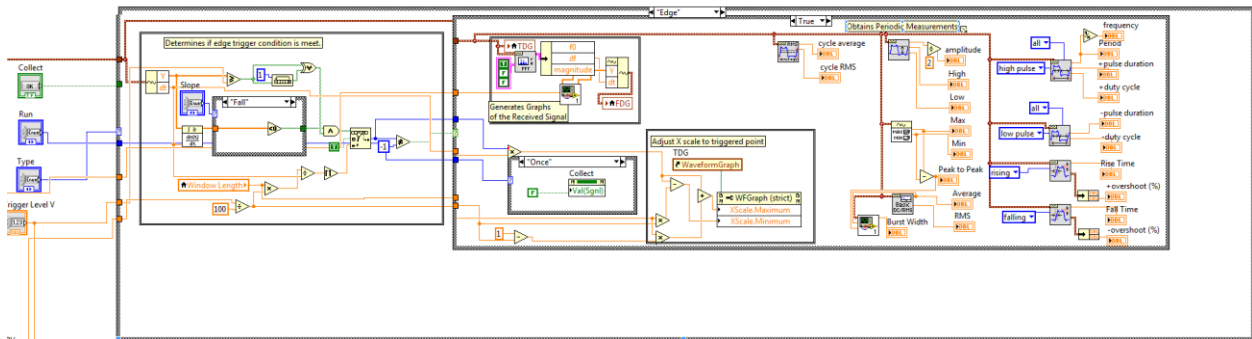
Figure 11-143: Oscilloscope Block Diagram, Periodic Measurements Section.

Table 11-6: Oscilloscope Periodic Measurements.

Measurement	Description
Amplitude	The peak amplitude.
High	The maximum steady amplitude of a square waveform.
Low	The minimum steady amplitude of a square waveform.
Max	The highest point of the signal.
Min	The lowest point of the signal.
Peak to Peak	The difference between the highest and lowest points.
Cycle Average	The average value per period of the signal.
Cycle RMS	The Root Mean Square per period of the signal.
Average	The average value of the full signal.
RMS	The Root Mean Square of the full signal.
Burst Width	The duration of a burst signal.
Frequency	The number of times the signal repeats per second.
Period	The time before a signal repeats.
+Pulse Duration	The duration of the high pulse in a square wave.
+Duty Cycle	The percent of time a square wave is high.
-Pulse Duration	The duration of the low pulse in a square wave.
-Duty Cycle	The percent of time a square wave is low.
Rise Time	The time it takes for a square wave to transition from low to high.
+Overshoot	The percent of the amplitude is overshoot in a square wave high pulse.
Fall Time	The time it takes for a square wave to transition from high to low.
-Overshoot	The percent of the amplitude is overshoot in a square wave low pulse.
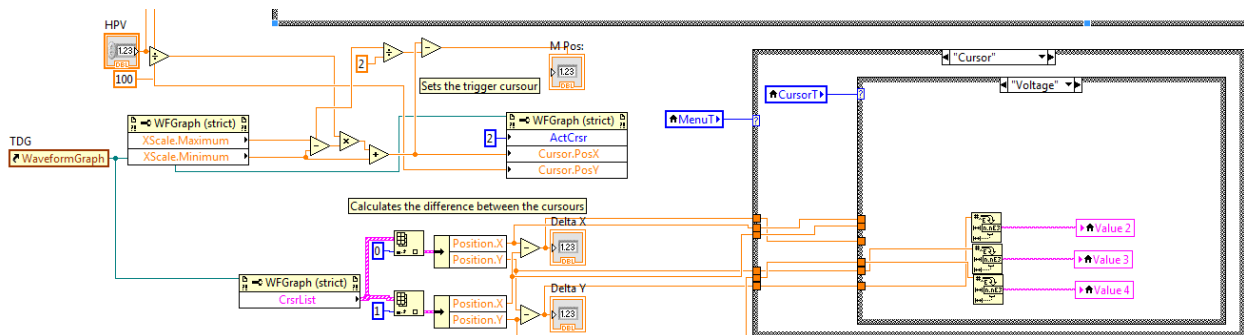
Figure 11-144: Oscilloscope Block Diagram, Cursor Measurements Section.
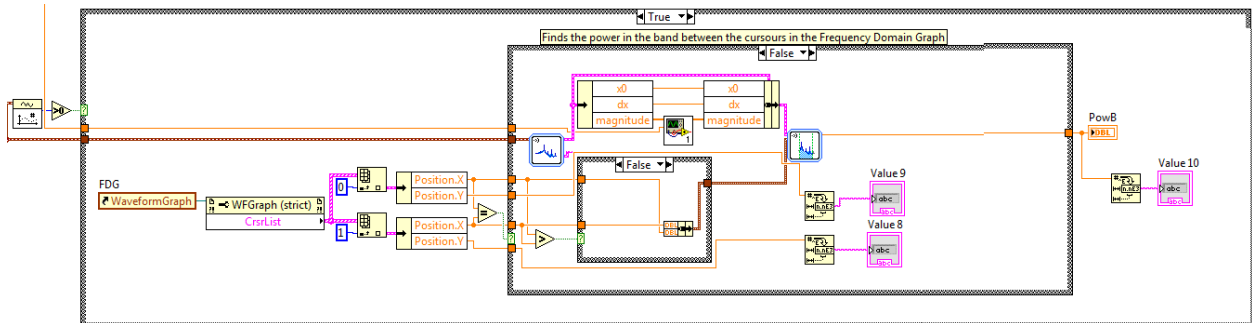
Figure 11-145: Oscilloscope Block Diagram, Spectrum Analyzer Section

11.8 Selective Signaling Block Diagram

Selective signaling functionally performs transmit and receive independently as shown in, **Figure 11-146**. The block diagram of the Selective Signaling VI can be broken into different sections: event case, queue, transmit, and receive sections, **Figure 11-147**.
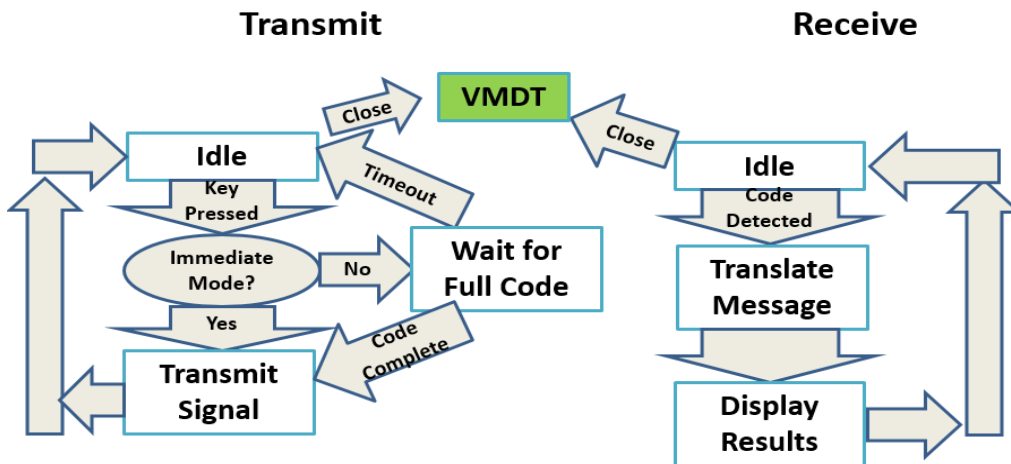
Figure 11-146: Selective Signaling Program Flow.
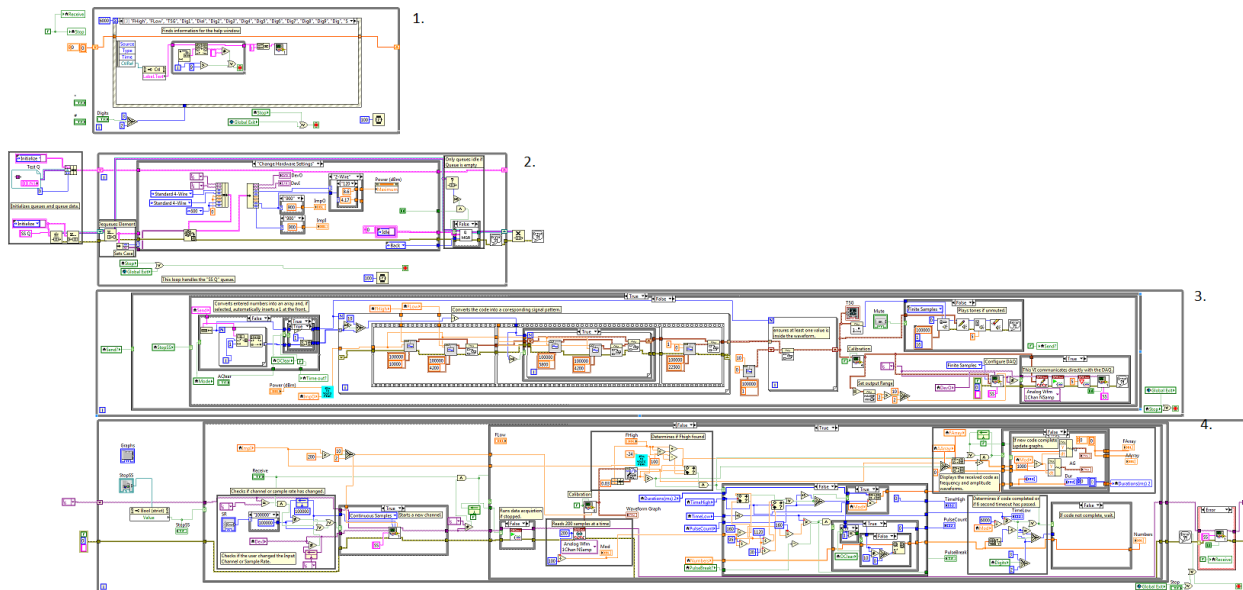
Figure 11-147: Selective Signaling Block Diagram

