

# USER'S MANUAL



WINDOWS SOFTWARE FOR  
SYSTEM ONE

# APWIN System One User's Manual



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# Capabilities

System One is a comprehensive high-performance audio testing system. The basic analog System One (SYS-22) tests the amplitude, noise, phase, frequency, and total harmonic distortion plus noise characteristics of analog input/output audio devices. Options may be added for intermodulation distortion (IMD), wow and flutter (W&F), and generation of sinewave bursts, squarewaves, and white and pink noise (BUR). System One + DSP (SYS-222) adds a DSP module for enhanced testing of analog signals including FFT spectrum analysis, digital storage oscilloscope operation, multitone testing, individual harmonic analysis and other selective measurements, and

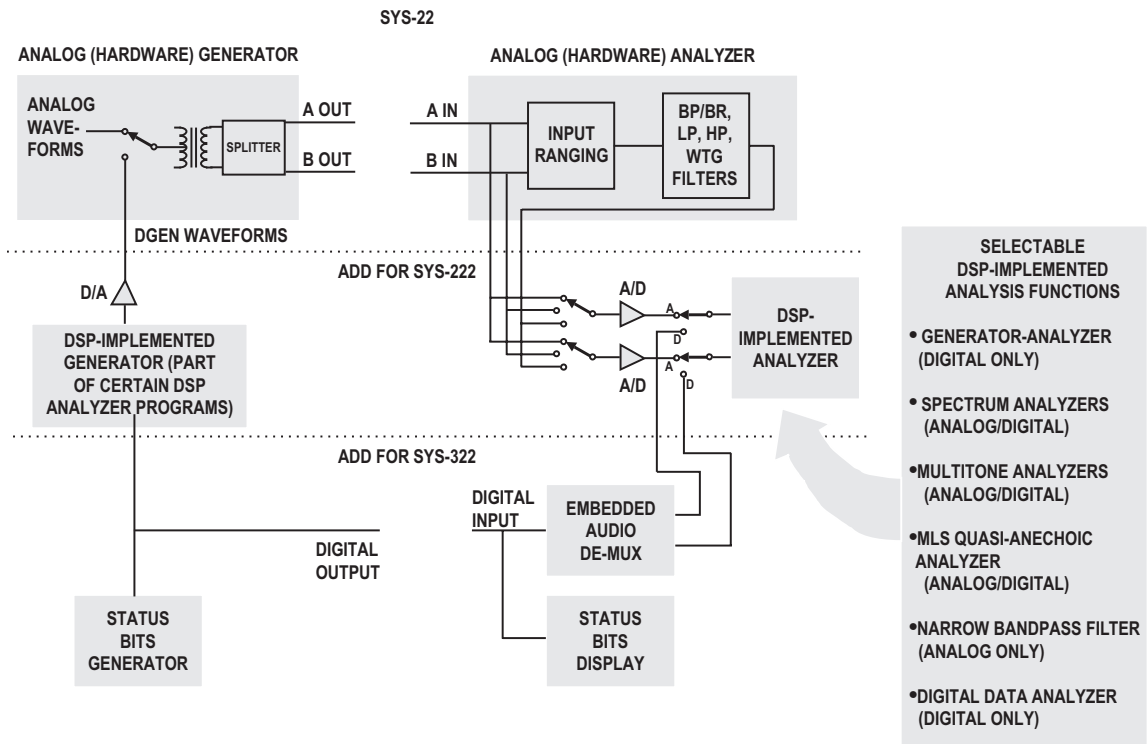


Figure 1-1 Conceptual Block Diagram, System One

quasi-anechoic testing by the MLS technique. System One Dual Domain (SYS-322) adds digital input and output capability in the AES/EBU, consumer (SPDIF), and serial and parallel formats for driving and measuring digital inputs and outputs.

System One has no controls or displays on the instrument chassis itself. Instead, System One connects to a Windows-compatible personal computer for user interface functions including control, display, sweeps and graphing, results storage, automatic test procedures with Pass/Fail limits, and interface to printers and plotters. The interconnection between System One and the computer is via the APIB (Audio Precision Interface Bus) and an ISA-WIN or PCM-WIN interface card for standard ISA slots or PCMCIA slots, respectively. All signal generation and analysis is done in the System One chassis, including storage in the DSP module of signals acquired for waveform display or FFT analysis.

## APWIN Software

APWIN software operating under Windows 3.11 or Windows 95 supplies all user interface functions. Panels (small Windows) on the APWIN screens are provided for each instrument function such as analog generator, DSP analyzer, accessory signal switchers, and for major functions such as sweep setup, sweep settling control, and data display by X-Y graph or bargraph. APWIN also provides flexible units conversion, translating the basic measured values from the System One chassis into a wide range of units such as dBu, dBV, octaves, decades, and dB below digital full scale. APWIN is graphically-oriented, normally plotting data in real time as a sweep progresses. Instrument setups and acquired data may be saved to disk as test files. Multiple test files may be linked into complete test procedures. User-created limits may be applied to test results for automatic Pass/Fail decisions by the computer.

APWIN is operated by a combination of techniques that will be familiar to those who have used other Windows applications. APWIN is most-easily operated with a mouse, but keyboard alternatives are available for almost every function. There are typically several alternatives for the most common operations, including toolbar icons

for the mouse and keyboard shortcuts. Toolbars permit rapid display of all instrument control and display panels. A menu provides access to all major features. The menu is operable by the mouse or from the keyboard.

---

## APWIN Tutorial

This User's Manual is written primarily as a reference manual containing full details of the operation of System Two via APWIN software. Thus, it is not organized optimally for initially learning APWIN operation. We strongly urged that first-time or relatively-inexperienced users of APWIN go through the System One Tutorial booklet. That Tutorial is specifically designed for teaching the usage of System One and APWIN, starting with basic concepts and building up to more sophisticated techniques. The Tutorial also includes dozens of exercises, most of which use example tests set up expressly for learning purposes.

---

## Standard Test Files

A large number of sample tests are supplied with APWIN and are automatically installed. These tests are intended to provide a quick start into testing typical audio devices. The tests are located under the C:\APWIN\S1 directory. They are further organized into four subdirectories. These directories are "A-a", for testing analog in-analog out devices, "A-d" for analog input-digital output equipment, "D-a" for digital in-analog out devices, and "D-d" for digital i/o devices. Sample procedures are also furnished, including some which perform a quick performance check and more complete diagnostics of the System One hardware. These test files are described in the final chapter of this manual.

---

## Connections to Device Under Test

---

### Upper-Left Connector Panel

---

The **upper-left** connector panel contains all of System One's analog generator output connectors, with Channel A at the left and Channel B at the right. APWIN software panels, described in the

Analog Generator chapter, permit driving A only, B only, or both simultaneously. The two channel outputs may be selected in phase or one inverted with respect to the other. The double-banana connectors are hard-wired in parallel with the XLRs. A banana jack is connected to the output transformer center-tap.

## Upper-Right Connector Panel

The **upper-right** connector panel contains all of System One's analog analyzer input connectors, with Channel A at the left and Channel B at the right. The

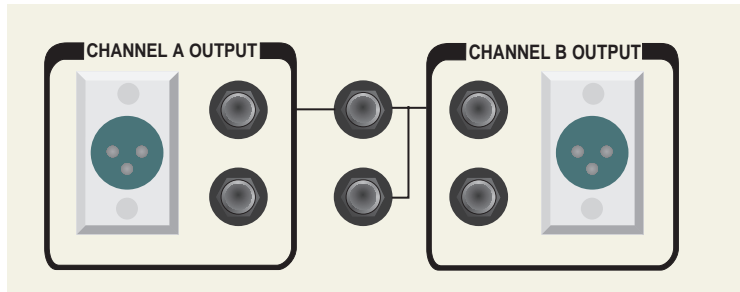


Figure 1-2 Analog Generator (Upper Left) Output Connector Panel

Analog Analyzer software panel permits selection on Channel A of either the BNC AUX INPUT or the parallel-wired XLR and banana jacks. The Analog Analyzer software panel also selects which input connector drives which meters.

## Lower-Left Connector Panel

A **lower-left** connector panel is provided with Dual Domain (SYS-322) units. The Toslink optical input and output connectors are at the left, RCA phono input and output connectors near the center, and XLR balanced connectors at the right. The XLR input may be

terminated in 110 Ohms or operating in bridging (high impedance) mode as

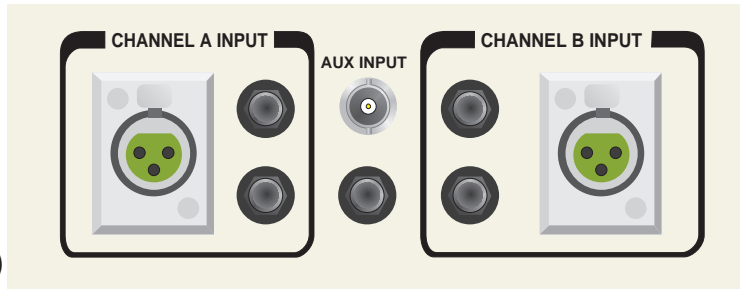


Figure 1-3 Analog Analyzer (Upper Right) Input Connector Panel

selected by the push-push switch. All three digital output connectors are driven

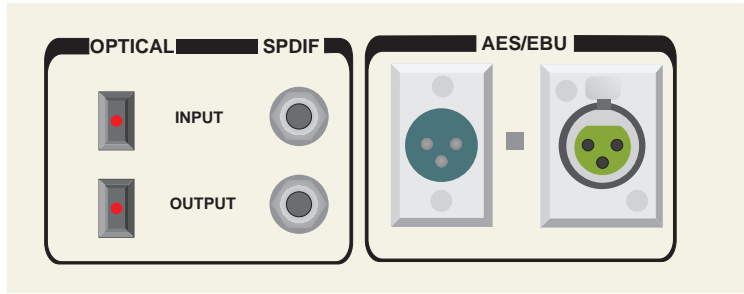


Figure 1-4 Digital Input/Output (Lower Left) Connector Panel

if Serial is selected in the Output section of the DIO (Digital Input Output) software panel. Automatic circuitry senses and selects the digital input connector that has signal. General-purpose serial and parallel input and output connectors are located on the rear panel of digital i/o versions of System One.

### Lower Right Connector Panel

The **lower-right** connector panel provides a number of BNC connectors that permit monitoring of key generator and analyzer signal points on an external oscilloscope or spectrum analyzer.

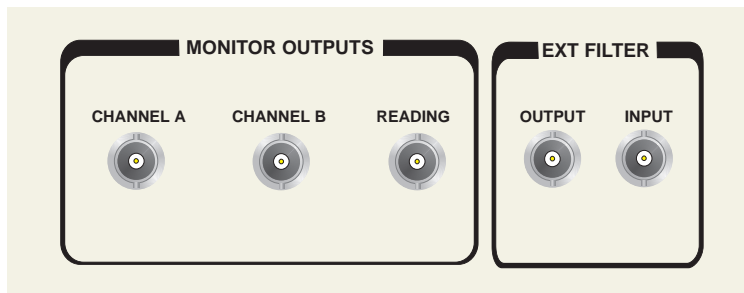


Figure 1-5 Monitor Outputs and External Filter Connections (Lower Right) Connector Panel





# System Two—APWIN Overview

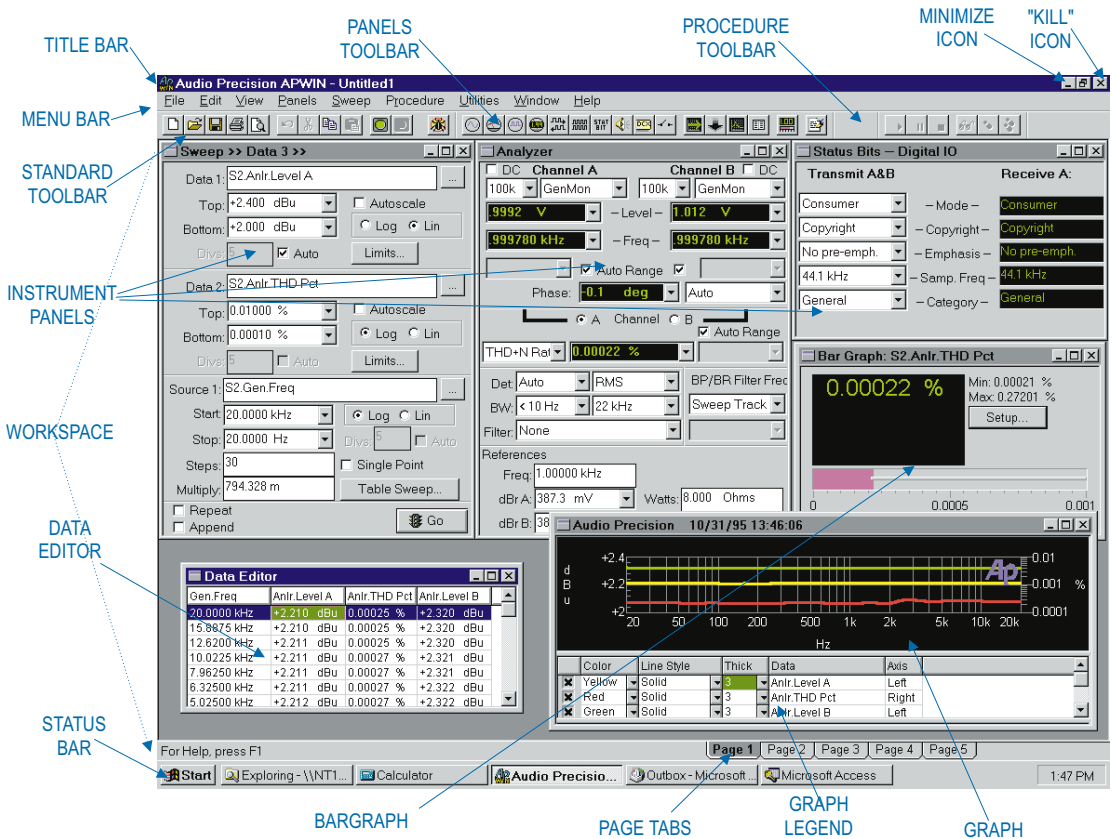


Figure 2-1 APWIN Screen

## Screen Overview

The APWIN Screen consists of the Menu Bar, up to four Toolbars (Standard Toolbar, Panels Toolbar, Procedures Toolbar, and Learn Mode Toolbar), a main Workspace, and the Status Bar with Page Tabs.