11.8.1 Selective Signaling Event Case

The Selective Signaling's event case handles the commands from the user. The primary functions of the Selective Signaling's event case is to handle number dialing for selective signaling signals. The Selective Signaling function has the following unique event cases:

1. Allow One Clear.
2. Number Buttons.

Allow One Clear: This event makes the 'Auto Clear' option available only when the 'One Clear' option is active, **Figure 11-148**.
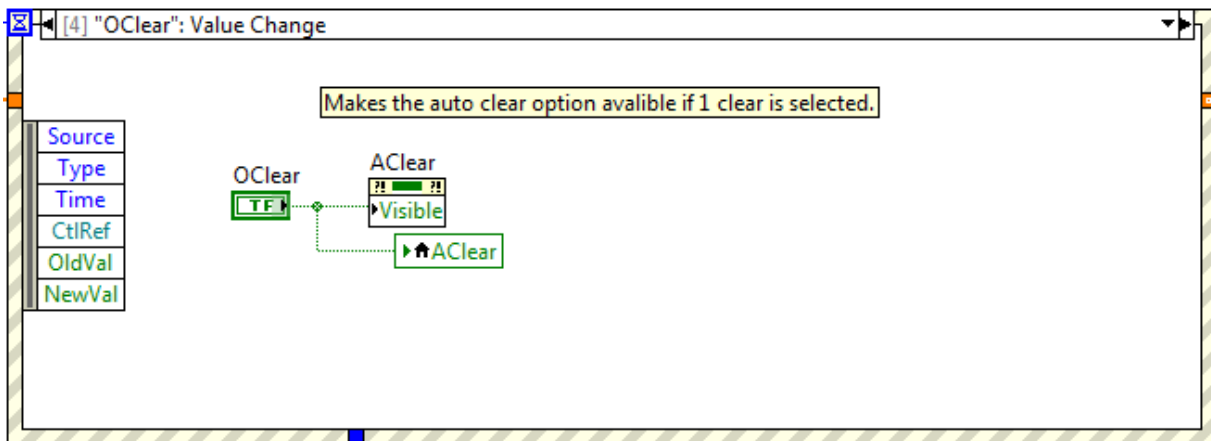
Figure 11-148: Selective Signaling Event Case, Allow One Clear Event.

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Number Buttons: Each of the event cases handling the number buttons function the same way, **Figure 11-149**. When a number button is pressed it is stored in an array of dialed numbers. If the 'Immediate' option is active the selected number is transmitted immediately. If the 'Immediate' option is not active the dialed number array is checked to verify if the appropriate number of digits have been entered, 2 or 3 depending on the current setting. If the appropriate number of digits have been dialed the numbers are all sent in a single burst, otherwise the dialed digits are held until more digits are entered. If no new numbers have been dialed for 6 seconds the dialed number array is automatically cleared by the idle event case.
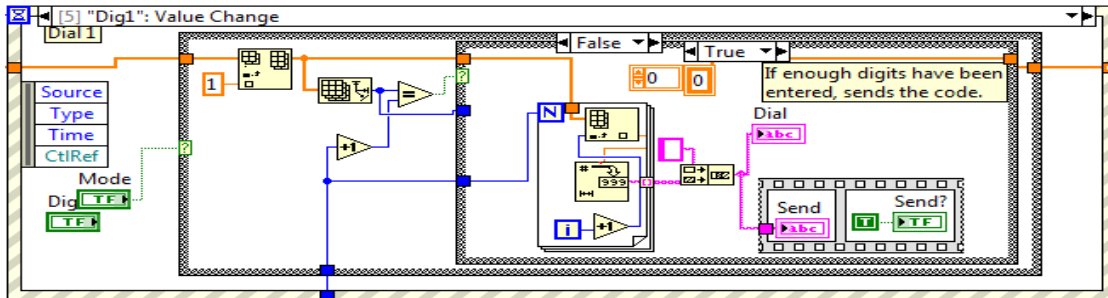
Figure 11-149: Selective Signaling Event Case, Number Buttons Event Example.

11.8.2 Selective Signaling Queue.

The Selective Signaling's Queue Loop handles all of the commands passed to the VI and passes commands to other VI's. The primary function of the Selective Signaling's Queue Loop is to turn on and off the receive function of the virtual test set. The Selective Signaling's Queue Loop does not process any unique commands.

11.8.3 Selective Signaling Transmit.

The transmit function of the Selective Signaling VI is only active when a valid number has been dialed. The process of transmitting the dialed number is performed in three stages: setup, build signal, and transmit signal, **Figure 11-150**. The setup stage checks the number of digits being sent and automatically transmits a 1 before the other numbers, if the option has been selected. The build signal stage constructs the signal according to the SS1 Standard **[17]**. Each number corresponds to the number of high and low frequency pulses to be transmitted, starting with one for 1 and going up to ten for 0. The first high pulse is 100ms long while each following high pulse is 58ms long with each high frequency pulse being followed by a 42ms low frequency pulse. Each burst of pulses is followed by 225ms of silence and the process repeats for each number dialed. Finally, the constructed signal is calibrated and transmitted via direct communication with the hardware. If the user chooses, the transmitted signal can be played through the computer speakers.

Figure 11-150: Selective Signaling Block Diagram, Transmit Loop.

11.8.4 Selective Signaling Receive.

The receive section of the Selective Signaling VI direct communicates with the hardware and decodes the received selective signaling signals. The Selective Signaling VI must setup, maintain, and close its own communication channel with the hardware because the VI is incapable of running in parallel with any of the other virtual test set functions, **Figure 11-151**. Once the hardware communication channel is setup, the received signal is measured in 200ms segments. Each one of these segments have their frequency and amplitude measured and are displayed in the Frequency and Amplitude graphs respectively. The decoding process is accomplished by counting the number of consecutive segments that qualify as a high frequency pulse, within 100hz of the specified high frequency and greater than -24dBm in power. According to SS1, each high frequency pulse has a range of durations it can be within, 95-160ms for the first pulse of a number and 30-120ms for each following pulse **[17]**. If the high pulse falls within one of these duration ranges the pulse counter is incremented. If no high pulse is detected for more than 160ms the number is considered complete and is added to the numbers array. If the number detected is 1 and 1 is set as a clear digit then the numbers array is cleared. The completed code is displayed when a valid number of digits has been detected or after six seconds without a high pulse.
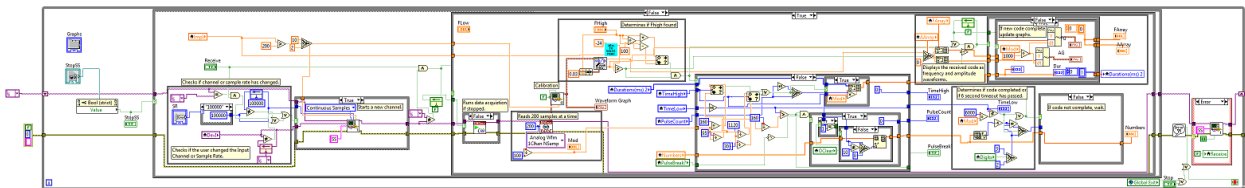
Figure 11-151: Selective Signaling Block Diagram, Receive Loop.

11.9 Frequency Shift Keying Block Diagram

Selective signaling functionally performs transmit and receive independently as shown in, **Figure 11-152**. The block diagram of the Frequency Shift Keying VI can be broken into different sections: event case, queue, transmit, receive, convert to binary, and translate binary sections, **Figure 11-153**.
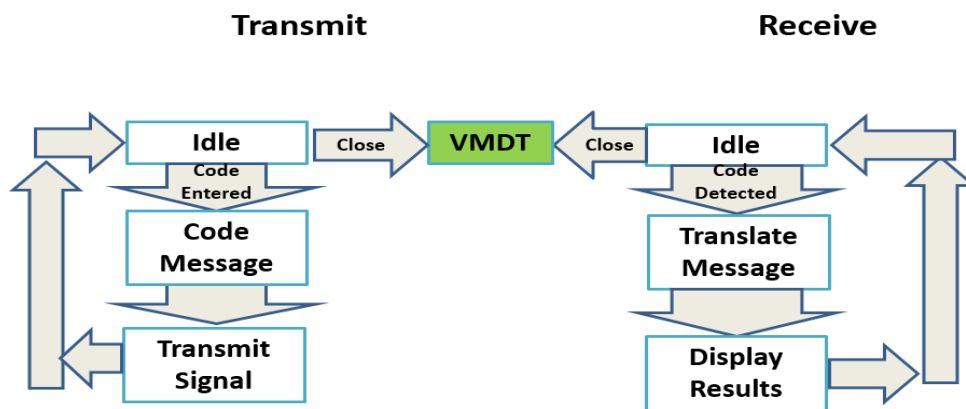
Figure 11-152: Frequency Shift Keying Program Flow
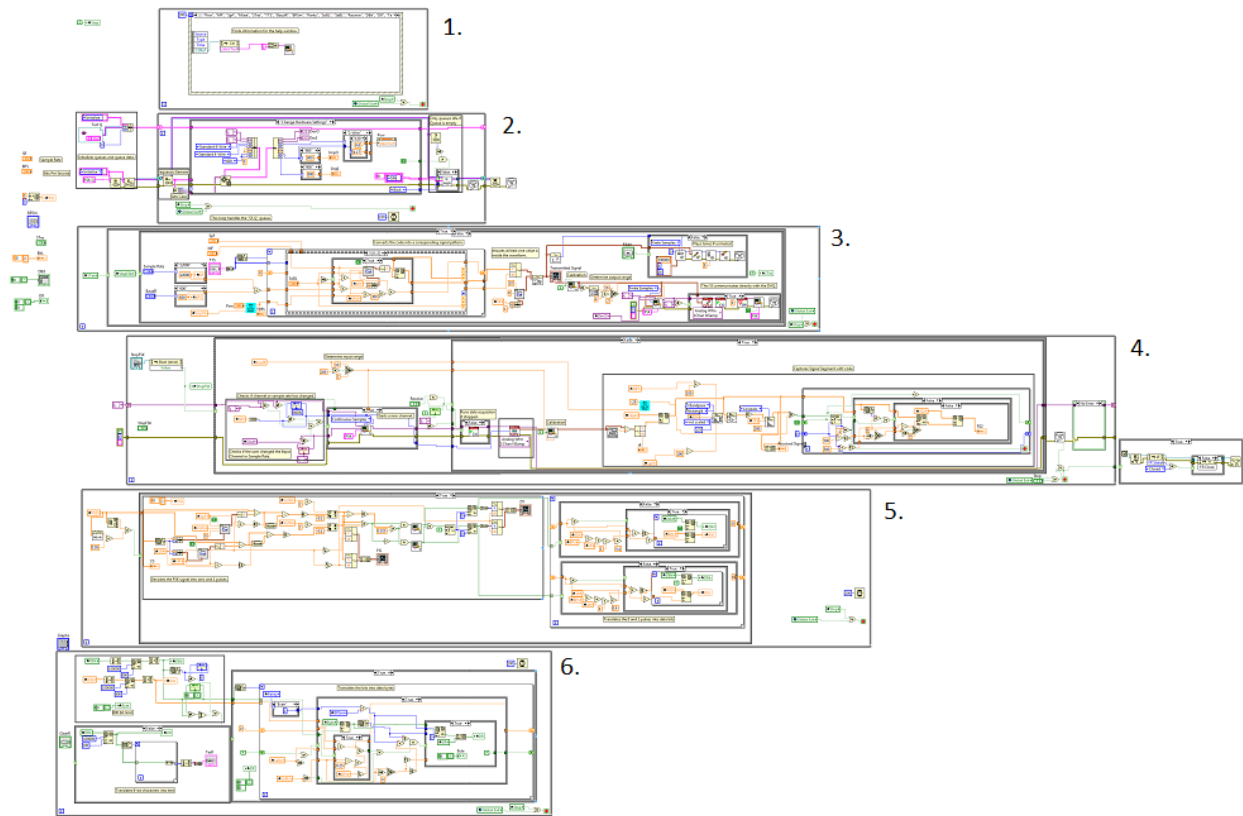
Figure 11-153: Frequency Shift Keying Block Diagram

11.9.1 Frequency Shift Keying Event Case

The Frequency Shift Keying's event case handles the commands from the user. The primary functions of the Frequency Shift Keying's event case is to handle the opening and closing of the help window. The Frequency Shift Keying function has no unique event cases.

11.9.2 Frequency Shift Keying Queue.

The Frequency Shift Keying's Queue Loop handles all of the commands passed to the VI and passes commands to other VI's. The primary function of the Frequency Shift Keying's Queue Loop is to turn on and off the receive function of the virtual test set. The Frequency Shift Keying's Queue Loop does not process any unique commands.

11.9.3 Frequency Shift Keying Transmit.

The transmit function of the Frequency Shift Keying VI is only active while a text message entered by the user is being transmitted. The process of transmitting the text message is performed in two stages: build signal, and transmit signal, **Figure 11-154**. The build signal stage constructs the signal according to Bell 202 standard **[5]**. Each letter of the text message is converted into an 8-bit binary array and converted to a signal consisting of a start bit, the character in binary, a parity bit, and a stop bit with the duration, frequency, and power of each bit specified by user settings. The entire message is created as a single burst signal with continuous phase between bits. Finally, the constructed signal is calibrated

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and transmitted via direct communication with the hardware. If the user chooses, the transmitted signal can be played through the computer speakers.
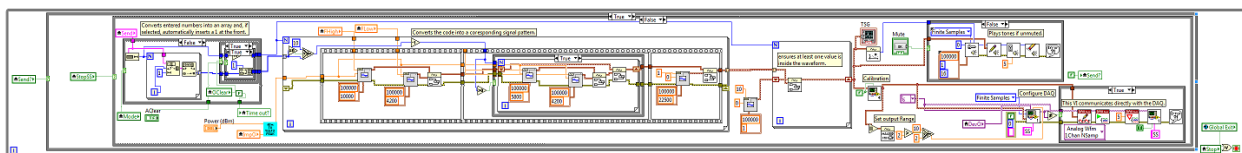

Figure 11-154: Frequency Shift Keying Block Diagram, Transmit Loop.

11.9.4 Frequency Shift Keying Receive.

The receive section of the Frequency Shift Keying VI direct communicates with the hardware and decodes the received selective signaling signals. The Frequency Shift Keying VI must setup, maintain, and close its own communication channel with the hardware because the VI is incapable of running in parallel with any of the other virtual test set functions, **Figure 11-155**. Once the hardware communication channel is setup, the section of the received signal containing the FSK signal is isolated using a band pass filter followed by taking the absolute value of the signal and passing it through a low pass filter. The isolated signal is then passed to the 'Convert to Binary' loop to be processed.

Figure 11-155: Frequency Shift Keying Block Diagram, Receive Loop.

11.9.5 Convert to Binary.

The purpose of the convert to binary loop is to translate the analog signal captured by the receive loop, **Figure 11-156**. When this loop is passed a new signal is compares every point of the signal with the signal one period of the give frequency ago, this is done for both the mark and space frequencies. The resulting signals will be near zero when the signal is at the given frequency. If the resulting signal is for the space comparison is greater than the mark comparison and the received signal amplitude is greater than a built in threshold, 0.025V, it is presumed to be a mark pulse. If the same conditions are met but the mark comparison signal is greater it is presumed to be a space pulse. A binary mark and a binary space signal are created as a result and both are passed through a 'Gap Fill' sub-VI, **Figure 11-157**, to fill small gaps in the signal caused by noise. Finally the mark and space signal can be translated into the individual 1s and 0s of the received Frequency Shift Keying signal. This is done by timing the duration of each mark and space pulse. For any pulse greater than 40% of a bit length, a number of bits equal to the rounded pulse length divided by the bit length is added to the binary translation. The binary translation along with the corresponding bit lengths are passed to the 'Translate Binary' loop.

Figure 11-156: Frequency Shift Keying Block Diagram, Convert Binary Loop.

Figure 11-157: Gap Fill Sub-VI Block Diagram.

11.9.6 Translate Binary.

This final loop translates the full array of 0s and 1s into the individual bytes, **Figure 11-158**. The Frequency Shift Keying VI can only process 800 bits at a time so any excess bits are dropped. The bits are sorted into bytes by looking for a start bit matching the specified start bit duration, counting out the specified bits per symbol plus the parity bit, and waiting for a stop bit of the specified length and stored in the Data Received array. Finally, the bytes are translated into 8-bit characters to be displayed as a text message.
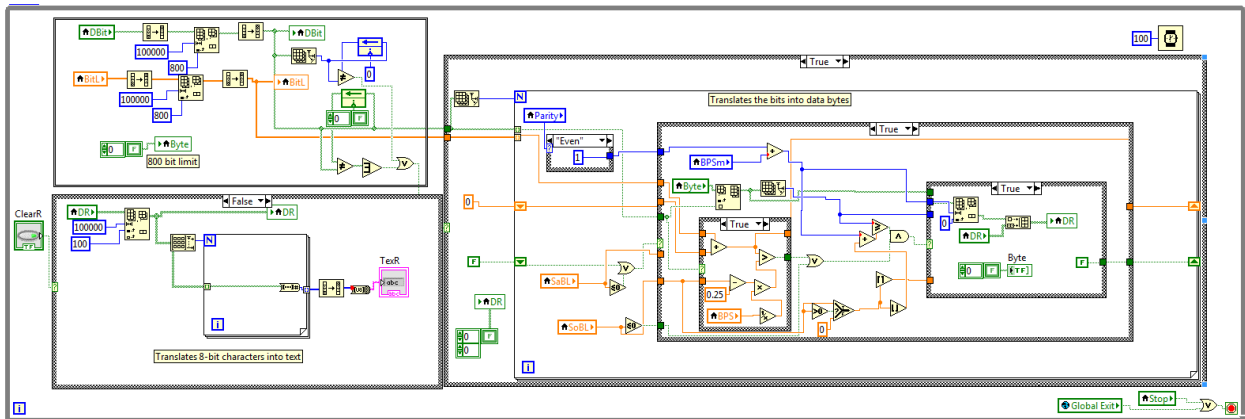
Figure 11-158: Frequency Shift Keying Block Diagram, Translate Binary Loop.