### Menu Bar

The Menu bar is immediately below the APWIN Title Bar. It has ten top-level Menu categories—File, Edit, View, Panels, Sweep,

Compute, Procedure, Utilities, Window, and Help. Each category has sub-categories, and in many cases there are further levels below sub-categories. Menu categories are displayed in a drop-down list by clicking on the category name with the left mouse button, or from the keyboard by pressing the **ALT** key simultaneously with the key corresponding to the underlined character in the Menu name. For example, pressing **ALT** **P** selects the Panels category. To select a sub-category, click on the sub-category name with the mouse or press the key corresponding to the underlined character of the sub-category name. To leave a category without making a selection, use the **ESC** key or click anywhere outside the drop-down list box.

The Menu categories and sub-categories are described in individual sections.

## Main Workspace

The main workspace occupies all the screen except for Menu bar, Toolbar, and Status bar. Instrument control panels, graphs, and bargraphs may be freely located in the main workspace. There are five pages of main workspace available, selectable by the **CTRL** **PG**↑ and **CTRL** **PG**↓ keys, by the **CTRL** **1** through **CTRL** **5** key combinations, from the View menu, or by clicking on the Page tabs in the Status Bar/Tabs.

## Status Bar/Tabs

The Status Bar and page tabs are located across the bottom of the APWIN screen. The most common message in the Status area is “For Help, press F1”. When the mouse cursor is located on a Toolbar icon for approximately one second (without pressing a mouse button), the Status Bar displays a short message explaining the function of that particular icon.

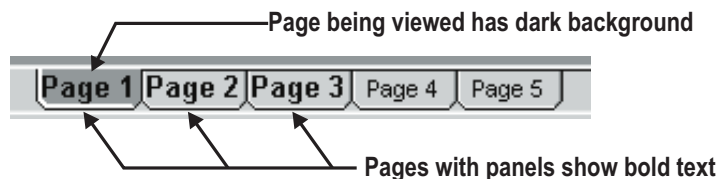


Figure 2-2 Page Tabs on Status Bar

The Page tabs are displayed at the right center of the bar. Clicking on a page tab will cause the contents of that page of the main workspace to be immediately displayed. If one or more panels are displayed on a page, the label on the page tab is displayed in bold characters. The **CTRL** **PG**↑ and **CTRL** **PG**↓ keys may also be used to step sequentially from page to page, or the **CTRL** **1** through **CTRL** **5** key combinations used to jump directly to a page.

## Toolbar Overview

Four toolbars are primary features of the APWIN screen. These are the Standard Toolbar, the Panels Toolbar, the Procedure Toolbar and the Learn Mode Toolbar. Each toolbar consists of a number of icons that can be clicked to quickly initiate an action or jump to a commonly-used function or panel of APWIN software. When the mouse cursor is held on a Toolbar icon for approximately one second (without pressing a mouse button), a small yellow box (“Tool Tip”) appears at the mouse cursor with the name of the icon, while the Status Bar at the bottom left of the screen also describes the function of the icon.



Figure 2-3 Tool Tip When Mouse Cursor Rests Over Icon

Each toolbar is “floating” and can be dragged to any desired location on the panel. When the toolbar is dragged to either side of the screen until the mouse cursor touches the screen boundary, the toolbar becomes vertical. It can later be re-oriented as horizontal by dragging it until the mouse cursor touches a top or bottom screen boundary. On screens with sufficient resolution, it is normally most desirable to drag the toolbars to an end-to-end orientation on the same horizontal row. On lower resolution screens, it may be more desirable to orient them in two rows so that scrolling is not necessary to reach any icon. Each toolbar may be displayed or hidden by use of the View Toolbar menu command. If a toolbar is pushed completely off screen by dragging another toolbar, the “lost” toolbar may be returned to the screen by first unchecking its name in the View menu, then re-opening the View menu and re-checking the toolbar name.

## Standard Toolbar



The Standard Toolbar icons are briefly described below:



New: click on this icon to remove all present panel settings and measurement data and restore the normal APWIN start-up panel settings and panel locations on pages, exactly equivalent to the menu command File New Test



Open: click on this icon to display a dialog for navigating through directories (folders) and selecting test file names, exactly equivalent to the menu command File Open Test



Save: click on this icon to immediately save the current test (.AT2) if a name has already been established (equivalent to File Save Test command). If a name has not been established, the Save As dialog will be displayed which permits navigating through directories (folders) and supplying a test file name, exactly equivalent to the menu command File Save As Test






Save All: click on this icon to immediately save the current test (.AT2) and all open procedures (.APB)









Print: click on this icon to produce a print-out of the current graph according to the settings of the Print Setup and Page Setup dialogs





Print Preview: click on this icon to display the Print Preview screen. If you wish to change attributes of the print-out, use the menu File Page Setup or File Print Setup commands first



Undo: click on this icon  to un-do the last text editing action, equivalent to the Edit Undo ( ) menu command

Cut: click on this icon  to remove the presently selected text to the Windows clipboard, exactly equivalent to the menu command Edit Cut ( )

Copy: click on this icon  to copy the presently selected text to the Windows clipboard without removing it from its present location, exactly equivalent to the menu command Edit Copy ( )

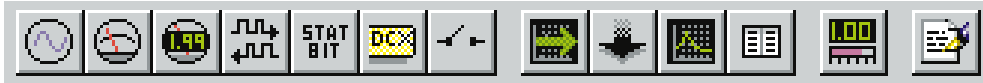
Paste: click on this icon  to copy any text presently on the Windows clipboard to the present mouse cursor location, exactly equivalent to the menu command Edit Paste ( )

Run Sweep (Go): click on this “green light” icon  to start a new sweep, exactly equivalent to the menu command Sweep Start, the  function key, or clicking on the GO button at the bottom of the Sweep Panel


Stop Sweep: click on this “red light” icon  to stop the present sweep, exactly equivalent to the menu command Sweep Stop, the  key, or clicking on the STOP button at the bottom of the Sweep Panel. The toolbar Stop icon and the STOP button on the Sweep panel are available only when a sweep is in progress.


## Panels Toolbar


The Panels Toolbar provides a fast route to displaying any instrument panel and other test-related panels such as Sweep and Sweep Settling. Many of the panels are available in both small and large size versions. Simply clicking on an icon causes it to be displayed





in the small version. Holding down the **SHIFT** key while clicking on the icon produces a display of the large version (if there are two versions). The Panels icons are:


Analog Generator: click on this icon  to display the Analog Generator panel, exactly equivalent to the menu command Panels Analog Generator or the **CTRL G** keystrokes.



Analog Analyzer: click on this icon  to display the Analog Analyzer panel, exactly equivalent to the menu command Panels Analog Analyzer or the **CTRL A** keystrokes.

Digital Generator: click on this icon  to display the Digital Generator panel, exactly equivalent to the menu command Panels Digital Generator or the **CTRL D** keystrokes.

Digital Analyzer: click on this icon  to display the Digital Analyzer panel, exactly equivalent to the menu command Panels Digital Analyzer or the **CTRL Y** keystrokes. Digital features are available only on DSP-based units (System Two+ DSP and System Two Dual Domain)


Digital I/O.: click on this icon  to display the Digital Input/Output panel, exactly equivalent to the menu command Panels Digital I/O or the **CTRL I** keystrokes. Digital features are available only on DSP-based units

Sync/Ref Input: click on this icon  to display the Sync/Ref Input panel, exactly equivalent to the menu command Panels Sync/Ref Input. This panel is functional only on System Two Dual Domain units


Status Bits: click on this icon  to display the Status Bits panel, exactly equivalent to the menu command Panels Status Bits or the  keystrokes. The Status Bits panel is functional only with a System Two Dual Domain unit and AES/EBU or consumer interfaced digital devices under test


Headphone/Speaker: click on this icon to display the Headphone/Speaker panel, exactly equivalent to the menu command Panels Headphone/Speaker

DCX: click on this icon  to display the DCX-127 panel, exactly equivalent to the menu command Panels DCX

Switcher: click on this icon  to display the switcher control panel, exactly equivalent to the menu command Panels Switcher

Sweep: click on this icon  to display the Sweep Panel, exactly equivalent to the menu command Panels Sweep or the  keystrokes.

Sweep Settling: click on this icon  to display the Sweep Settling panel, exactly equivalent to the menu command Panels Sweep Settling


Graph: click on this icon  to display a graph window, exactly equivalent to the menu command Panels Graph

Data Editor: click on this icon  to display the Data Editor, exactly equivalent to the menu command Panels Data Editor







New Bar Graph: click on this icon  to create a new bar graph, exactly equivalent to the menu command Panels Bar Graph


Procedure Editor: click on this icon  to display the Procedure Editor, exactly equivalent to the menu command Panels Procedure Editor or the menu command Procedure Show Panel

## Procedure Toolbar


The Procedure Toolbar icons are:

OLE Automation Browser: click on this icon  to display the OLE Automation Browser, equivalent to the menu command Procedure Browse. This Browser provides an organized list of all APWIN Basic commands for controlling System Two

Run Procedure: click on this icon  to start the procedure presently loaded, exactly equivalent to the menu command Procedure Run


Pause Procedure: click on this icon  to cause the procedure presently running to halt, exactly equivalent to the menu command Procedure Pause

End Procedure: click on this icon  to stop the procedure presently running, exactly equivalent to the menu command Procedure End


Toggle Break Points: click on this icon  to toggle procedure break points on or off, equivalent to the menu command Procedure Toggle Break Points




Quick Watch: click on this icon   exactly equivalent to the menu command Procedure Quick Watch

Step Into: click on this icon , exactly equivalent to the menu command Procedure Step Into



Step Over: click on this icon , exactly equivalent to the menu command Procedure Step Over

Step Out:  
click on this

icon ,  
exactly  
equivalent to  
the menu  
command  
Procedure  
Step Out

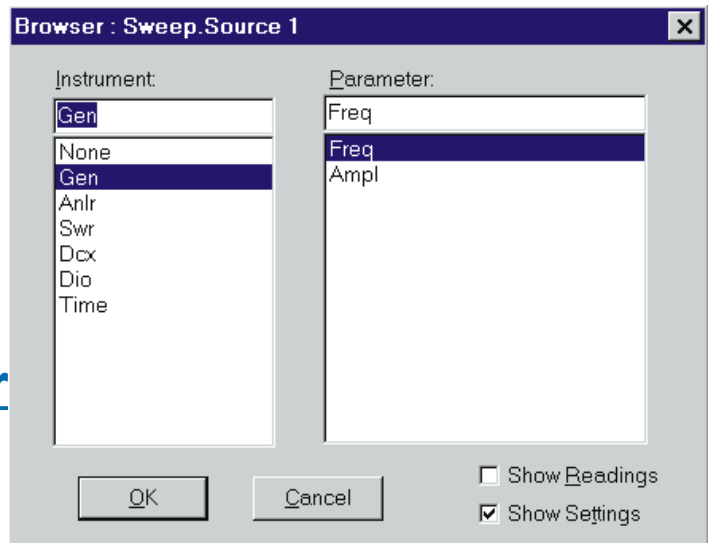



Figure 2-4 Browser (Sweep Source, System One)

## Learn Mode Toolbar

Start  
Learn Mode:  
click on this  
icon , exactly  
equivalent to

the menu commands Procedure Learn Mode or Utilities Learn Mode, to put APWIN into the mode of writing lines of APWIN Basic in the Procedure editor for each user mouse click or panel field entry

Stop Learn Mode: click on this icon , exactly equivalent to un-checking the menu commands Procedure Learn Mode or Utilities Learn Mode, to stop APWIN from writing APWIN Basic code based on user actions.

## Browser

A “Browser” (see Figure 2-4) is a Windows dialog box designed to permit the user to make selections from some organized structure or hierarchy. Browsers at the Sweep panel Source and Data areas permit the user to browse among different “Instrument” and “Parameter” to find the desired entity to be swept or graphed.

File-related browsers permit the user to move

up and down among directories and sub-directories (called Folders in Windows 95) and among disk drives to locate the desired file name. File browsers are displayed in response to many common file-related commands including File Open and File Save As. Browsers in Windows 95 are considerably different in detail from those in Windows 3.11, but both support most of the same functions.

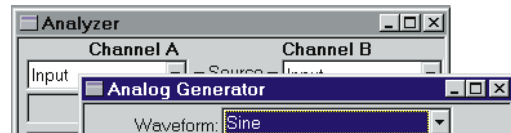


Figure 2-5 Analog Generator Panel Has the Focus, Analyzer Panel Does Not

## Control and Entry Conventions







APWIN uses standard Windows control and data entry conventions. Changes can be made to panel settings only by placing the “focus” on that panel. The “focus” is put on any panel (window) simply by clicking the mouse anywhere on the panel, or by using the   key combination to move the focus through all the panels (windows) displayed on the present Workspace page. The title bar of the panel with



Figure 2-6 Multiple Choice Control

the focus will be blue while the others are gray if the standard Windows default color scheme is used.

Many panels may be displayed in either small or large versions. To change between small and large versions, double click anywhere in the title bar (control bar) of the panel. Or, click the mouse on the icon in the upper right corner (Windows 3.11) or on the center of the three upper right icons (Windows 95). Or from the keyboard, use the   or   keys to change between small and large versions.

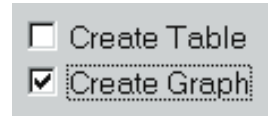








Figure 2-7 Checkbox Controls. Lower is Selected

Multiple-choice fields are indicated by a down arrow at the right end of the field. Clicking the mouse on the down arrow displays the list of available selections. Clicking the mouse on the desired selection makes the change and closes the list. From the keyboard, the  key moves from field to field on a panel, generally in a sequence from upper left to lower right.   moves back upwards on the panel. A selected multiple choice field is indicated by the present entry displayed with inverse video. The  key causes the selection list to display when on a multiple choice field. Operations of the  and  keys then highlight selections, and pressing Enter chooses the highlighted selection.

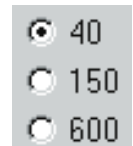













Figure 2-8 Radio Buttons

Check boxes may be checked or un-checked by clicking on them with the mouse, or by moving to them with the  or   keys and pressing the Space bar to check or un-check them. Selection of a check box is indicated by a rectangular outline around the label of the checkbox.

“Radio buttons” are groups of two or more buttons where only one may be selected at any one time. Use the mouse to click directly on the desired radio button. From the keyboard, use the  or   keys to move to a group of radio buttons. Selection is indicated by a rectangular outline around the label of the presently-selected radio button. Use the  or  to change between the buttons in the group.

Numeric entry fields require that new values be typed in from the keyboard, followed by either the  (which leaves the field selected for further entries) or the  or  keys (which move to the following or preceding field).

## Function Keys



Help (context-sensitive if a specific Help topic exists for the panel feature presently selected by the mouse cursor)



When focus is on a bargraph: re-set the maximum and minimum value storage feature of the presently-selected bargraph



When focus is on the Data Editor: permit character-by-character editing of Data Editor cell



Set the analog generator dBr reference value to the present analog generator amplitude setting



Set the analog generator frequency reference value to the present analog generator frequency setting



Set the analog analyzer dBr reference value



Set the analog analyzer frequency reference value to the value presently displayed by the analog analyzer frequency counter



Perform a new transform on the data presently in the DSP acquisition buffer and graph the results according to the present settings of the Sweep and Digital Analyzer panels



Graph the present data in DSP memory according to the present settings of the Sweep and Digital Analyzer panels without performing a new transform



Graph the data presently computer memory



Start a new sweep; equivalent to clicking on the Go icon in the toolbar or on the Sweep panel



Start a new sweep while retaining previous data, appending the new to the old

Start a new repeating sweep



If a sweep is in progress, pause. If a sweep is paused, start it again (toggle)



Turn off all generator outputs (“panic button”)



Turn back on the generator outputs turned off by 


## APWIN Menus

### File Menu

The File Menu consists of the following commands: New, Open, Save, Save As, Save All, Set Working Directory, Export, Import, Print, Print Preview, Print Setup, Page Setup, the list of recently-used test files, Exit

#### File New

The File New command allows the user to create a new Test or Procedure file, or to erase data from the test file presently loaded.

A Test (.AT1) file is the fundamental APWIN file that contains complete setup instructions for all fields of all panels, graphs, and bargraphs. The Test file also contains any test data that was present when the file was saved. A new test file is always presented with the small versions of the Analog Generator and Analog Analyzer on page 1, the small version of the Sweep panel plus a graph on page 2, and the small version of the DIO panel on page 3. The File New Test command may also be issued by clicking on the blank sheet icon  on the Standard toolbar.

A Procedure (.APB) file contains a set of instructions in a visual Basic language, which can initiate and control a sequence of actions including loading test files from disk and executing them, comparing data to limits, taking conditional actions, directly changing instrument settings, and a variety of other tasks.

The File New Data command erases all data presently in memory without altering the test setup.

#### File Open

The File Open command displays a subsidiary command menu which allows the user to load from disk into computer memory any

existing file of the Test, Procedure, Data, Limit Data, EQ Curve, Sweep Tables, and Stereo and Mono Waveform types.

The File Open command can be issued by:



clicking on the File menu entry, then clicking on Open in the list box which descends

or

from the keyboard by  (File Open).

Issuing the File Open command displays a “fly-out” command list that permits selection of the file type to be opened. Selecting the desired file type from this list displays the Open dialog box with the desired file type selected, permitting navigation between directories (and disk drives, if necessary) and selection of the specific file name either by typing the name or clicking on a displayed file name.

### File Open Test

The File Open Test command displays a list of the Test files (.AT1 for System One, .AT2 for System Two) in the current directory. The dialog displayed also permits navigation to other directories or disk drives. A file is opened by selecting the file name and clicking the Open button or by double-clicking the file name. The File Open Test command is identical to clicking on the open folder icon  on the Standard toolbar. The keystrokes  will also display a File Open dialog for Test files, equivalent to File Open Test.

A **Test** (.AT1 or .AT2) file is the fundamental APWIN file that contains complete setup instructions for all fields of all panels, graphs, and bargraphs. The Test file also contains any test data present when the file was saved. A user commonly saves tests whenever a setup or the data resulting from a sweep may be desired at a later time. Audio Precision furnishes a large number of standard, typical tests already set up; see the list of furnished standard test files in the final chapter.



## “Importing” System One Tests

With APWIN operating with System One, only System One (.AT1) files may be opened. When operating with System Two, it is also possible to open System One (.AT1) files in addition to the normal opening of System Two (.AT2) files. Click the down arrow at the right of the “Files of Type” field to select whether .AT2 or .AT1 files will be loaded. Thus, System One files may be imported and then saved as System Two files. Since DOS files (.TST, .LIM, .EQ, etc.) may be imported into APWIN when operating in System One mode by the File Import S1.EXE Test command and saved as .AT1 files, this provides a path from old DOS tests to System Two tests. In a few instances it may be necessary to make modifications to an imported .AT1 file before it will operate System Two hardware correctly, since the architectures of the two instruments are slightly different.

## File Open Procedure




The File Open Procedure command displays a list of the Procedure files (.APB file type) in the current directory. The dialog displayed also permits navigation to other directories or disk drives. A file is opened by selecting the file name and clicking the Open button or by double-clicking the file name.

A **Procedure** (.APB) file contains a set of instructions in a visual Basic language, which can initiate and control a sequence of actions including loading test files from disk and executing them, comparing data to limits, taking conditional actions, directly changing instrument settings, and a variety of other tasks.

## File Open Data

The File Open Data command displays a list of the Data files (.ADA file type) in the current directory. The dialog displayed also permits navigation to other directories or disk drives. A file is opened by selecting the file name and clicking the Open button or by double-clicking the file name.

A **Data** (.ADA) file contains test data only, with no information on panel setup and no definition of display format. To view the

newly-loaded data graphically, press the  function key or use the Sweep Compare Data to Limits () command. To view the new data in a tabular presentation, click on the Data Editor icon  or use the Panels Data Editor menu commands.

### File Open Limit Data

The File Open Limit Data command displays a list of the Limit files (.ADL file type) in the current directory. The dialog displayed also permits navigation to other directories or disk drives. A file is opened by selecting the file name and clicking the Open button or by double-clicking the file name.

A Limit Data file is a form of a data file used to define upper or lower limits of acceptable performance, so that automatic pass/fail decisions may be made by the software. See the Limits chapter starting on page 17-1 for more information on Limits.

### File Open Sweep Tables

The File Open Sweep Tables command displays a list of the Sweep Table files (.ADS file type) in the current directory. The dialog displayed also permits navigation to other directories or disk drives. A file is opened by selecting the file name and clicking the Open button or by double-clicking the file name.

A Sweep Table is a form of a data file used to define a list of specific Source 1 values which will be used during a sweep, as opposed to the software computing Source 1 values based on the Start and Stop values, Log/Lin selection, and number of steps or step size. See the Table Sweeps section starting on page 15-12 for more information.

### File Open EQ Curve

The File Open EQ Curve command displays a list of the EQ Curve files (.ADQ file type) in the current directory. The dialog displayed also permits navigation to other directories or disk drives. A file is opened by selecting the file name and clicking the Open button or by double-clicking the file name.

An EQ (equalization) Curve is a form of a data file used to define a relationship between amplitude and frequency. It may be used to automatically control a generator amplitude during a sweep so as to follow the equalization curve. See the equalization sections of the Analog Generator chapter and the Digital Generator section of the DSP Overview chapter for more information. The EQ Curve may also be used by the Compute Equalize function to modify data following a sweep to the values it would have had if the equalization curve had been in use. See the Compute Menu chapter for more information. EQ Curves furnished by Audio Precision are located in the C:\APWIN\EQ directory.

### File Open Stereo Waveforms

The File Open Stereo Waveforms command displays a list of previously-acquired and saved Stereo Waveform files (.AAS file type) in the current directory. The dialog displayed also permits navigation to other directories or disk drives. A file is opened by selecting the file name and clicking the Open button or by double-clicking the file name.

Stereo waveforms files are files containing two channels of previously-acquired signal which had been saved to disk for later use. The Open Stereo Waveform command loads such a file into both channels of the DSP analyzer where it may be further viewed in time or frequency domain representations. For more information, see the Saving and Loading waveforms sections of the Spectrum Analyzers and Multitone Analyzers chapters.

### File Open Mono Waveforms


The File Open Mono Waveform command displays a list of the Mono Waveform files (.AAM) in the current directory. The dialog displayed also permits navigation to other directories or disk drives. A file is opened by selecting the file name and clicking the Open button or by double-clicking the file name.

Mono waveforms files are files containing one channel of previously-acquired signal which had been saved to disk for later use. The Open Mono Waveform command loads such a file into either channel (as selected by the user) of the DSP analyzer where it may be

further viewed in time or frequency domain representations. For more information, see the Saving and Loading waveforms sections of the Spectrum Analyzers and Multitone Analyzers chapters.

## File Save Test

---

The File Save command displays a subsidiary “fly-out” menu with the further choices of Test and Procedure. The File Save Test command allows the user to save to disk the test setup information and data currently in memory. The File Save Test command may also be issued by clicking on the diskette icon  on the Standard toolbar

or


from the keyboard by pressing  (for File Save Test).

If the test in memory was previously loaded from disk (via File Open) or previously saved and thus already has a name, the current information will be saved under that existing name, over-writing the previous version of the same named test.

If no test name has been assigned, the Save As dialog box appears. The Save As dialog box is similar to the File Open dialog box.

## File Save Procedure

---

The File Save command displays a subsidiary “fly-out” menu listing the choices Test and Procedure. Selecting the Procedure command saves the procedure presently in the Procedure Editor to disk. The File Save Procedure command may also be issued by the keystrokes  or by clicking on the diskette icon on the Procedure Editor panel.


## File Save As

---

Issuing the File Save As command displays a “fly-out” command list which permits selection of the file type to be saved from the choices Test (.AT1), Procedure (.APB), Data (.ADA), Limit Data (.ADL), Sweep Tables (.ADS), EQ Curve (.ADQ), Stereo Waveform (.AAS), or Mono Waveform (.AAM). Selecting the desired file type from this list displays the Save As dialog box, permitting navigation between directories (and

disk drives, if necessary) and displaying existing file names of the type being saved.

If the file presently in memory was previously loaded from disk (via File Open) or previously saved and thus already has a name, that name will be suggested. If accepted by the user, current data will be saved under that existing name, over-writing the previous version of the same named test.

A test file includes all instrument setups, Sweep panel and Sweep Settling settings, Graph settings, Page Setup settings for graph print-out, etc., etc., plus measurement data from the last  operation before saving. Data, Limits, Sweep, and EQ files save only the measurement data and certain Sweep panel information. The Procedure file saved consists of the contents of the Procedure Editor. The Waveform selections save the contents of one or both FFT Acquisition Buffers as selected in the dialog box displayed after the file name is entered or selected. Stereo Waveform files have two waveforms and Monaural Waveforms contain only one. For more information, see the Saving and Loading waveforms sections of each FFT-based program chapter.

If the present contents of memory (test, data, procedure, waveform, etc.) was previously loaded from disk (via File Open) or previously saved and thus already has a name, that name will be suggested. If accepted by the user, current data will be saved under that existing name, over-writing the previous version of the same named test.

### **File Save As Test**

The File Save As Test command permits saving present instrument and sweep setup information, plus any test data present if a test has already been run, as a Test file (.AT1). The file will be saved in the current directory or the dialog permits navigation to other directories or disk drives.

### **File Save As Procedure**




The File Save As Procedure command permits saving the contents of the Procedure Editor as a Procedure file (.APB file type) in the current

directory. The dialog displayed also permits navigation to other directories or disk drives.

A Procedure (.APB) file contains a set of instructions in a visual Basic language, which can initiate and control a sequence of actions including loading test files from disk and executing them, comparing data to limits, taking conditional actions, directly changing instrument settings, and a variety of other tasks.

### File Save As Data

The File Save As Data command permits saving the present test results data as a Data file (.ADA file type) in the current directory. The dialog displayed also permits navigation to other directories or disk drives.

A Data (.ADA) file contains test data only, with no information on panel setup and no definition of display format. To view the newly-loaded data graphically, press the  function key or use the Sweep Compare Data to Limits () command. To view the new data in a tabular presentation, click on the Data Editor icon  or use the Panels Data Editor menu commands.

### File Save As Limit Data

The File Save As Limit Data command permits saving the data points presently in the Data Editor as a Limit file (.ADL file type) in the current directory. The dialog displayed also permits navigation to other directories or disk drives.

A Limit Data file is a form of a data file used to define upper or lower limits of acceptable performance, so that automatic pass/fail decisions may be made by the software. For more information on Limits, see the Limits chapter beginning on page 17-1.

### File Save As Sweep Tables

The File Save As Sweep Tables command permits saving the data points presently in the Data Editor as a Sweep Table file (.ADS file type) in the current directory. The dialog displayed also permits navigation to other directories or disk drives.

A Sweep Table is a form of a data file used to define a list of specific Source 1 values which will be used during a sweep, as opposed to the software computing Source 1 values based on the Start and Stop values, Log/Lin selection, and number of steps or step size. See the Table Sweeps section on page 15-12 for more information.

### **File Save As EQ Curve**

The File Save As EQ Curve command permits saving the data points presently in the Data Editor as an EQ Curve file (.ADQ file type) in the current directory. The dialog displayed also permits navigation to other directories or disk drives.