11.10 Digital Multi-Meter Block Diagram

The block diagram of the DMM VI can be broken into the event case and measurement sections, **Figure 11-159**.
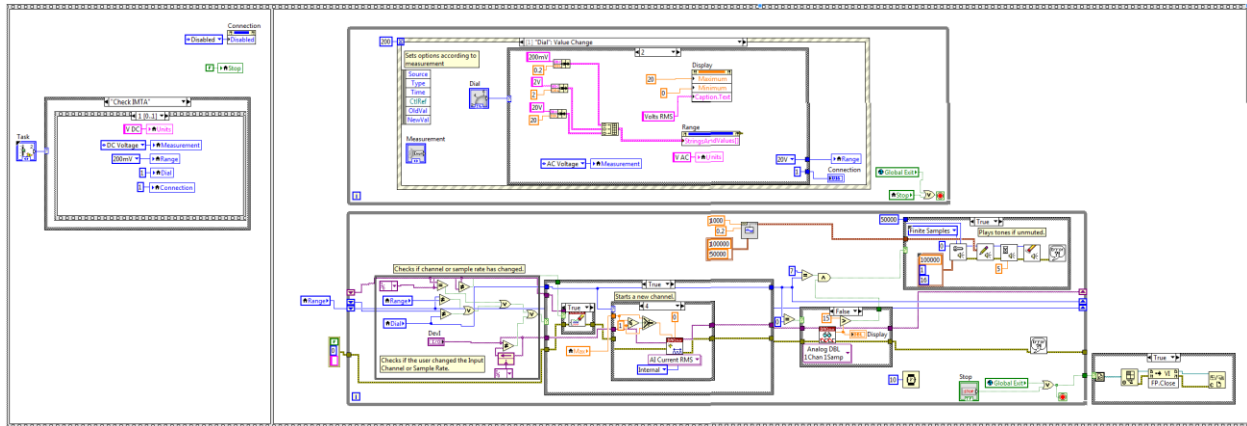
Figure 11-159: Digital Multi-Meter Block Diagram

11.10.1 Digital Multi-Meter Event Case

The Digital Multi-Meter's event case handles the commands from the user. The primary function of the Digital Multi-Meter's event case is to change the measurement settings based on the user's input and change the information displayed in the help VI. The Digital Multi-Meter function has the following unique event cases:

1. Change Measurement.
2. Change Range.

Change Measurement: This event case is activated whenever the user selects a new measurement using the dial. This event sets the measurement that will be made, changes the display units, and changes the options in the range menu, **Figure 11-160**. This event also changes the display indicating the physical connection needed for the measurement.
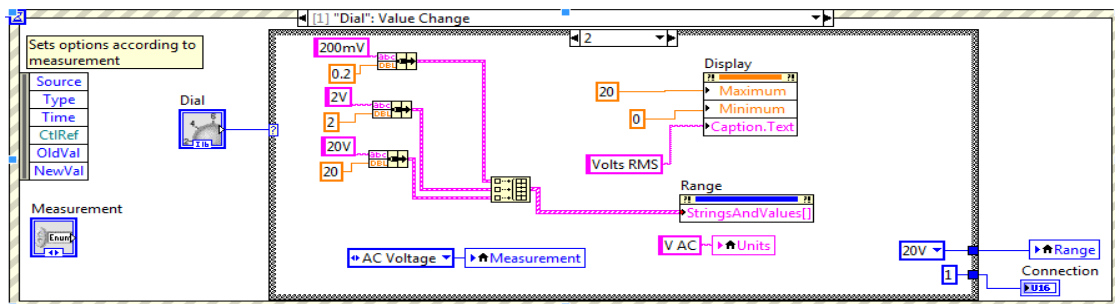
Figure 11-160: Digital Multi-Meter Event Case, Change Measurement Event.

Change Range: This event case is activated whenever the user selects a new measurement range from the drop down menu. This event changes the value range accepted by the hardware as well as the maximum value capable of being displayed, **Figure 11-161**.
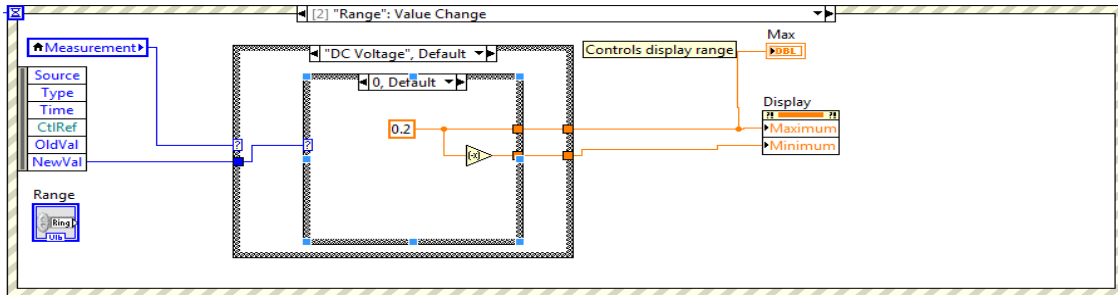
Figure 11-161: Digital Multi-Meter Event Case, Change Range Event.

11.10.2   DMM Measurements.

The measurement section of the DMM VI direct communicates with the DMM circuit built into the MyDAQ. The DMM VI must setup, maintain, and close its own communication channel with the hardware, **Figure 11-162**. While the DMM inputs are independent of the analog input and output pins and can be used in parallel with other measurements, no other VIs use the DMM. Once the hardware communication channel is setup, the DMM VI is capable of measuring various voltage, current, and resistance measurements, **Table 11-7**. The hardware communication is reset whenever the measurement or measurement range is changed and an audible beep is produced when a continuity measurement is made.

Table 11-7: DMM Measurement Descriptions

Measurement	Description
DC Voltage	The DC voltage difference between the DMM probes.
AC Voltage	The RMS AC voltage difference between the DMM probes.
DC Current	The DC current passing through the DMM probes.
AC Current	The RMS AC current passing through the DMM probes.
Resistance	The resistance between the DMM probes.
Diode Voltage	The voltage blocked by a diode.
Continuity	Beeps when the resistance between the DMM probes is less than 15 Ohms.
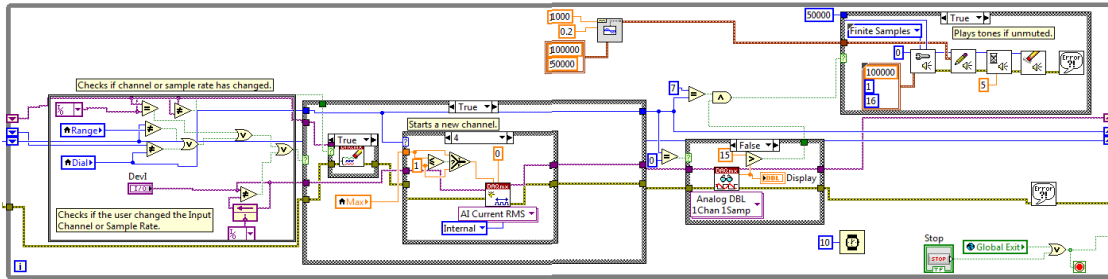
Figure 11-162: DMM Block Diagram, Receive Measurement Loop

11.11 Calibration Block Diagram

The block diagram of the Calibration VI can be broken into different sections: event case, queue, and calibration sections, **Figure 11-163**.
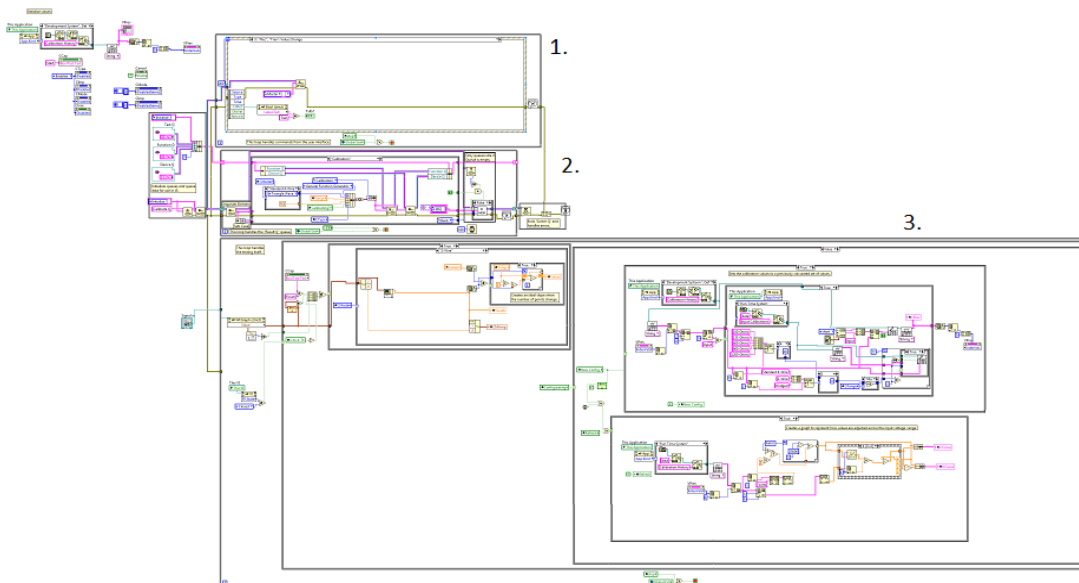
Figure 11-163: Calibration Block Diagram

11.11.1   Calibration Event Case

The Calibration VI's event case handles the commands from the user. The primary function of the Calibration VI's event case is to manage the calibration process and allow the user to view previous calibration values. The Calibration VI has the following unique event cases:

1. Start Calibration.
2. Change Calibration Range or Impedance.
3. Set Input or Output.
4. Set 2 or 4 Wire Mode.
5. Switch Between Calibration Results.
6. Rollback to Previous Calibration.