An EQ (equalization) Curve is a form of a data file used to define a relationship between amplitude and frequency. It may be used to automatically control a generator amplitude during a sweep so as to follow the equalization curve. See the equalization sections of the Analog Generator chapter and the digital generator section of the Digital Domain Audio Analyzer (GENANLR.AZ1) chapter for more information. The EQ Curve may also be used by the Compute Equalize function to modify data following a sweep to the values it would have had if the equalization curve had been in use. See the Compute Equalize section later in this chapter for more information. EQ Curves furnished by Audio Precision are located in the C:\APWIN\EQ directory.

### **File Save As Stereo Waveform**

The File Save As Stereo Waveform command permits saving to disk the sampled signal presently in both channels of the DSP acquisition buffers. The resulting disk file may later be downloaded to the DSP via the Open Stereo Waveform command so that it may be further viewed in time or frequency domain representations. For more information, see the Saving and Loading waveforms sections of each FFT-based program chapter.

### **File Save As Mono Waveform**

The File Save As Mono Waveform command permits saving to disk the sampled signal presently in one channel of the DSP acquisition buffers.

The resulting disk file may later be downloaded to the DSP via the Open Stereo Waveform command so that it may be further viewed in time or frequency domain representations. For more information, see the Saving and Loading waveforms sections of each FFT-based program chapter.

## File Save All

---

The File Save All command saves to disk both the current test and all currently-open procedures. The File Save All command may be issued by:

Selecting File menu and the Save All command

or

Typing    (File save aL) from the keyboard

or

Clicking on the multi-diskette icon  on the Standard Toolbar

## File Set Working Directory

---

APWIN tests often include a number of linked files, including Sweep Tables, Limit files, and (for DSP versions) generator waveform files. The “Current Working Directory” is the directory where APWIN expects to find files that do not have an explicit path name supplied. The Set Working Directory command displays a disk and sub-directory navigation dialog that permits the user to specify the current working directory. When a test file is opened, APWIN expects to find related files in the current working directory. This supports one possible disk organization philosophy of grouping the test files, limit files, sweep tables, etc. for different testing activities into different directories, such as a power amplifier directory or a directory for a specific product model number.

If an explicit path name is furnished for a file when a test is set up, APWIN expects to find the file in that specific directory instead of the current working directory. If, for example, different directories have been set up for different products or models but all use the same



generator waveform file, the waveform file name and explicit path may be specified as each test is set up so that it is not necessary to store duplicate copies of the waveform file in many directories.

By default, all File Open commands initially display the current working directory contents. The user may navigate to other directories if desired. The File Open command with the Test or Procedure file types specified will also set the current working directory. The File Save As command is independent from the current working directory and will display the contents of the last directory used by this command.

The current working directory is stored when exiting from APWIN and will automatically be used the next time APWIN is started. Note that the current working directory established for APWIN is independent of the current working directory of all other Windows applications.

## File Export

The File Export menu command displays a “fly-out” dialog with two additional choices—ASCII Data and Graphics.

### File Export ASCII Data

This command saves the test data presently in memory in the Export Data File format (.ADX file type) in order to easily import it into other software programs. The .ADX file consists of ASCII characters with commas as delimiters. The basic structure is similar to the format displayed in the Data Editor. For standard (not nested) sweeps, the columns from left to right are Source 1 followed by the Data columns. For a nested sweep, the columns are Source 1, the Data columns, and Source 2. Examples of both standard and nested sweep .ADX files are shown below for cases of two Data parameters, Anlr Level A at Data 1 and Anlr THD at Data 2. Use of Data 3 through Data 6 would add additional columns. Unused data columns are represented by commas.

Normal Sweep

C:\APWIN\SAMPLES\EXAMPLE.AT1, 08/22/95 11:09:55

Gen.Freq,	Anlr.Level A,	Anlr.THD Pct,	,,,,,
Source 1,	Data 1,	Data 2,	,,,,,
Hz,	dBV,	%,	,,,,,
20000,	0.038598,	0.000387,	,,,,,
5025,	0.031718,	0.000426,	,,,,,
1262,	0.024833,	0.000348,	,,,,,
317,	0.028276,	0.000348,	,,,,,
79.625,	0.024833,	0.00031,	,,,,,
20,	0.020526,	0.000378,	,,,,,

### Nested Sweep

C:\APWIN\SAMPLES\EXAMPLE2.AT1, 08/22/95 11:13:23

Gen.Freq,	Anlr.Level A,	Anlr.THD Pct,	,,,,,	Gen.Ampl
Source 1,	Data 1,	Data 2,	,,,,,	Source 2
Hz ,	dBV,	%,	,,,,,	Vrms
20000,	0.031718 ,	0.000387,	,,,,,	1
5025,	0.024833 ,	0.000426,	,,,,,	1
1262,	0.021388,	0.000349,	,,,,,	1
317,	0.024833 ,	0.000348,	,,,,,	1
79.625,	0.017942 ,	0.000349,	,,,,,	1
20,	0.018803 ,	0.00032,	,,,,,	1
20000,	7.990518 ,	0.00034,	,,,,,	2.5
5025,	7.990518 ,	0.000341,	,,,,,	2.5
1262,	7.98501,	0.000279,	,,,,,	2.5
317,	7.98501,	0.000279,	,,,,,	2.5
79.625,	7.97674 ,	0.000372,	,,,,,	2.5
20,	7.979499,	0.000287,	,,,,,	2.5

### File Export Graphics

The File Export Graphics command creates a Windows Metafile or Enhanced Metafile copy of the present graph as a disk file. A dialog permits navigation to any desired directory and supplying a file name for the Metafile. A Metafile is a standard graphics format of the vector

type, as opposed to bitmaps. Metafiles can be imported into most drawing, desktop publishing, and word processing programs. Resolution of a Metafile depends only on the output device, as opposed to a bitmap whose resolution in pixels is fixed at the moment of capture. The Enhanced Metafile (.emf file type) contains more information than the original Windows Metafile (.wmf file type) and is preferred when the program into which it will be imported will accept it. Older Windows programs may accept only the original (.wmf) format. The parameters of the Metafile graph (trace line styles, trace color, trace thickness, whether comments and legend are part of the metafile, etc.) are controlled from the File Page Setup screen.

As an alternate to the Metafile for graph export, bitmap copies of graphs (and other panels) may be copied to the Windows clipboard by the Edit Copy to Clipboard command.

## File Import

The File Import command produces a menu of three subsidiary selections: ASCII data, S1.EXE test, and S1.EXE procedure. The two S1.EXE choices will be gray and unavailable when APWIN has been started for System Two, but are available when started for System One.

### Importing ASCII Data

Popular spreadsheet software, text editors, and the Audio Precision MAKEWAV2 utility (for generating multitone waveform files) can generate ASCII data files. The S1.EXE files may be imported only if APWIN has been started in its System One mode.

The File Export ASCII Data description above shows examples of the format of the comma-delimited file format. Spreadsheet software can create such comma-delimited files, with the first four lines of text typed in manually or pasted in from an .ADX file via the Windows clipboard.

The File Import dialog box permits the user to specify whether .ADX or .ADF files are shown in the browser. The format of these two file types is identical. The .ADX file type is automatically created when the File Export command is used from APWIN. The .ADF file type is

automatically created when MAKEWAV2 generates its list of exact fundamental frequencies in a multitone file.

## Importing S1.EXE Files

Importing S1.EXE files makes it possible to transfer test setups originally prepared with DOS software to the Windows environment with APWIN. They may then be saved as .AT1 files. Importing a DOS procedure will also automatically import the test, limit, sweep, etc. files, resulting in a complete operating procedure for System One under APWIN.

## File Print

---


To print a graph:

select the File Print command under the File Menu

or

click on the Print icon 


or

from the keyboard, press the   key combination

or

from the key board, press    (for File Print).

The File Print command displays the standard Windows Print dialog box. The features of this dialog box vary somewhat between Windows 3.11 and Windows 95. Both permit selection of the number of copies and show which printer will be used. The Setup button of the Windows 3.11 Print dialog and the Properties button of the Windows 95 both give access to additional features including Landscape versus Portrait orientation of the graph on the page (which will be overridden if the Portrait/Landscape selection is changed on the Page Setup screen), and the paper tray to be used on multiple-tray printers..

When all selections are made, click on the OK button or press the  key. A message will be displayed while APWIN sends the data to Windows and the printer.

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## File Print Preview

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See the Graphs and Printing chapter section starting on page 16-20.

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## File Print Setup

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The File Print Setup command permits selection of the printer to be used, the paper orientation (landscape versus portrait), and the paper size and source on printers which have multiple paper trays. To display the Print Setup dialog box:

select Print Setup under the File menu

or

from the keyboard, press the    keys (for File pRint setup).

---

## File Page Setup

---

See the Page Setup section of the Graphs and Printing chapter starting on page 16-22.

---

## File Recent File

---

The names of the eight most-recently-used test files (.AT1 or .AT2) are listed in the lower section of the File menu. Any of those files (for the instrument type presently used) may be opened by a single click of the left mouse button on the file name. This is particularly convenient when working repetitively with a small number of test files, particularly if some of them are located in different directories (folders) from others.

---

## File Exit





---

The File Exit command terminates an APWIN session. To execute this command:

click on Exit under the File menu

or

from the keyboard, press    (for File eXit).

Other alternative methods for ending an APWIN session under Windows 3.11 include double-clicking the  icon in the upper left corner of the screen, or single-clicking that icon and selecting Close, or using the   keystrokes. Under Windows 95, APWIN may also be terminated by clicking the  icon at the upper right of the screen.

## APWIN File Types

APWIN uses a number of different types of files, distinguished both by their internal structure and by the file extension (three characters following the “.”). All extensions use the letter “A” (for Audio Precision) as the first character. They are:

### Test Files

Test File, System One	.AT1
Test File, System Two	.AT2

### Data Files

Test Results Data	.ADA
Limits Data	.ADL
Sweep (Step) Table	.ADS
EQ Curves	.ADQ
Exported (delimited ASCII) Data	.ADX
MAKEWAV2 Exact Fundamental Frequency List	.ADF

### DSP Files

DSP Program, Sys One	.AZ1
DSP Program, Sys Two	.AZ2
Acquired Waveform, 1 channel	.AAM
Acquired Waveform, 2 channels	.AAS
Generator Waveform, 1 channel	.AGM
Generator Waveform, 2 channels	.AGS

### Miscellaneous Files



APWIN Basic Procedure	.APB
Log File	.ALG

Note that Data, Limit, Sweep, and EQ Curve files all have identical formats and may be used interchangeably.



## Edit Menu

The Edit Menu consists of the following commands: Undo, Cut, Copy, Paste, Copy Panel to Clipboard, Set Analyzer dBr Reference, Set Analyzer Frequency Reference, Set Generator dBr Reference, Set Generator Frequency Reference, Reset Bargraph Max/Min, Insert Row Before, Insert Row After, Insert Row at End, and Delete Row.



### Undo

The Edit Undo command (equivalent to the  icon on the Standard Toolbar or the  keystrokes) will undo the last deletion or typing action in the various text editors such as the Comments Editor or Procedure editor. A second operation will undo the undo.



### Cut

The Cut command (equivalent to the  icon on the Standard Toolbar or the  keystrokes) removes selected (inverse video highlighted) text from an editor such as the Comments or Procedure editors, or from a numeric entry field, to the Windows clipboard. The text may then be pasted into another location or another Windows program.

### Copy

The Copy command (equivalent to  icon on the Standard Toolbar or the  keystrokes) copies selected (inverse video highlighted) text from an editor such as the Data Editor, Comments Editor, or Procedure Editor, or from a numeric entry field on a panel, to the Windows clipboard. The text may then be pasted into another location or into another Windows program.




### Paste from clipboard

The Paste command (equivalent to  icon on the Standard Toolbar or the  keystrokes) duplicates at the current cursor location the text presently on the Windows clipboard. That clipboard text would have been placed there by using the Cut or Copy command

to bring the text from another location within APWIN, or from another Windows program. Note that pasting numeric data with “illegal” units into a numeric entry field will result in an error message and retention of the previous entry. For example, it is not possible to paste a frequency value in Hz units into an Amplitude field.

## Copy Panel to Clipboard


---

The Copy Panel to Clipboard command in the Edit menu will copy (as a bitmap) to the Windows clipboard the APWIN panel, graph, bargraph, or editor (Window) which presently has the focus. The focus is placed on a panel, graph, etc. by clicking on the panel or, if no mouse is available, using the  keystroke. The panel with the focus has a blue title bar at the top. From the clipboard, it may then be copied into any Windows-compliant application by the Paste command of that application (or ). Thus, a bitmap copy of a graph, bargraph, editor, or instrument panel may be easily placed into a word processing document or paint or graphics program for further processing or printing. A bitmap is a fixed-resolution format, with the pixel relationship determined when the panel is captured. Note that the  keystroke will not copy a panel or graph since that keystroke is defined for text copy among APWIN’s editors; the Edit Copy Panel to Clipboard command must be used.

Vector graphics print to the full resolution of the output device. If a vector file of the APWIN graph is desired, see the File Export Graphic command.

## Set Analyzer dBr Ref

---

The Set Analyzer dBr Reference command (equivalent to the  key) causes the presently-measured analog amplitude value to be written into the dBr reference field near the bottom of the large form of the Analog Analyzer panel. This is often referred to as “setting zero dBr”. Since both the Reading meter and Level meters may be reading amplitude, the following priorities are used in determining which meter reading is automatically entered into the dBr Reference field:


If the Reading meter is not in dBr units, the Level meter measurement will be written into the dBr Reference field. If the



Reading meter unit is dBr, the Reading meter's measurement will be written into the reference field.


### Set Analyzer Freq Ref

---

The Set Analyzer Frequency Reference command (equivalent to the  keystrokes) causes the presently-measured frequency value to be written into the Frequency Reference field near the bottom of the large form of the Analog Analyzer panel.


### Set Generator dBr Ref

---

The Set Generator dBr Reference command (equivalent to the  key) causes the presently-set analog generator output amplitude to be written into the dBr Reference field near the bottom of the large form of the Analog Generator panel.


### Set Generator Freq Ref

---

The Set Generator Frequency Reference command (equivalent to the  keystrokes) causes the presently-set analog generator frequency to be written into the Frequency Reference field near the bottom of the large form of the Analog Generator

### Reset Bar Graph Max/Min

---

The Reset Bar Graph Max/Min command (equivalent to the  function key) causes the maximum and minimum hold feature of the Bargraph which has the Windows focus to reset to the currently-measured value. Windows focus is indicated by the title bar across the top of the bargraph being blue rather than gray. Clicking anywhere on a bargraph brings the Windows focus to it.

### Insert Row Before

---

When the Data Editor has the focus (Data Editor Title Bar is blue), the Insert Row Before command will insert a new row into the Data Editor, immediately before the presently-selected row. The same function is also available as a right mouse button selection when the mouse cursor is in the Data Editor.

## Insert Row After

---

When the Data Editor has the focus (Data Editor Title Bar is blue), the Insert Row After command will add a new row into the Data Editor, immediately after the currently-selected row of data. The same function is also available as a right mouse button selection when the mouse cursor is in the Data Editor.

## Add Row to the End

---

When the Data Editor has the focus (Data Editor Title Bar is blue), the Add Row to the End command will add a new row of data at the end of the Data Editor. The same function is also available as a right mouse button selection when the mouse cursor is in the Data Editor.

## Delete Row

---

When the Data Editor has the focus (Data Editor Title Bar is blue), the Delete Row command will delete the currently-selected row of data. The same function is also available as a right mouse button selection when the mouse cursor is in the Data Editor.

## View Menu

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The View Menu consists of the following commands: Standard Toolbar, Panel Toolbar, Procedure Toolbar, Learn Mode Toolbar, Status Bar, Page 1, Page 2, Page 3, Page 4, Page 5

## Standard Toolbar

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The Standard Toolbar contains icons for common activities such as opening and saving files, printing graphs, and copying to and pasting from the Windows Clipboard. Checking or unchecking the View Standard Toolbar command will display or hide the Standard Toolbar. It is normally most convenient to have toolbars visible, but it may be a worthwhile compromise to hide some or all to get more workspace on lower resolution displays such as 640 x 480 pixels. Toolbars may be dragged to another location by clicking and holding the left mouse button on any toolbar area that is not part of an icon. With higher resolution screens, it is normally desirable to drag all toolbars onto the same line rather than having them one above another. Toolbars may

also be re-oriented to vertical orientation by dragging them to the left or right margin of the screen, until the outline visible while dragging changes to a vertical box. If a toolbar is pushed completely off screen by dragging another toolbar, the “lost” toolbar may be returned to the screen by first unchecking its name in the View menu, then re-opening the View menu and re-checking the toolbar name.

## Panel Toolbar

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The Panel Toolbar contains icons for every instrument panel plus general panels such as Sweep Settling. Checking or unchecking the View Panel Toolbar command will turn the Panel Toolbar on or off. It may be dragged as described above under Standard Toolbar.

Many of the panels are available in both small and large size versions. Simply clicking on an icon causes it to be displayed in the small version. Holding down the  key while clicking on the icon produces a display of the large version (if there are two versions).

## Procedure Toolbar

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The Procedure Toolbar contains icons for activities including OLE Automation Browser, Run Procedure, Pause Procedure, End Procedure, Toggle Break Points, Quick Watch, Step Into, Step Over, and Step Out. Checking or unchecking the View Procedure Toolbar command will turn the Procedure Toolbar on or off. It may be dragged as described above under Standard Toolbar.

## Learn Mode Toolbar

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The Learn Mode Toolbar (illustrated in chapter 2) contains icons to start or stop Learn Mode. When Learn Mode is activated, operator actions including the result of mouse clicks, menu selections, and text or numeric entries into panel fields, will result in lines of APWIN Basic language code being automatically written into the Procedure Editor. The resulting procedure can then be run to re-create the series of actions. Checking or unchecking the View Learn Mode Toolbar command will turn the Learn Mode Toolbar on or off. It may be dragged as described above under Standard Toolbar.

## Status Bar







The Status Bar at the bottom of the APWIN screen displays at the left short messages about the present activity or the icon under the present mouse cursor location. In the illustration, the mouse cursor was on the Analog Analyzer icon on the Panels Toolbar, producing the message at the left end. At the right center, the Status Bar displays the five page tabs. If one or more panels are displayed on a page, the label on the page tab will be in bold characters. Checking or unchecking the View Status Bar command will turn the Status Bar on or off. It is normally most convenient to have the Status Bar visible, but to get more workspace on lower resolution displays such as 640 x 480 pixels, it may be a worthwhile compromise to hide the Status Bar.

## Page 1 through Page 5

The View Page 1 through View Page 5 commands display the specified page of the workspace. They have the same effect as clicking on the Page 1 through Page 5 tabs in the Status Bar or using the

 through  keystrokes.

## Panels Menu

The System One Panels Menu consists of the following commands: Analog Generator () , Analog Analyzer () , Digital Analyzer, () , Sweep () , Graph, Sweep Settling, DCX, Switcher, Bar Graphs, Data Editor, Digital I/O () , Status Bits () , Procedure Editor, Diagnostic, and Regulation.

Detailed information on the panels may be found in the chapters indicated:

Analog Generator	Analog Generator chapter 4
Analog Analyzer	Analog Analyzer chapter 5
Digital I/O Panel	DSP Introduction chapter 6
Status Bits Panel	DSP Introduction chapter 6

Displays the Analog Anlayzer Panel.

**Page 1** Page 2 Page 3 Page 4 Page 5

Figure 3-1 Status Bar and Page Tabs

Selective Analysis of Analog Signals	Narrow Bandpass Filter chapter 7
Digital Domain Signal Analyzer	Digital Domain Tester chapter 8
Digital Data Analyzer	Digital Data Analyzer chapter 9
General Purpose Spectrum Analyzers	Spectrum Analyzers chapter 11
Multitone Analyzers	Multitone Analyzers chapter 12
Quasi-Anechoic Analyzer	Quasi-Anechoic Acoustical Tester (MLS) chapter 13
Sweep	Sweep chapter, page 15-1
Graph and Bar Graphs	Graph and Printing chapter, page 16-1
Sweep Settling	Sweep chapter, page 15-23
DCX	DCX-127 chapter, page 23-1
Switcher	Switchers chapter, page 20-1
Data Editor	Limits chapter, page 17-1

All panels not listed above are described in this chapter.

## Procedure Editor

The Procedure Editor can be brought to the screen by:

clicking on the Procedure Editor icon  on the Panels Toolbar

or

selecting Panels Procedure Editor from the menu bar

or

from the keyboard, pressing    (for Panels Procedure)

or

from the keyboard, pressing    (for pRocedure shOw panel)

Procedure files (.APB file type) are APWIN Basic files which define and control automatic processes. Procedure files can load test files from disk and can also directly control the settings of all instruments. Creation and operation of procedures and details of the APWIN Basic

language are covered in the APWIN Basic Programmers Manual and in the Help menu under Help APWIN Basic, Help APWIN Basic Extensions, and Help APWIN Basic Editor.

## Diagnostic Panel

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The diagnostic panel is used for instrument service, diagnosis and calibration at Audio Precision and Audio Precision authorized Service Centers. It is not intended for use in normal operation.

## Regulation Function

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Regulation is a software servo-mechanism which forces a measured parameter to a user-specified target value by automatically adjusting a user-specified setting parameter. Examples include automatically adjusting the generator frequency until measured response is exactly 3.0 dB below the midband reference level or automatically adjusting the generator amplitude until measured distortion at the output of a power amplifier is exactly 1.0%. Regulation can be manually triggered while viewing instrument panels, or may be automatically invoked at each step of a Source-1 sweep.

Regulation setup and manual triggering of regulation cycles is accomplished with the Regulation panel. This is displayed from the Menu command Panels Regulation.

The Regulation field permits selection of the measurement which will be brought to a target value (regulated) by the function. Clicking the ellipsis (. . .) button at the right of this field displays a browser which may be used to select any measurement parameter of the instrument. The next field to the right (labeled “To”) permits entry of the target value to which the measurement is to be forced. The down arrow at the right of this field allows selections of all available units for the parameter being regulated. The field at the end of this row provides entry capability for the tolerance range around the target value that is deemed acceptable. Tolerance may be set in any of three fashions—%, dB, or Absolute (Abs) as selected by the “radio buttons” above this field. With % selected, the measurement must be within the entered numeric percentage of the target for regulation to be judged complete. For example, with a 1.00 Volt target and a 5% tolerance, regulation is judged complete when the reading falls within the band of

0.95 Volts to 1.05 Volts ( $1.00 \pm 5\%$ ). With the dB mode selected, the reading must be brought within the specified decibel deviation. With a 1.00 Volt target (0.0 dBV) and a 0.5 dB tolerance, successful regulation would be a result between -0.5 dBV and +0.5 dBV (0.944 Volts to 1.059 Volts). With the Abs mode, the target must be within the specified variation from the target in absolute units. For example, if the regulated parameter is THD+N in % units and the target is 1.0% THD, an Abs tolerance of 0.5% would cause any measurement between 0.5% and 1.5% ( $1.0\% \pm 0.5\%$ ) to be accepted.

The “by varying” field is for selection of the setting or controlling parameter which will be automatically varied in an effort to reach the target measurement described above. Care must be taken to select a “by varying” parameter which controls or influences the measurement. For example, one would not normally expect to affect the interchannel Phase measurement of a two-channel device by varying the generator amplitude driving that device, but you could expect to affect the distortion of a power amplifier by varying either amplitude or frequency of the generator driving it and may affect the distortion of a D/A converter by varying the amplitude or frequency of injected jitter on the digital output from System Two. The ellipsis button at the right of the field permits selecting any setting in the instrument. The High Bound and Low Bound fields at the right permit setting upper and lower limits beyond which the Regulation function will not attempt to set the “by varying” parameter. Any available unit for these Bounds fields may be selected via the down arrow at the right.

The “Operation” field selects the mathematical algorithm which

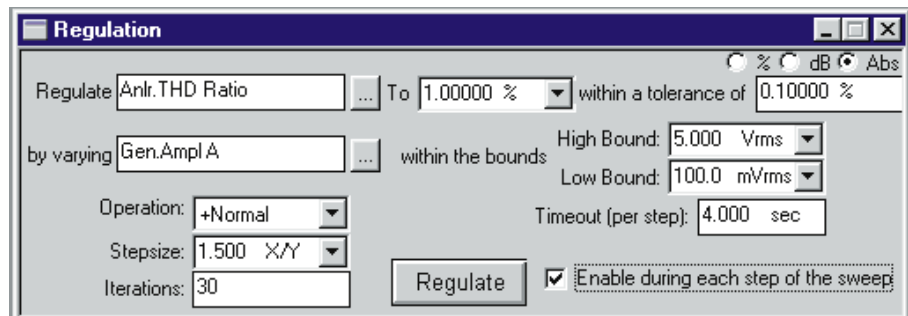


Figure 3-2 Regulation Panel, Typical Setup for Power Bandwidth Sweep

relates the “by varying” parameter and the “Regulate” parameter. There are five choices: Linear, +Normal, -Normal, Maximum, and Minimum.

**Linear** initially assumes that any change in the “by varying” parameter should cause the exact same change in the measured “Regulate” parameter. This is the normal relationship between input and output signal level of an amplifier over its normal operating range. If the initial measured output level from a device is 4.96 dB below the “To” (target) value, the generator amplitude selected in the “by varying” field is increased by 4.96 dB. If this single change in “by varying” value does not bring the “Regulate” value to the target, additional attempts will be made until the target is finally achieved. The “Stepsize” field is gray when Linear is selected, since the actual step change in the “by varying” parameter will be exactly the amount that the measurement of the “Regulate” parameter differs from the target “To” value. The Linear Operation is the fastest and best selection to use when setting device outputs to a reference level or making frequency response measurements at constant output level or constant modulation percentage.

The “**+Normal**” selection assumes that the “By varying” parameter and the “Regulate” parameter are proportional, but not necessarily in a linear fashion. This is typical of the relationship between generator amplitude and measured distortion of an amplifier near the maximum power point, or between generator frequency and measured amplitude from a high-pass filter on its attenuation skirt. In “+Normal” operation, the “by varying” parameter starts from its present setting. If the measured “Regulate” value is below the target, the “by varying” parameter is increased by the “Stepsize” amount and the “Regulate” parameter is measured again. When the first increase by the “Stepsize” amount causes the measured “Regulate” parameter to go above the target value, the “by varying” parameter will then be *decreased* by half the “Stepsize” amount. Each time the “Regulate” parameter measurement crosses through the target, the direction of change of the “by varying” parameter is reversed and the step size is again cut in half. Successful regulation is finally achieved when Regulate is brought within the specified tolerance of the target. The



Regulation process will halt earlier if the specified number of “Iterations” is reached.

The “**-Normal**” algorithm assumes an inverse proportional relationship between “by varying” and “Regulate”. This is typical of the relationship between generator frequency and low-pass filter output amplitude on the attenuation skirt, or between generator amplitude and measured THD+N in relative units (dB or %) in the lower, noise-limited area of operation of a device. “-Normal” works exactly like “+Normal” except that the direction of change of the “by varying” parameter is reversed.

The “**Maximum**” algorithm controls the “By varying” parameter so as to locate a peak response point. The “by varying” parameter is always started from the present value and is increased in “Stepsize” amounts as long as the “Regulate” parameter is increasing. When the “Regulate” parameter makes a decrease, the direction of change is reversed and the step size is cut in half. The “Maximum” operation continues until the “by varying” parameter has been reversed in direction the number of times entered in the “Iterations” field. The “**Minimum**” operation is conceptually similar but expects the “Regulate” parameter to decrease initially and reverses direction each time the “Regulate” parameter goes through a null and starts increasing.