Start Calibration: This event is queued whenever the 'Start Capture' button is pressed and both starts and finishes the calibration process, **Figure 11-164**. When a calibration is started the other controls on the front panel are disabled, instructions for the calibration are displayed, and the calibration command is queued to begin the calibration process. Once the calibration process has begun the start button becomes a finish button. When the finish button is pressed the other controls are re-enabled and the calibration command is queued again to finish the calibration process.

Figure 11-164: Calibration Event Case, Start Calibration Event

Change Calibration Range or Impedance: This event is activated whenever the user changes the range or impedance setting for the calibration. This event updates the maximum voltage that will be used in the calibration process and displays the result from the last calibration performed with the chosen settings, **Figure 11-165**.

Figure 11-165: Calibration Event Case, Change Calibration Range or Impedance Event

Set Input or Output: This event is activated whenever the user changes the type of calibration to be performed. This event changes the options available in the other settings, toggles the appropriate calibration VIs on or off, updates the maximum voltage that will be used in the calibration process, and displays the result from the last calibration performed with the chosen settings, **Figure 11-166**.

Figure 11-166: Calibration Event Case, Set Input or Output Event

Set 2 or 4 Wire Mode: This event is activated whenever the user changes the hardware mode of the calibration to be performed. This event changes the options available in the other settings, updates the maximum voltage that will be used in the calibration process, and displays the result from the last calibration performed with the chosen settings, **Figure 11-167**.
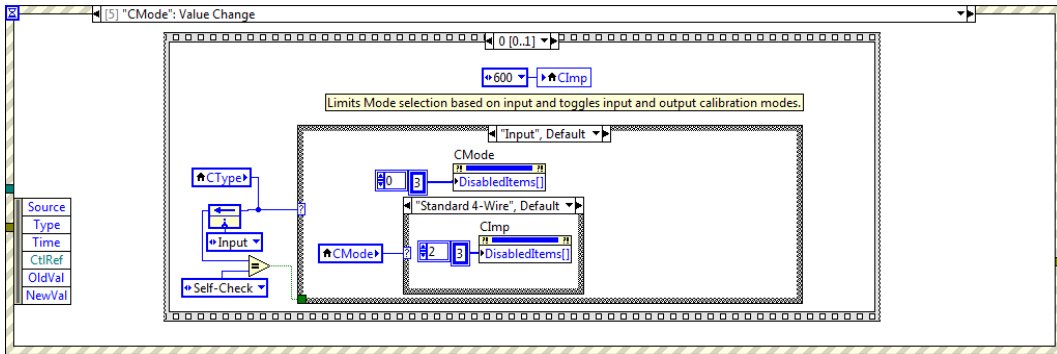
Figure 11-167: Calibration Event Case, Set 2 or 4 Wire Mode Event

Switch Between Calibration Results: This event is activated when the user presses the next or previous buttons. This event allows the user to view calibrations that were previously performed, allowing the user to roll back the device to previous calibration settings if necessary, **Figure 11-168**.
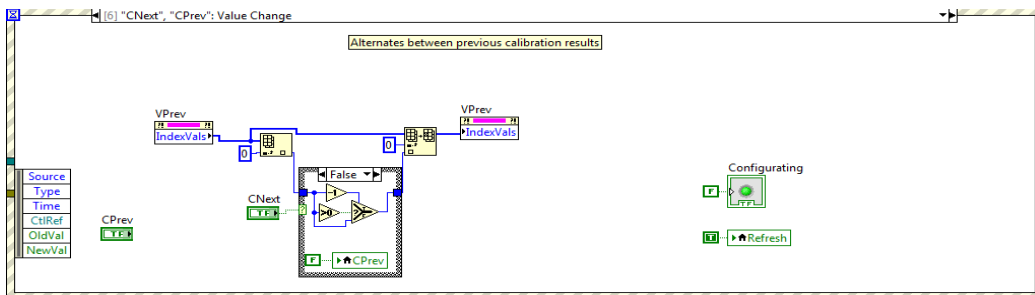
Figure 11-168: Calibration Event Case, Switch Between Calibration Results Event

Rollback to Previous Calibration: This event is activated when the user selects the rollback button. This event starts the process to replace the calibration values for the chosen settings with the values of the calibration currently being viewed, **Figure 11-169**.
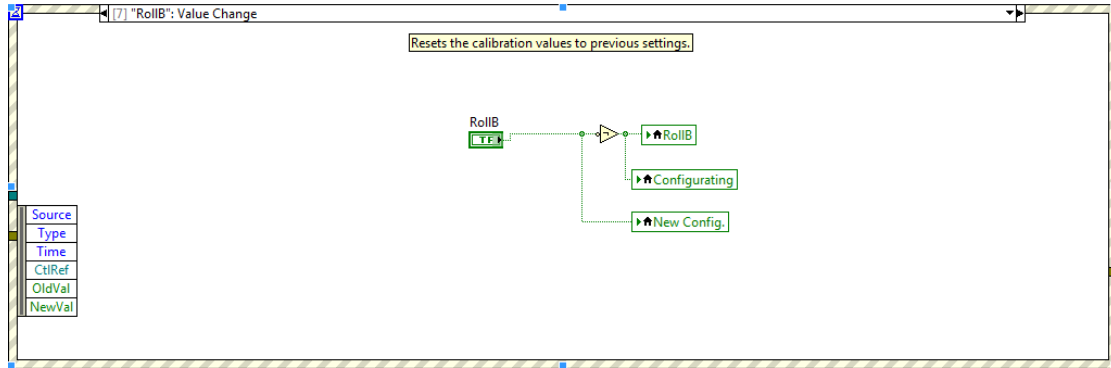
Figure 11-169: Calibration Event Case, Rollback to Previous Calibration Event

11.11.2 Calibration Queue.

The Calibration VI's Queue Loop handles all of the commands passed to the VI and passes commands to other VI's. The primary function of the Calibration VI's Queue Loop is to take control of the function generator. The Calibration VI's Queue Loop only contains one unique command case, calibration. This command seizes control of the function generator to produce the signals used in the calibration process, **Figure 11-170**. This command is different from the other signal requesting commands in that it causes the function generator to ignore the normal restrictions placed on the amplitude of the signals generated. This is done to allow the function generator to generate the amplitude of the signal output by the hardware rather than the amplitude that is expected to be transmitted by the virtual test set. This command is only used when the output of the virtual test set is being calibrated and must be toggled on and off. This event allows the Calibration VI to know the exact voltage values that are being transmitted by the hardware.
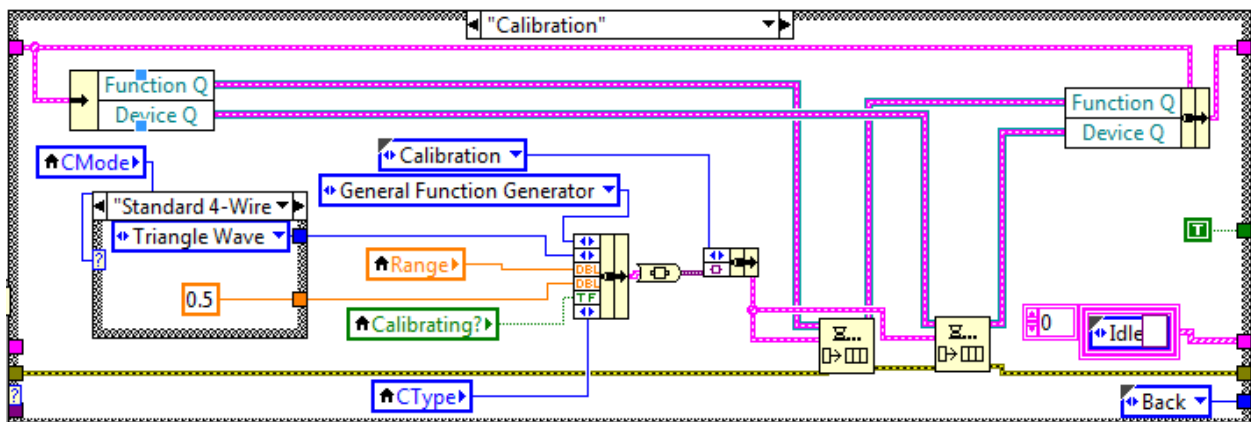
Figure 11-170: Calibration Queue, Calibration Command.