The input information to the Regulation process is a stream of settled readings from an instrument, processed through the Sweep Settling algorithm. If the variability of readings from the instrument is larger than the values set for that parameter on the Sweep Settling panel, the Settling algorithm may not be able to obtain a settled value to pass on to the Regulation algorithm. The Timeout field on the Regulation panel sets a time interval beyond which the Regulation process will not wait for a settled input. Anytime that the Settling algorithm cannot deliver a properly-settled reading to the Regulation process within the Timeout period, the most recent stream of up to 32 unsettled readings from the instrument will be averaged and used as input to Regulation.

A Regulation cycle may be manually triggered by clicking the “Regulate” button on the Regulation panel. If the “Enable during




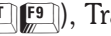



every step of the sweep” checkbox is checked, a Regulation cycle will automatically be triggered at each step, including the Start value, of a Source-1 sweep. The Sweep panel will set the Source 1 parameter to the Start value, a complete Regulation cycle will take place until either success or an exit from Regulation with an Unregulated message, Source 1 steps to the next value, Regulation is triggered again, etc. During a Regulation cycle, whether manually trigger or as part of a sweep, the panel fields displaying the “Regulate” measurement parameter and the “by varying” settings parameter will change to a yellow background.

When making a sweep test with Regulation enabled, it is not typically useful to plot the parameter being regulated, since it will plot as an essentially-flat horizontal line. Plotting the parameter may be desirable during setup to be sure that Regulation is working properly. Here are several examples of Regulated sweeps:



- Using Regulation to hold a power amplifier output distortion constant (perhaps at 1.0%) by varying generator amplitude, but plotting amplifier output Level in Watts versus generator frequency to produce a graph of power bandwidth.
- Using Regulation to find a minus three dB point on a device, then making a single-point “sweep” to display the measured frequency of that point.
- Using Regulation to hold the modulation percentage (deviation) of a preemphasized broadcast transmitter constant (BNC connector fed from precision demodulator composite or baseband output) while measuring and plotting THD+N versus frequency (XLR connectors fed from de-emphasized outputs of demodulator).
- Using Regulation to hold the modulation percentage (deviation) of a preemphasized broadcast transmitter constant (BNC connector fed from precision demodulator composite or baseband output) while measuring and plotting the generator amplitude vs. frequency required to obtain that constant modulation value. To plot generator amplitude or any other

setting value requires checking the “Show Settings” box in the Data 1 through Data 6 browser on the Sweep Panel. This curve is essentially a modulation sensitivity test. Plotting the generator amplitude using normal units will result in a curve that is inverted from the actual preemphasis curve shape of the transmitter. Less generator amplitude is required at high frequencies where the preemphasis curve rises, producing higher modulation sensitivity. Therefore, the dBrInv unit (relative decibels, inverted) may be selected. The dBrInv unit exchanges plus and minus signs to rotate dBr values around the dBr zero value and will produce a preemphasis curve of the expected shape.



## Sweep Menu

The Sweep Menu consists of the following commands: Show Panel, Show Settling Panel, Start () , Stop () , Start with Append () , Start with Repeat () , Transform Data without Acquire () , Reprocess Data () , Compare Data to Limits () , Copy Data 1 Settings to Data 2 Settings, Copy Data 2 Settings to Data 1 Settings.

### Show Panel



The Show Panel selection on the Sweep menu causes the Sweep panel to be displayed. This menu command is equivalent to clicking on the Sweep icon , selecting Sweep on the Panels menu, or using the  keystrokes from the keyboard.

### Show Settling Panel

The Show Settling Panel selection causes the Sweep Settling panel to be displayed. This is equivalent to clicking on the Sweep Settling icon , selecting Sweep Settling on the Panels menu, or using the  (for Panels settLing) keystrokes.



## Start

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The Start selection on the Sweep menu starts a new sweep test according to the definitions presently set up on the Sweep panel. This menu command is equivalent to pressing the  function key, clicking on the Go button at the bottom of the Sweep panel, or clicking on the icon with the green “light”  on the Standard Toolbar.






## Stop

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The Stop selection on the Sweep menu stops any sweep currently in progress. This menu command is equivalent to pressing the  key, clicking on the button at the bottom of the Sweep panel while the button label is “Stop”, or clicking on the icon with the red “light”  on the Standard Toolbar.

## Start with Append




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The Start with Append selection on the Sweep menu starts a new sweep test and retains any sweep data already in computer memory. This menu command is equivalent to the   keystroke or to checking the “Append” box on the Sweep panel, then pressing  or clicking on the Go button or green icon. Note that the “Append” checkbox is not automatically unchecked at the end of a sweep initiated with the Start with Append menu command or the   keystrokes. Future sweeps made with the current setup will continue to be appended to all accumulated sweep data unless the user specifically unchecks the Append checkbox.

Repeating sweeps and appended sweeps may be combined to produce a continuously-repeating sweep that accumulates all the sweep data, by checking both the Append and Repeat checkboxes.

## Start with Repeat


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The Start with Repeat selection on the Sweep menu starts a new sweep test which will continue with repeated sweeps indefinitely until manually halted with the  key or Stop icon. This menu command is equivalent to the   keystroke or to checking the “Repeat” box


on the Sweep panel, then pressing  or clicking on the Go button or green icon. Note that the “Repeat” checkbox is not automatically unchecked when a repeating sweep is interrupted with the  key or Stop button or icon. Future sweeps started by any method ( key, Go button, green light icon, etc.) will repeat continuously unless the user specifically unchecks the Repeat checkbox.

Repeating sweeps and appended sweeps may be combined to produce a continuously-repeating sweep that accumulates all the sweep data, by checking both the Append and Repeat checkboxes.

## Transform Data without Acquire


FFT-based (batch mode) DSP programs have three distinct, sequential phases to their operation when the  key is pressed or Go button or green light icon is clicked. First, data is accumulated into the acquisition buffer until the buffer is filled to the specified acquisition length. Second, a Fast Fourier Transform (FFT) is performed to obtain amplitude (and sometimes phase) versus frequency data which is stored in a different memory buffer from the acquired signal (amplitude versus time). Third, a post-processed version of the amplitude versus time or amplitude versus frequency data (depending upon Sweep panel settings at Source 1 and Data 1 or 2) is transmitted from the DSP module in the test system to the computer for graphing by APWIN software.

It is often desirable to re-transform the data already present in the acquisition buffer, rather than acquiring new data. This will be the case when a previously-acquired waveform is downloaded from computer disk to the DSP module for further analysis. It will also be the case if it is desired to re-transform the existing contents of the acquisition buffer while using a different window function.


The Transform Data without Acquire command on the Sweep menu, or the equivalent  keystrokes, cause the second and third phases described above to take place (FFT transformation and transmission of results to computer). See the FFT functional description in the FFT Overview chapter.

## Reprocess Data

---


FFT-based (batch mode) DSP programs have three distinct, sequential phases to their operation when the  key is pressed or Go button or green light icon is clicked. First, data is accumulated into the acquisition buffer until the buffer is filled to the specified acquisition length. Second, a Fast Fourier Transform (FFT) is performed to obtain amplitude (and sometimes phase) versus frequency data which is stored in a different memory buffer from the acquired signal (amplitude versus time). Third, a post-processed version of the amplitude versus time or amplitude versus frequency data (depending upon Sweep panel settings at Source 1 and Data 1 or 2) is transmitted from the DSP module in the test system to the computer for graphing by APWIN software.

It is sometimes desirable to perform a new post-processing of the FFT results and transmission of the post-processed data to the computer for display, without acquiring new data and without re-transforming the data already present in the acquisition buffer. This is faster than re-transforming followed by new post-processing. It is automatically done (unless turned off in the Utilities Configuration menu) when zooming in on a section of an FFT display and wishing to obtain optimum resolution of the new sub-set of the original data, or when changing multitone post-processing methods between the normal (response and level), distortion, or noise functions.

The Reprocess command on the Sweep menu, or the equivalent  keystrokes, cause the third phase described above to take place (post-processing of FFT results and transmission of results to computer). See the FFT functional description in the FFT overview chapter.

## Compare Data to Limits

---

The Compare Data to Limits command in the Sweep menu causes any sweep results data presently in computer memory to be re-graphed and compared to limits if limits files (.ADL) are “connected to” the test via the Limits button for each Data variable on the Sweep panel. This menu command is equivalent to the  keystroke.

## Copy Data 1 Settings to Data 2 Settings and Vice-Versa

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The “Copy Data 1 Settings to Data 2 Settings”, and “Copy Data 2 Settings to Data 1 Settings” assist Sweep panel setup for certain types of two-channel and stereo device testing. If Data 2 is set to None when the “Copy Data 1 Settings to Data 2 Settings” command is selected, an exact copy of every Data 1 field on the Sweep panel is written into the corresponding Data 2 field. If Data 2 has already been selected to an Instrument and Parameter before the command is selected, and if the Data 1 and Data 2 parameters are compatible (both amplitude values or both frequency values, for example), the Data 2 Instrument and Parameter will be left as they were but the Data 1 units, graph top and bottom values, log/lin selection, and divisions settings will all be copied into the corresponding Data 2 fields. If Data 2 has already been selected and the Data 1 and Data 2 parameters are not compatible (dBV at one and Hz or degrees at the other, for example), an error message is displayed and no changes are made. The “Copy Data 2 Settings to Data 1 Settings” works in exactly the same manner but from Data 2 to Data 1.

Note that the Stereo Sweep checkbox at the bottom of the Sweep panel is also an aid in setting up two-channel and stereo device tests, with each technique having its own advantages and disadvantages. The Stereo Sweep checkbox automatically writes Data 1 settings into Data 3 (and Data 2 settings into Data 4). Data 3 and Data 4 do not have their own selectable units since the graph has only two vertical axes (left and right) for calibration. In the Stereo Sweep mode, Data 3 plots onto the Data 1 axis using Data 1 units.

## Compute Menu

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The Compute Menu contains a number of powerful mathematical computations that may be applied to the data set in memory following a sweep. The Compute Menu capabilities are described in the Compute chapter.

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## Procedure Menu

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The Procedure Menu consists of the following commands: Show Panel, Open Uses, Learn Mode, Close, Close All, Save All, Run, Pause, End, Step Into, Step Over, Step Out, Step to Cursor, Toggle Breakpoint, Quick Watch, Add Watch, Browse, Set Next Statement, Show Next Statement, and a list of the most recently used Procedures. The functions and operations of these commands is described in detail in the APWIN Basic User's Guide and Programmer's Reference.

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## Utilities Menu

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The following commands make up the Utilities menu: Restore hardware, Hardware status, Configuration, Filters, Turn all outputs off (**F12**), Turn outputs back on (**CTRL F12**), Clear log file, View log file, Learn Mode.

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### Restore Hardware

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If APWIN software is started with the measurement instrument not connected to the Audio Precision Interface Bus (APIB), or connected but power not turned on, the Utilities Restore Hardware menu command must be used to set the instrument to the present settings of the software. Utilities Restore Hardware may also sometimes be effectively used as a general reset when the instrument does not appear to be responding normally.

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### Hardware Status

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The Utilities Hardware Status menu command displays key information (see Figure 3-3) about hardware found connected to the Audio Precision Interface Bus (APIB) by APWIN software. This includes the type and address of the interface card and detection of the presence of major instrument modules.

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### Utilities Configuration

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The Utilities Configuration command permits custom-tailoring of a number of APWIN functions to the user's preference, described in the following sections. When any changes have been made to the Utilities Configuration dialog box (see Figure 3-4), click the OK button if you



wish the changes to become effective. Click the Cancel button if you do not wish the changes to take effect.

## Prompt to Save Test

When the “Prompt to Save Test when a test is closed” checkbox on the Utilities Configuration dialog is checked, APWIN will prompt the user to save the test currently in memory whenever an action is initiated which would destroy the test in memory. Examples of actions which destroy the test in memory include clicking on the new test icon



or selecting the File Open Test or Open Data or File New Test menu commands. If the box is not checked, no reminder is given that the test presently in memory will be replaced by the action initiated.

## Last test save and load

The Utilities Configuration dialog box contains two check boxes to manage LAST.AT1 or LAST.AT2 activities.

Load LAST.AT1 or LAST.AT2 on startup: when this box is checked, APWIN will automatically load a test named LAST.AT1 or LAST.AT2 located in the “home” directory. The “home” directory is the directory specified by the user or is automatically selected as C:\APWIN by the installation procedure when APWIN is initially installed.

Save LAST.AT1 or LAST.AT2 on exit: when this box is checked, APWIN automatically saves as LAST.AT1 or LAST.AT2, in the home directory, the test in memory at the time that APWIN is closed.

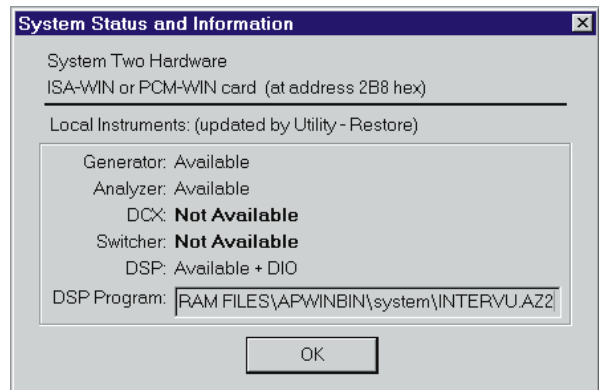


Figure 3-3 Utilities Hardware Status Information

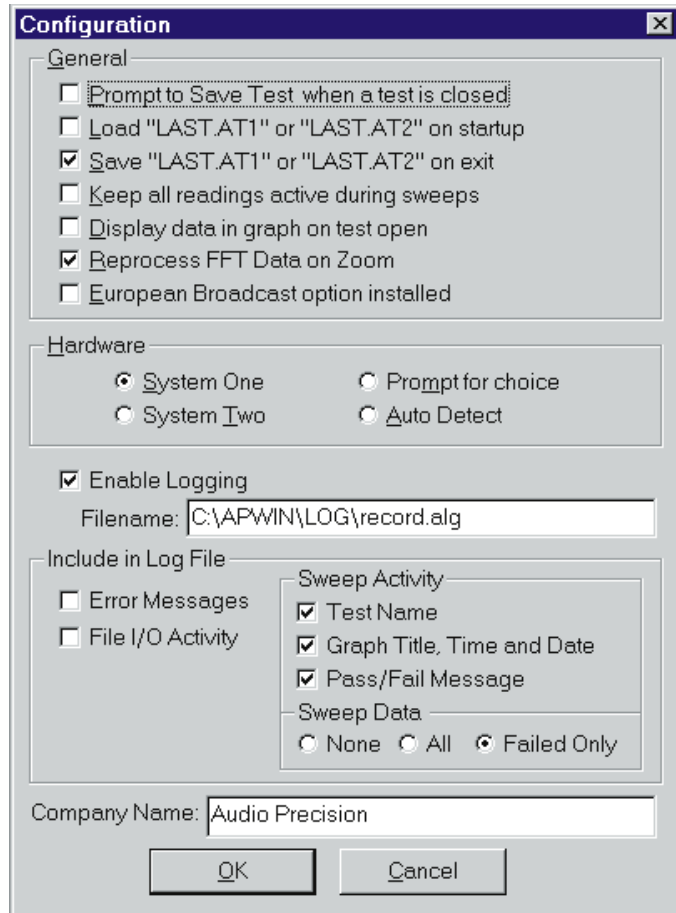



Figure 3-4 Utilities Configuration Dialog Box

## Readings active during sweeps

When the “Keep all readings active during sweeps” checkbox on the Utilities Configuration dialog is checked, all meters on all panels will actively update their readings even during sweeps. If this box is not checked, only the meters whose measurements are selected for plotting at Data 1 through 6 or Source 1 through 2 will be active during a sweep, with all other meters “frozen” during the sweep. Keeping the readings active during a sweep permits other parameters to be observed while a sweep is in progress, but it will slow down the sweep itself since the computer processor must service those additional meters.

## Graph Data When Test Opened

When the “Display data in graph on test open” box is checked, any data stored with a test will be automatically graphed (and compared to limits, if any) when the test is opened. If the box is not checked, data loads into data memory but will not be graphed until the  key is pressed or the Sweep Compare Data to Limits menu command is issued.

## European Broadcast Impedances

A hardware option is available for System One which changes output termination (source) resistance values in the analog generator and input termination values on the analog analyzer. The European option analog generator balanced output impedances are 40, 200, or 600 Ohms instead of the standard 40, 150, or 600 Ohms. The analog analyzer European Broadcast option input termination selections are 300, 600, or 100,000 Ohms instead of the standard 150, 600, or 100k. In order for APWIN software to properly show these selections on the panel and use the correct generator source value in calculating output power with generator dBm and Watts units, the “European Broadcast Option Installed” checkbox must be checked. Do not check this box if the hardware option has not been installed.


## Instrument Model

The Utilities Configuration dialog box contains a “Hardware” section with four choices: System One, System Two, Prompt for choice, and Auto detect. One and only one of these four “radio buttons” may be selected.

- If **System One** is selected, APWIN will always configure itself for System One hardware when started.
- If **System Two** is selected, APWIN configures itself for System Two hardware when started.
- If **Prompt for Choice** is selected, APWIN will display a dialog box at start-up and wait for the user to select the System One or System Two button.

- If **Auto Select** is selected, APWIN looks for coding unique to System Two modules and configures itself for System Two if they are found, otherwise configures itself for System One. The System One or System Two instrument must have power applied at the time APWIN is started for this selection to be effective; if no instrument is detected (which can be due to power not being applied), the “Prompt for Choice” dialog will be displayed.

## Re-Process FFT Data on Zoom

When this box is checked, each graph zoom in action (by dragging the mouse) or Zoomout or Zoomout to Original action (right mouse button menu on graph) will be accompanied by a “Sweep Re-Process Data” () command. If the acquired signal is still in DSP memory, this assures that the resolution is always the maximum supported by either the number of points graphed or the FFT transform length. See the FFT functional description in the FFT Overview chapter.

## Log Files

The Log File (.ALG file type) can serve as a record of many important APWIN operations such as limits failures, test names, date and time each test ran, file loading and saving activity, and error messages. The Log File can be viewed on screen or printed to paper.

The Utilities Configuration menu command displays a dialog box (see Figure 3-4) with controls to allow the user to establish a file name and directory and to define the contents of the Log File. Unless manually cleared by the Utilities Clear Log File menu command, all new information (of the categories defined by the user) is added to the bottom of the Log File so that it becomes a growing record of the specified activities of the software. Thus, a Log File can become a complete record of an entire audio testing procedure.

## Establishing the Log File Name

Type the name to be created for a Log File into the Filename box. The .ALG extension will be automatically furnished. If no information other than the file name is furnished, the log file will be saved in the

“Log” subdirectory under the “Home” directory. With the default installation procedure, the log file will thus be saved into the C:\APWIN\LOG directory. If the user supplies an explicit path with the log file name, it will be saved into the specific directory specified. If the user types “.\filename”, the log file will be saved into the Current Working Directory. See the discussion of Current Working Directory in the File Menus section on page 3-10.

## Turning Logging On and Off

The “Log Errors” checkbox controls whether logging actually takes place. With the box checked, information of the types specified by the checkboxes and radio buttons below will be logged into the file name established. If the box is unchecked, no logging takes place.

## Information to Include in Log File

The Log File can contain a number of different types of information, at the user’s option. The selectable information types include:

**Error messages:** when this box is checked, any APWIN or Windows error messages which occur during the period that logging is turned on will be written into the Log File.

**File I/O Activity:** when this box is checked, a text message will be written into the Log File for every disk file opened or every file saved to disk. The message includes the name and full path name of the file and the date and time at which it was opened or saved.

The Log Sweep Activity section permits inclusion or exclusion from the Log File of several pieces of information relating to the running of tests. The selections are:

- **Test Name:** the name, including path name, of the test that was executed
- **Graph Title, Time and Date:** the Graph Title and Time and Date at which the test was executed, exactly as they are displayed in the title bar of the graph.

- **Pass/Fail Message:** when this box is checked, an error summary message will be written into the Log File each time a test is run. The first word of the message will be PASSED or FAILED. Following a colon (:), the error message will include the number of measurements which were below the lower limit, the number of measurements which were above the upper limit, and the number of Timeouts which occurred. If the box is not checked, no error summary message is written into the file.
- **Log Sweep Data; None, All, Failed Only:** these three radio buttons permit control of whether no test point values (None), all test point values (All), or only those test points which were outside limits (Failed Only) are written into the Log File. Any values written into the Log File which were outside limits will have parenthesis at the end with the (less than) or (greater than) symbol and the value of the limit which they failed.

### Example Log File

The Log File is an ASCII file that can be viewed in or printed from almost any text editor, such as Windows Notepad. An example is shown below:

```
C:\Program Files\APWIN\SAMPLES\sample.at1
Sample Response Test w/Limits 07/09/95 11:10:13
FAILED : Lower Limit 3, Upper Limit 4, Timeouts 0
Gen.Freq(Hz)      Anlr.Level A(dBr)
20000             5.000000(>2)
10022.5           4.200000(>2)
5025              2.100000(>1)
2517.5            1.800000(>1)
79.625            -1.800000(<.5)
39.9              -2.100000(<.5)
20                -2.700000(<.5)
```

The first line is the test file name and path. The second line is the Graph Title, date, and time. The third line is the “Pass/Fail Message” which summarizes failures. The “Failed Only” option was in use, so the remaining lines consist of the points which were outside limits. The

points below 2517.5 Hz and above 79.625 Hz were within limits and thus are not shown. The “greater than” (<) and “less than” (>) symbols and numbers at the end of each row show where the value was above the upper limit or below the lower limit, and the value of the limit itself.


## Company Name

---

The Company Name box permits entry of a name (or other information) up to 32 characters that will appear in the Title Bar of the Graph. The standard Company Name is Audio Precision, but it may be replaced with text of the user’s choice. Note that even though a Company Name is in use, the user has the choice via the Title and Labels dialog box obtained by a right mouse click on the graph of whether it displays on screen in the Graph Title Bar, and via the Page Setup screen of whether the Company Name prints as part of a graph hard copy.



## Utilities Filters


---

The Utilities Filters menu command permits proper labeling of filter types plugged into each of the five sockets (slots). Once the filter-to-slot relationship has been defined, filters may then be selected by name (rather than only by slot number) in the Filters list box on the Analog Analyzer panel. To identify a filter after plugging it into any of the five slots, designate the filter name by clicking on the down arrow at the right end of the box on the Utilities Filters dialog to display a list of names. Use the scroll bars if necessary to locate the particular filter type, and click on the filter name (or use up and down arrows and the  key). Click on the OK button to write the selections to the analyzer panel.



## Turn All Outputs Off

---

The Utilities Turn All Outputs Off command, or the  function key, turns off the outputs of all analog and digital generators. It is thus equivalent to the  function key of the earlier S1.EXE DOS software. The intent of this function is for a “panic button” or emergency button if it is determined that the device under test is potentially being damaged or people are being disturbed by the application of signal. All outputs which are turned off by this command may be turned back

on by the Utilities Turn Outputs Back On command or the  function key.

### Turn Outputs Back On

This menu command and the  keystroke alternative serves as the “restore” for the Utilities Turn All Outputs Off command. It will turn back on all outputs which were turned off by an  function key operation or the Utilities Turn All Outputs Off command, but will not turn on any outputs which were not on when that command was issued.

### Clear Log File

The Utilities Clear Log File command erases all contents from the log file. It does not change any selections for what will be written into the new log file when the next test is run.

### View Log File

The Utilities View Log File launches the Windows Notepad editor with the Log File loaded. Windows Notepad then supports printing the Log File, saving it to another directory or under another name, etc.

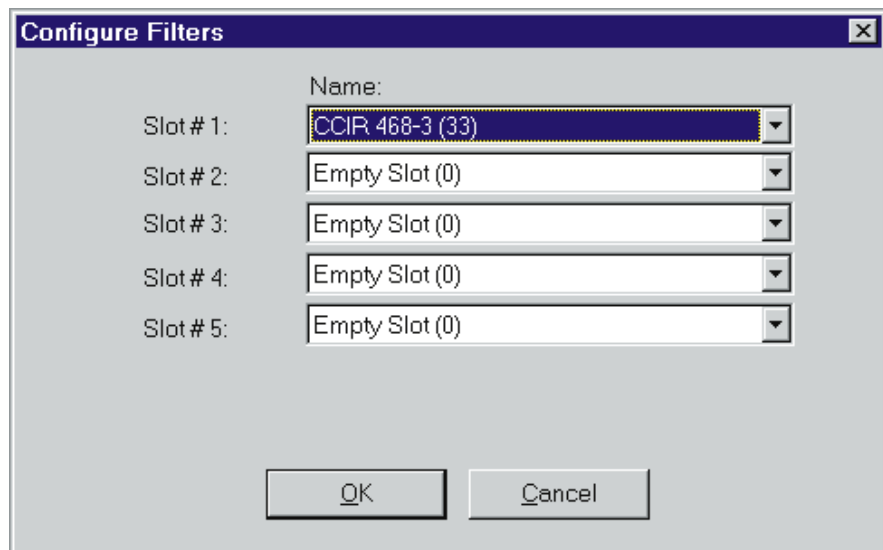


Figure 3-5 Utilities Filters Dialog Box



## Learn Mode

---

The Utilities Learn Mode command is identical to Procedure Learn Mode, or clicking on the first icon on the Learn Mode toolbar. It starts the process of creating a procedure (macro) by writing code into the Procedure Editor for each mouse click or keyboard action made by the user. Learn Mode is halted by executing the Utilities Learn or Procedure Learn command again.

## Window Menu

---

The Window Menu consists of these commands: Tile, Arrange Icons, Close All.


### Tile

---

The Windows Tile command arranges all open panels (windows) on the current page to be visible and non-overlapping if possible. If there is insufficient open space, the Tile command may place windows on top of other windows.

### Arrange Icons

---

The Window Arrange Icons command organizes all minimized panel icons at the bottom of the workspace so that they are visible. Panels may be minimized in Windows 3.11 by clicking on the down arrow icon (second icon from the right at top right corner). In Windows 95, panels are minimized by clicking on the  icon near the upper right corner of a panel or window.

### Close All

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The Windows Close All command closes all windows (panels) on the current page. It is thus equivalent in Windows 95 to clicking on the icon at the upper right of each individual panel on the current page, or in Windows 3.11 to clicking on the icon at the upper left corner of each panel and selecting Close. Closed panels still exercise the same control settings over the hardware or software functions that they did when visible.

## Help Menu

The Help Menu consists of the following topics: Index, Using Help, APWIN Basic Language, APWIN Basic Extensions, APWIN Basic Editor, About APWIN, Release Notes.

Context-sensitive Help is implemented for most panel fields of APWIN. Select the panel control you desire information on, using the mouse cursor or **TAB** and **SHIFT TAB** keys, and press **F1**. If context-sensitive Help is not available for a specific panel control selected, the Contents topic of Help will be displayed when the **F1** key is pressed.

### Index

The Help Index command launches the Windows Help Viewer with the APWIN Help file loaded. The Contents section, equivalent to a Table of Contents in a printed book, will be initially displayed. The user can click on the green underlined text of a listed topic to jump directly to that topic. Alternately, the Search button may be clicked to bring up a key word location feature similar to the index in a printed book.

### Using Help

The Help Menu consists of the following topics: Index, Using Help, APWIN Basic Language, APWIN Basic Extensions, APWIN Basic Editor, About APWIN, Release Notes.

Context-sensitive Help is implemented for most panel fields of APWIN. Select the panel control you desire information on, using the mouse cursor or **TAB** and **SHIFT TAB** keys, and press **F1**. If context-sensitive Help is not available for a specific panel control selected, the Contents topic of Help will be displayed when the **F1** key is pressed.

### Index

The Help Index command launches the Windows Help Viewer with the APWIN Help file loaded. The Contents section, equivalent to a Table of Contents in a printed book, will be initially displayed. The

user can click on the green underlined text of a listed topic to jump directly to that topic. Alternately, the Search button may be clicked to bring up a key word location feature similar to the index in a printed book.