11.11.3 Calibration Function Section.

The calibration function section is the part of the block diagram that performs the calibration itself as well as implements the new calibration constants. This section can be further broken into four sub-sections. The first section uses the received signal to generate the calibration curve, **Figure 11-171**. The calibration VI uses triangle and sine waves in its calibration processes. When using a triangle wave the full downward slope is captured and used for calibration. When a sine wave is used a full second of the sine wave captured and every point is arranged from the maximum value to the minimum value, for a pure sine wave this always creates a half a cosine wave and a converter VI, **Figure 11-172**, uses an ideal cosine function to generate the downward slope used for calibration. At the same time that the signal is captured, an ideal slope with the same number of data points is generated as a reference using the expected range as the maximum and the negative range as the minimum value.
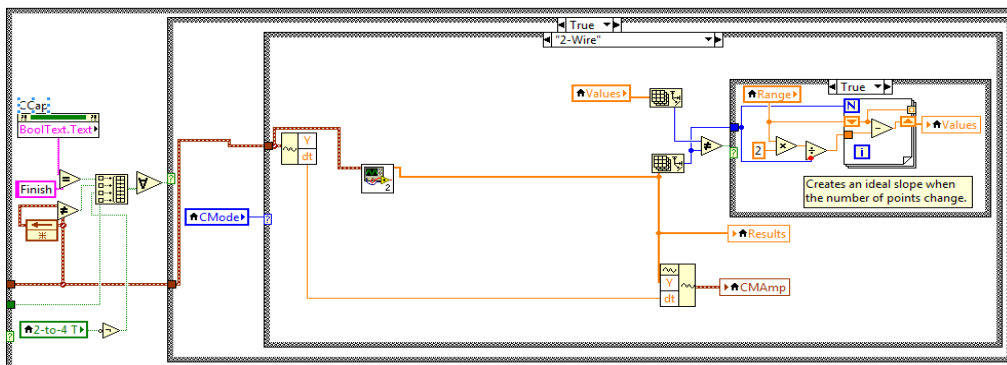
Figure 11-171: Calibration Block Diagram, Signal Capture Section.
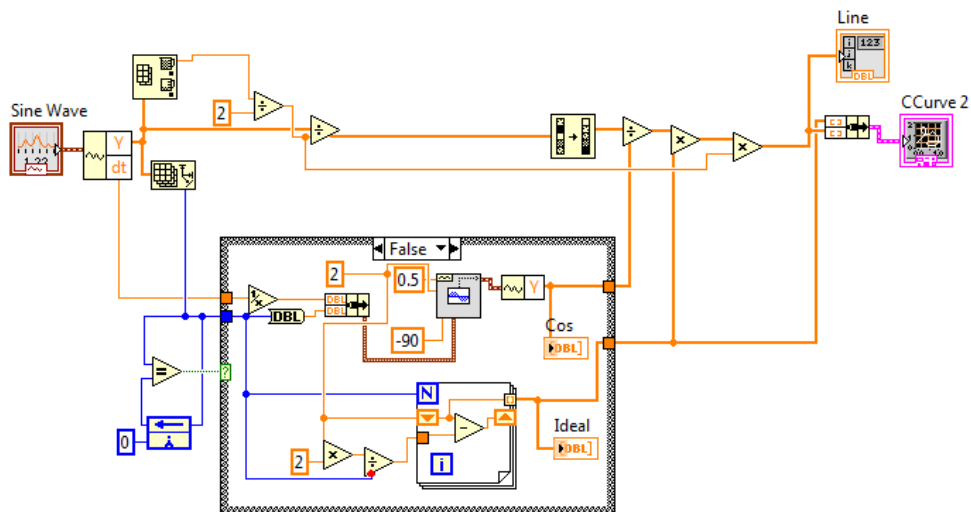
Figure 11-172: Sine to Line Sub-VI Block Diagram.

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The second section generates the calibration constant by finding the difference between the slope generated from the captured signal and the ideal slop, **Figure 11-173**. The best fit polynomial function is calculated and each of the multiples is saved as the calibration constants of the hardware settings used in the calibration. The Calibration VIs use these constants to compensate for any losses or gains caused by the hardware.
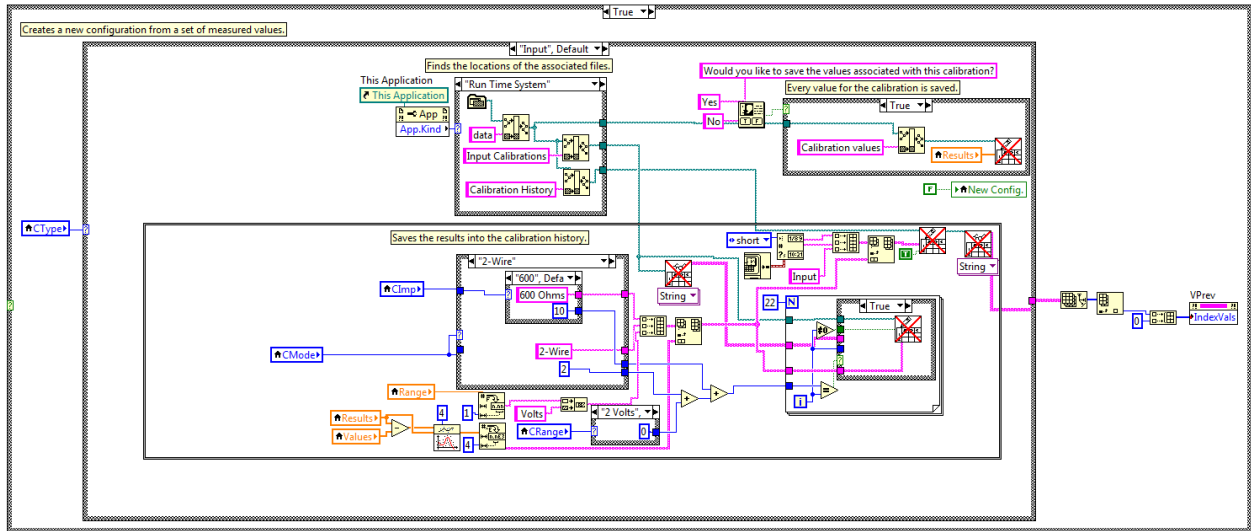
Figure 11-173: Calibration Block Diagram, Save Calibration Section.

The third section displays the calibration curve associated with the current calibration settings, **Figure 11-174**. This section uses the same method to find the calibration constants as the Calibration VIs and generates the calibration curve by using those calibration constants to calibrate an ideal slope. For a transmitted signal the ideal curve represents the amplitude that is to be transmitted and the calibration curve represents the amplitude that must be output by the hardware to transmit the desired amplitude. For a received signal the ideal curve represent the amplitude measured by the hardware and the calibration curve represents the amplitude received from the transmission line.
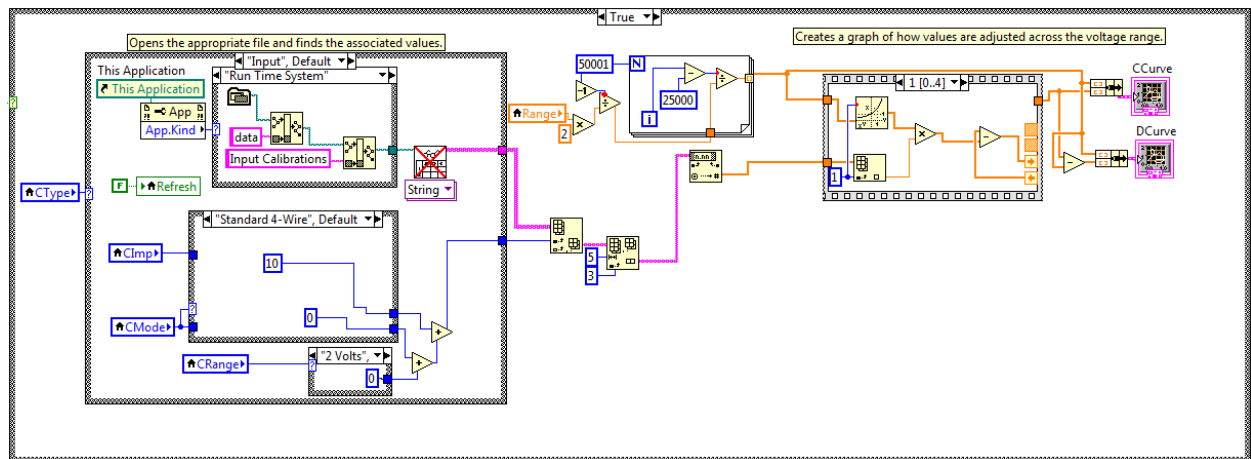
Figure 11-174: Calibration Block Diagram, Generate Calibration Curves Section.

The fourth and final section will replace the current calibration constants with those generated in a previous calibration, **Figure 11-175**. This is done by looking up the calibration constants based on the current calibration settings and the chosen calibration. The new calibration constants then simply replace the corresponding calibration constants used by the Calibration VIs.
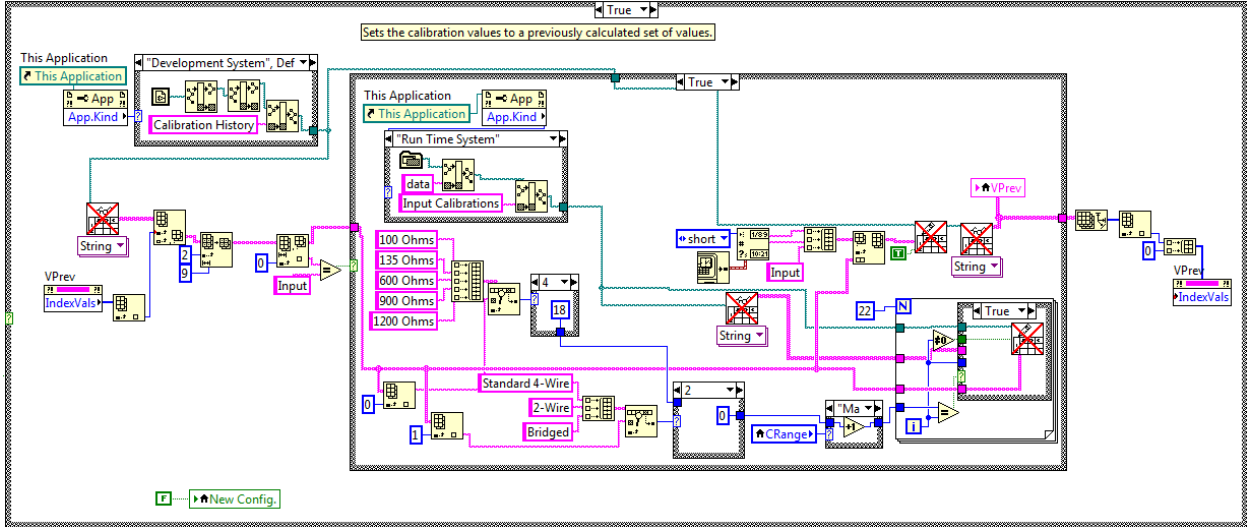
Figure 11-175: Calibration Block Diagram, Return to Previous Calibration Section.

12 Appendix C: Testing Measurements

Table 12-1 Output Level

VIRTUAL TEST SET Output Setting	VIRTUAL TEST SET Measured	Halcyon Measured
10 dBm 600 Ohm, 1004 Hz	10.00 dBm	10.0 dBm
0 dBm 600 Ohm, 1004 Hz	0.02 dBm	0.0 dBm
-10 dBm 600 Ohm, 1004 Hz	-9.99 dBm	-10.0 dBm
-19 dBm 600 Ohm, 1004 Hz	-18.99 dBm	-19.0 dBm
-40 dBm 600 Ohm, 1004 Hz	-39.99 dBm	-40.0 dBm
10 dBm 900 Ohm, 1004 Hz	10.00 dBm	10.0 dBm
0 dBm 900 Ohm, 1004 Hz	0.012 dBm	0.0 dBm
-10 dBm 900 Ohm, 1004 Hz	-9.99 dBm	-10.0 dBm
-19 dBm 900 Ohm, 1004 Hz	-18.99 dBm	-19.0 dBm
-40 dBm 900 Ohm, 1004 Hz	-39.98 dBm	-39.9 dBm

Table 12-2 Single Tone Source Noise Level

VIRTUAL TEST SET Output Setting	VIRTUAL TEST SET Measured	Halcyon Measured
10 dBm 600 Ohm, 1004 Hz	59.92 dB SNR	71.2 dB SNR
-10 dBm 600 Ohm, 1004 Hz	60.00 dB SNR	51.4 dB SNR
-40 dBm 600 Ohm, 1004 Hz	39.99 dB SNR	36.9 dB SNR
10 dBm 900 Ohm, 1004 Hz	59.89 dB SNR	70.5 dB SNR
-10 dBm 900 Ohm, 1004 Hz	60.00 dB SNR	51.2 dB SNR
-40 dBm 900 Ohm, 1004 Hz	40.04 dB SNR	38.6 dB SNR

The significant difference in the readings in **Table 12-2** was caused by a high noise floor that was detected when the virtual test set was connected to the Halcyon. Additional testing lead to the conclusion that the noise is likely caused by grounding differences.

Table 12-3 Single Tone Frequency Output

VIRTUAL TEST SET Output Setting	VIRTUAL TEST SET Measured	Halcyon Measured
-13 dBm, 50 Hz	50.00 Hz	50 Hz
-13 dBm, 200 Hz	200.00 Hz	200 Hz
-13 dBm, 500 Hz	500.00 Hz	500 Hz
-13 dBm, 1000 Hz	1000.00 Hz	1000 Hz
-13 dBm, 5000 Hz	5000.00 Hz	5000 Hz
-13 dBm, 9999 Hz	9999.00 Hz	9999 Hz
-13 dBm, 10000 Hz	10000.00 Hz	10000 Hz
-13 dBm, 15000 Hz	15000.00 Hz	15000 Hz
-13 dBm, 20000 Hz	20000.00 Hz	20001 Hz
-13 dBm, 40000 Hz	40000.00 Hz	40002 Hz

Table 12-4 Single Tone Flatness

VIRTUAL TEST SET Output Setting	VIRTUAL TEST SET Measured	Halcyon Measured
-13 dBm, 200 Hz	-12.99 dBm	-13.1 dBm
-13 dBm, 500 Hz	-12.99 dBm	-13.1 dBm
-13 dBm, 1000 Hz	-12.99 dBm	-13.0 dBm
-13 dBm, 5000 Hz	-13.04 dBm	-13.1 dBm
-13 dBm, 9999 Hz	-13.19 dBm	-13.0 dBm
-13 dBm, 10000 Hz	-13.19 dBm	-13.0 dBm
-13 dBm, 15000 Hz	-13.42 dBm	-13.2 dBm

Table 12-5 Input Level Measurement

VIRTUAL TEST SET Output Setting	VIRTUAL TEST SET Measured	Halcyon Measured
10 dBm, 20 Hz	10.00 dBm	6.7 dBm
10 dBm, 200 Hz	10.00 dBm	9.9 dBm
10 dBm, 1004 Hz	10.00 dBm	10.0 dBm
10 dBm, 15000 Hz	9.99 dBm	9.8 dBm
10 dBm, 40000 Hz	9.97 dBm	5.2 dBm
0 dBm, 20 Hz	0.00 dBm	-0.9 dBm
0 dBm, 200 Hz	0.00 dBm	-0.1 dBm
0 dBm, 1004 Hz	0.00 dBm	0.0 dBm
0 dBm, 15000 Hz	-0.01 dBm	-0.3 dBm
0 dBm, 40000 Hz	-0.02 dBm	-4.1 dBm
-19 dBm, 20 Hz	-18.99 dBm	-32.2 dBm
-19 dBm, 200 Hz	-18.99 dBm	-19.1 dBm
-19 dBm, 1004 Hz	-18.99 dBm	-19.1 dBm
-19 dBm, 15000 Hz	-19.07 dBm	-19.2 dBm
-19 dBm, 40000 Hz	-19.35 dBm	-23.1 dBm
-50 dBm, 20 Hz	-50.01 dBm	-53.9 dBm
-50 dBm, 200 Hz	-50.02 dBm	-48.9 dBm
-50 dBm, 1004 Hz	-50.02 dBm	-48.7 dBm
-50 dBm, 15000 Hz	-50.10 dBm	-48.8 dBm
-50 dBm, 40000 Hz	-50.39 dBm	-50.7 dBm

Table 12-6 Noise Measurement

Noise Power (1-40k Hz)	Filter	VIRTUAL TEST SET Result	Halcyon Result
13.6 dBm	C-Message	90.2 dBrn +/- 0.2 dB	90.0 dBrn +/- 0.5 dB
13.6 dBm	C-Message	89.9 dBrn +/- 0.2 dB	89.5 dBrn +/- 0.5 dB
13.6 dBm	3.4kHz Flat	92.6 dBrn +/- 0.2 dB	91.2 dBrn +/- 0.3 dB
13.6 dBm	3.4kHz Flat Notched	92.7 dBrn +/- 0.2 dB	90.8 dBrn +/- 0.4 dB
13.6 dBm	D-Filter	92 dBrn +/- 0.2 dB	91.2 dBrn +/- 0.3 dB

13.6 dBm	D-Notched	92 dBrn +/- 0.2 dB	90.8 dBrn +/- 0.4 dB
13.6 dBm	15kHz Flat	98.7 dBrn +/- 0.2 dB	98.3 dBrn +/- 0.3 dB
-26.3 dBm	C-Message	50.8 dBrn +/- 0.2 dB	50.6 dBrn +/- 0.5 dB
-26.3 dBm	C-Notched	50.7 dBrn +/- 0.2 dB	50 dBrn +/- 0.5 dB
-26.3 dBm	3.4kHz Flat	53.4 dBrn +/- 1 dB	50.2 dBrn +/- 0.3 dB
-26.3 dBm	3.4kHz Flat Notched	53.4 dBrn +/- 0.2 dB	50.2 dBrn +/- 0.3 dB
-26.3 dBm	D-Filter	52.9 dBrn +/- 0.2 dB	50.2 dBrn +/- 0.3 dB
-26.3 dBm	D-Notched	52.7 dBrn +/- 0.2 dB	49.7 dBrn +/- 0.4 dB
-26.3 dBm	15kHz Flat	59.5 dBrn +/- 0.2 dB	59.1 dBrn +/- 0.2 dB
-64.8 dBm	C-Message	13.4 dBrn +/- 0.4 dB	14.4 dBrn +/- 0.4 dB
-64.8 dBm	C-Notched	13.4 dBrn +/- 0.4 dB	13.6 dBrn +/- 0.5 dB
-64.8 dBm	3.4kHz Flat	15.3 dBrn +/- 0.2 dB	16.4 dBrn +/- 0.4 dB
-64.8 dBm	3.4kHz Flat Notched	16.1 dBrn +/- 0.1 dB	15.7 dBrn +/- 0.3 dB
-64.8 dBm	D-Filter	15.3 dBrn +/- 0.2 dB	16.4 dBrn +/- 0.4 dB
-64.8 dBm	D-Notched	15.3 dBrn +/- 0.2 dB	15.7 dBrn +/- 0.3 dB
-64.8 dBm	15kHz Flat	21.9 dBrn +/- 0.1 dB	Unable to Measure

Table 12-7 Signal to Noise Ratio Measurements

VIRTUAL TEST SET Output Setting	VIRTUAL TEST SET Measured	Halcyon Measured
10 dBm 1004 Hz Plus 9.1 dBm Noise (1-40k Hz)	14.1 +/- 0.3 dB C-Filtered	14.8 +/- 0.5 dB C-Filtered
10 dBm 1004 Hz Plus -7.0 dBm Noise (1-40k Hz)	29.8 +/- 0.3 dB C-Filtered	30.3 +/- 0.3 dB C-Filtered
10 dBm 1004 Hz Plus -21.9 dBm Noise (1-40k Hz)	45.5 +/- 0.4 dB C-Filtered	45.5 +/- 0.4 dB C-Filtered
10 dBm 1004 Hz Plus -27.0 dBm Noise (1-40k Hz)	50.5 +/- 0.4 dB C-Filtered	50.4 +/- 0.4 dB C-Filtered

Table 12-8 Phase Jitter Measurement

VIRTUAL TEST SET Output Setting	VIRTUAL TEST SET Measured	Halcyon Measured
10 dBm 1 kHz Plus -2 dBm 1.1 kHz	29.2º Pk-Pk	22.4º Pk-Pk
10 dBm 1 kHz Plus -10 dBm 1.1 kHz	11.5º Pk-Pk	8.2º Pk-Pk
10 dBm 1 kHz Plus -40 dBm 1.1 kHz	0.4º Pk-Pk	0.0º Pk-Pk
-10 dBm 1 kHz Plus -20 dBm 1.1 kHz	37.1º Pk-Pk	26.5º Pk-Pk
-10 dBm 1 kHz Plus -30 dBm 1.1 kHz	11.5º Pk-Pk	8.2º Pk-Pk
-10 dBm 1 kHz Plus -40 dBm 1.1 kHz	3.6º Pk-Pk	4.1º Pk-Pk
-40 dBm 1 kHz	37.4º Pk-Pk	35.5º Pk-Pk

Plus -50 dBm 1.1 kHz		
-40 dBm 1 kHz Plus -60 dBm 1.1 kHz	11.8º Pk-Pk	12.3º Pk-Pk
-40 dBm 1 kHz Plus -70 dBm 1.1 kHz	4.1º Pk-Pk	8.0º Pk-Pk

Table 12-9 Amplitude Jitter Measurement

VIRTUAL TEST SET Output Setting	VIRTUAL TEST SET Measured
10 dBm 1 kHz Plus -2 dBm 1.1 kHz	26.0% Peak
10 dBm 1 kHz Plus -10 dBm 1.1 kHz	10.2% Peak
10 dBm 1 kHz Plus -40 dBm 1.1 kHz	0.3% Peak
-10 dBm 1 kHz Plus -20 dBm 1.1 kHz	32.8% Peak
-10 dBm 1 kHz Plus -30 dBm 1.1 kHz	10.2% Peak
-10 dBm 1 kHz Plus -40 dBm 1.1 kHz	3.2% Peak
-40 dBm 1 kHz Plus -50 dBm 1.1 kHz	33.0% Peak
-40 dBm 1 kHz Plus -60 dBm 1.1 kHz	10.6% Peak
-40 dBm 1 kHz Plus -70 dBm 1.1 kHz	3.6% Peak

Table 12-10 Impulse Noise Measurement

VIRTUAL TEST SET Output Setting	VIRTUAL TEST SET Measured	HST-3000
No Tone, 0.01 V Pulse	Counts at 43 dBrn No Filter	Counts at 40 dBrn No Filter
No Tone, 1 V Pulse	Counts at 83 dBrn No Filter	Counts at 80 dBrn No Filter
No Tone, 4 V Pulse	Counts at 95 dBrn No Filter	Counts at 90+ dBrn (Max) No Filter
1800Hz Tone 0.01 V Amplitude	Counts at 50 dBrn No Filter	Counts at 47 dBrn No Filter
1800Hz Tone 1 V Amplitude	Counts at 90 dBrn No Filter	Counts at 87 dBrn No Filter
1800Hz Tone 4 V Amplitude	Counts at 102 dBrn No Filter	NA No Filter

Table 12-11 Phase Hit Measurement

Threshold	VIRTUAL TEST SET Output required to measure on VIRTUAL TEST SET	VIRTUAL TEST SET Output required to measure on Halcyon
5º Phase Hit	-13 dBm 1004Hz Tone 4.0º 5ms Phase Hit	-13 dBm 1004Hz Tone 6.0º 5ms Phase Hit
25º Phase Hit	-13 dBm 1004Hz Tone 23.5º 5ms Phase Hit	-13 dBm 1004Hz Tone 23.5º 5ms Phase Hit
45º Phase Hit	-13 dBm 1004Hz Tone 39º 5ms Phase Hit	-13 dBm 1004Hz Tone 40º 5ms Phase Hit

Table 12-12 Gain Hit Measurement

Threshold	VIRTUAL TEST SET Output required to measure on VIRTUAL TEST SET	VIRTUAL TEST SET Output required to measure on Halcyon
2 dB Gain Hit	-13 dBm 1004Hz Tone 1.6 dB Gain Hit	-13 dBm 1004Hz Tone 2.1 dB Gain Hit
4 dB Gain Hit	-13 dBm 1004Hz Tone 3.9 dB Gain Hit	-13 dBm 1004Hz Tone 4.1 dB Gain Hit
10 dB Gain Hit	-13 dBm 1004Hz Tone 10 dB Gain Hit	-13 dBm 1004Hz Tone 10.1 dB Gain Hit

Table 12-13 23-Tone Level Measurement

VIRTUAL TEST SET Output Setting	VIRTUAL TEST SET Measured	Halcyon Measured
23-Tone -6 dBm	-6.00 dBm	-6.0 dBm
23-Tone -23 dBm	-23.00 dBm	-23.0 dBm
23-Tone -40 dBm	-39.99 dBm	-40.0 dBm
23-Tone -13 dBm With 1.2kHz 0.004 V (45 dB 3rd IMD)	-13.00 dBm	-13.0 dBm
23-Tone -13 dBm With 1.2kHz 0.009 V (40 dB 3rd IMD)	-13.00 dBm	-13.0 dBm
23-Tone -13 dBm With 1.2kHz 0.015 V (35 dB 3rd IMD)	-12.99 dBm	-13.0 dBm
23-Tone -13 dBm With 1.2kHz 0.026 V (30 dB 3rd IMD)	-13.00 dBm	-13.0 dBm

Table 12-14 Envelope Delay Distortion Measurement

VIRTUAL TEST SET Output Setting	VIRTUAL TEST SET Measured	Halcyon Measured

23-Tone -13 dBm	0 us	4 us
23-Tone -13 dBm 1921.87Hz phase +3	107 us	108 us
23-Tone -13 dBm 1921.87Hz phase +30	1067 us	1067 us
23-Tone -13 dBm With 1.2kHz 0.026 V (30 dB 3rd IMD)	0 us	4 us
23-Tone -13 dBm 1921.87Hz phase +3 With 1.2kHz 0.026 V (30 dB 3rd IMD)	106 us	106 us
23-Tone -13 dBm 1921.87Hz phase +30 With 1.2kHz 0.026 V (30 dB 3rd IMD)	1067 us	1067 us

Table 12-15 Intermodulation Distortion Measurement

VIRTUAL TEST SET Output Setting	VIRTUAL TEST SET Measured	Halcyon Measured
23-Tone -13 dBm With 1.2kHz 0.001 V (~60 dB 3rd IMD)	59.7 +/- 2.5 dB	60+ dB
23-Tone -13 dBm With 1.2kHz 0.0025 V (~50 dB 3rd IMD)	51.8 +/- 2.5 dB	56.3 +/- 2.5 dB
23-Tone -13 dBm With 1.2kHz 0.015 V (~35 dB 3rd IMD)	36.2 +/- 3 dB	36.2 +/- 3 dB
23-Tone -13 dBm With 1.2kHz 0.07 V (~20 dB 3rd IMD)	20.4 +/- 2.5 dB	39.1 +/- 1 dB
23-Tone -13 dBm With 1185Hz 0.001 V (~60 dB 2nd IMD)	59.7 +/- 2.5 dB	60+ dB
23-Tone -13 dBm With 1185Hz 0.0025 V (~50 dB 2nd IMD)	51.8 +/- 2.5 dB	56.3 +/- 2.5 dB
23-Tone -13 dBm With 1185Hz 0.015 V (~35 dB 2nd IMD)	36.2 +/- 2.5 dB	39.1 +/- 1 dB
23-Tone -13 dBm With 1185Hz 0.07 V (~20 dB 2nd IMD)	20.4 +/- 2.5 dB	25.4 +/- 1 dB