## Using Help

---

The Help Using Help command launches the Windows Help Viewer with the Help file loaded that describes how to use the Windows Help feature. It describes how to navigate among Help topics, how to annotate Help topics with your own personal notes, how to create “bookmarks” to make it easy to find frequently-used topics, how to copy a Help topic to the Windows clipboard or to print a Help topic, how to display the history of your navigation through Help topics, and provides specific information on each command of the Help Viewer. Under Windows 95, the Help on Help explains how to use new features such as changing font size and performing searches for individual words or phrases.

## APWIN Basic Language

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The APWIN Basic Language is compatible with Microsoft Visual Basic for Applications. It is a full-featured programming language with special features for creation of user dialogs containing various controls. Basic Language Help describes the general programming features of this language. See the separate APWIN Basic User’s Guide and Programmers Reference.

## APWIN Basic Extensions

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A large number of Audio Precision specific extensions have been added to the general programming features of the language. These extensions provide complete control over every stimulus and measurement function, sweeps, graphs, etc. Basic Extensions Help describes these specific additions for the instrument. See the separate APWIN Basic User’s Guide and Programmers Reference.

## APWIN Basic Editor

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The APWIN Basic Editor (Procedure Editor) is provided to create and modify procedures. Features of this editor are described in Basic

Editor Help. See the separate APWIN Basic User's Guide and Programmers Reference.

### **About APWIN**

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The Help About APWIN menu command displays information on the version numbers of key elements of APWIN software. This information may be requested by Audio Precision Applications Engineers if you call for technical support.

### **Release Notes**

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The Help Release Notes menu command launches the Windows Notepad editor with the APREADME.TXT file loaded. This file contains last-minute information on the particular software release and may include information not present in the User's Manual or on-line Help.






# Analog Generator

## APWIN Operation and Panels


To bring the Analog Generator panel to the screen:

Mouse: Click on the Analog Generator icon  on the toolbar or select Panels Analog Generator via the menu bar

Keyboard: Type  or  (Panels analog Generator).

The normal default start-up mode of APWIN software has the Analog Generator panel on page one.

The System One Analog Generator panel may be displayed in two sizes. The smaller version (displayed at start-up) shows only the most important controls—waveform selection, frequency, secondary parameters for certain waveforms, the channel select and Channel B invert controls, output on/mute button, and amplitude control. The larger version shows these same controls plus output configuration controls (impedance, balanced-unbalanced, common mode test mode) and reference values for the dBm, Watts, dBr, and relative frequency units. Clicking on the panel icon on the Panels Toolbar brings up the small version panel.

Holding down  while clicking on the toolbar icon brings up the large version. To switch between the small and large panels:

Mouse, Windows  
3.11 or 95:

Double-click anywhere  
on the control bar.

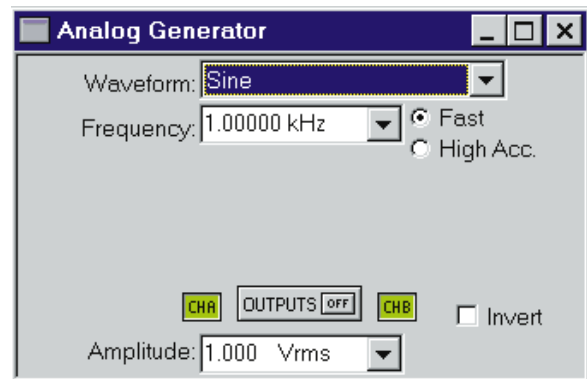




Figure 4-1 System One Analog Generator Panel, Small Version

Mouse, Windows 3.11: Click on the top right corner icon  on the panel, in the control bar.

Mouse, Windows 95: Click on the middle  of the three icons at the top right of the panel , in the control bar.

Keyboard: Press  or .

## Waveform Selection

The waveform field setting determines the type of signal waveform which will be generated. Availability of some waveforms requires that specific hardware modules be present. These include IMD (Intermodulation Distortion) test waveforms (IMD generator option), burst-triggered-gated sine, squarewaves, and white-pink-selective random or pseudorandom noise waveforms (BUR option), and all digitally-generated waveforms (DSP module).

Only the Sine and EQ Sine selections are available if no DSP module or analog generator options are installed. The SMPTE, CCIF, and DIM waveforms can be generated only if the IMD generator option is installed. The Square, Burst, Noise, and Pseudo Noise waveforms require the presence of the BUR

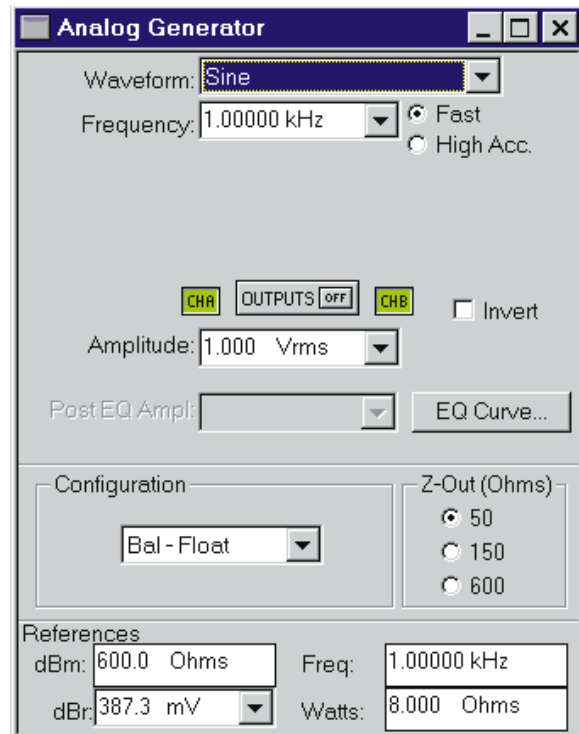


Figure 4-2 System One Analog Generator Control Panel, Large Version



option. The DGen waveforms require the DSP module, present in SYS-222 and SYS-322 models only.

There are interactions between waveform selection, frequency, and amplitude. Squarewave and burst waveforms cannot be generated across the full 10 Hz-204 kHz frequency range of sinewave generation. Maximum available sinewave amplitude at the frequency extremes (below 20 Hz, above 50 kHz) is half that at standard audio frequencies. Maximum amplitude with burst, noise, and square waveforms is half that with sinewaves.

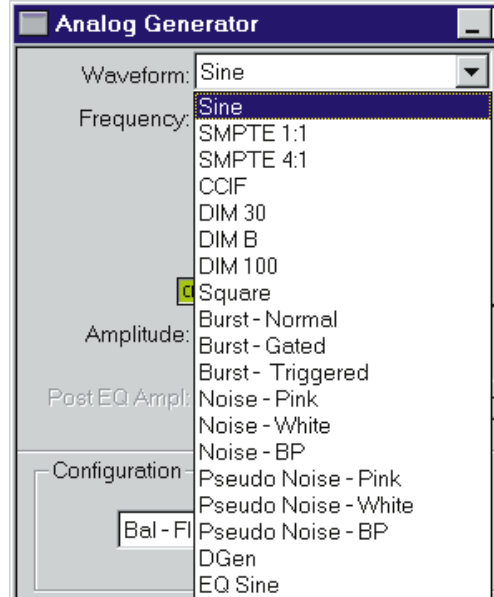


Figure 4-3 Analog Generator Waveform Selection List

## Sine EQ Sine—Generator Equalization

Real-time generator equalization of sinewave amplitude can be applied to the analog generator and the real-time digital generator section of the GENANLR digital-domain DSP program. Equalization works by specifying an amplitude vs. frequency curve in the form of a specific filename and data column within the file. Sine EQ Sine must be selected as the waveform. Equalization

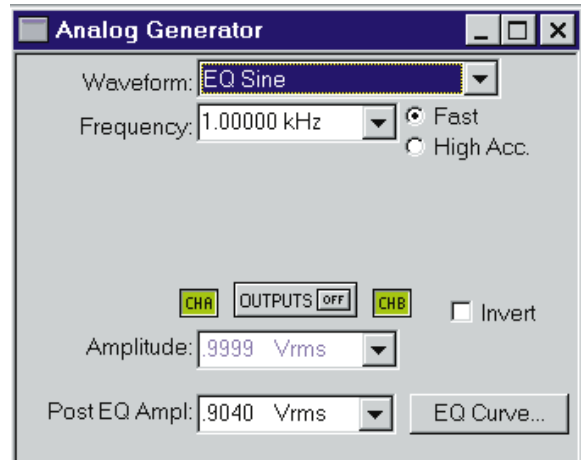


Figure 4-4 EQ Sine Waveform Selected, Post-EQ Amplitude Field Visible

function starts with the user-specified amplitude value, looks up (interpolating when necessary) the correction factor in the equalization file for the generator frequency presently in use, and sends the corrected amplitude command (post-EQ amplitude) to the instrument. The Post-EQ field shows the actual output level from the generator circuitry delivered to the external load, after adjusting the user-requested (pre-EQ) value by the value from the EQ file for the generator frequency currently in use. User entries may be made into either normal (pre-EQ) or post-EQ fields and the other field will be adjusted automatically. If a frequency value outside the frequency range span of the EQ file is entered on the generator panel or occurs during a sweep, the post-EQ amplitude will go to zero.

### Specifying the EQ Curve

Equalization will not occur until a file and column have been specified and the Sine EQ Sine waveform has been selected. The equalization file may be selected before or after selecting the EQ sine

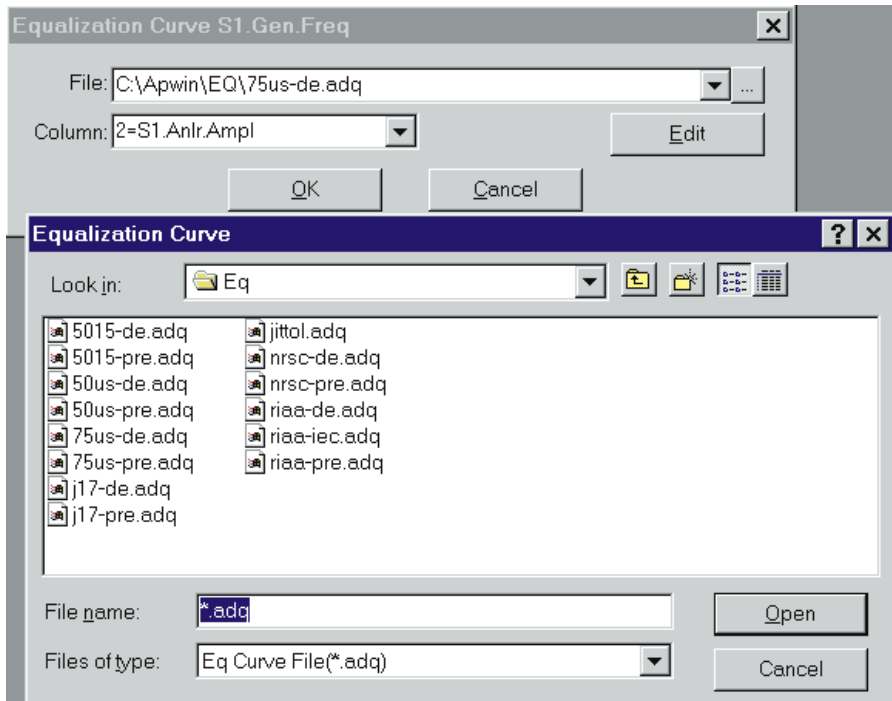


Figure 4-5 EQ Curve Selection Dialog Box and Browser

waveform. It is typically faster to first select the Sine EQ Sine waveform, since that results in a message that no file has been selected and the opportunity (by clicking the Yes button in the dialog) to select one. A browser appears which permits navigating to any desired directory and selecting a .ADQ file name. More than a dozen .ADQ files are furnished with APWIN, all located in the C:\APWIN\EQ directory. It is also possible to create your own custom EQ files as described later. After selecting the file, you must select a column that contains amplitude data. The furnished .ADQ files are all two-column files with frequency in column 1 and amplitude in column 2. The “Edit” button on the browser may be clicked to display, and edit if desired, the EQ curve data in the Data Editor.

If you prefer to select the file and column before selecting EQ sine waveform, click on the “EQ Curve” button on the generator panel that displays the same browser just described. Actual equalization will not take place until the Sine EQ Sine waveform is also selected. When the Sine EQ Sine waveform is selected, the generator amplitude field or fields will display the setting in blue characters rather than the normal black. If Sine EQ Sine waveform is selected without an equalization file and column having been specified, APWIN software considers the equalization factor to be unity at all frequencies, so no equalization takes place.

When the analog generator is swept in Sine EQ Sine waveform with a .ADQ file attached, the generator amplitude is momentarily reduced to zero before either the frequency or amplitude is changed. After both the new frequency and new amplitude commands have been sent to the hardware, the generator output is then brought back to the new level. This technique prevents over-voltage transients that could damage sensitive devices such as loudspeakers or high-power amplitude-modulated transmitters.

## Typical EQ Applications

The most common applications of generator equalization are to produce a nominally-flat output from a device that contains a known equalization function. Typical examples include fm and TV aural broadcast transmitters with 50 and 75 microsecond preemphasis curves, rf signal generators driving broadcast receivers with 50 and 75

microsecond deemphasis characteristics, phonograph preamplifiers with RIAA deemphasis networks, and the recording or playback section (individually) of a tape recorder with its equalization circuits. By connecting an equalization curve to the System One generator which is the inverse of the nominal curve used in the device under test, the resulting measured output should be nominally flat. Small variations from flatness will be a measurement of the accuracy of equalization in the device, since the .ADQ curves furnished are accurate to better than 0.01 dB and the specified generator flatness is of the same order.

Another application of generator EQ plus the Compute utilities is correcting for the small, intrinsic non-flatness of the System One generator and analyzer and the cables which connect the instrument to the device under test. This is particularly important when long cables are used. The process is to first run a frequency response measurement on the instrument with the cables that normally will go to the device under test connected together. Use the Compute Invert utility to turn this measurement upside down. Use Compute Normalize, if necessary, to push the curve up or down so that it goes through zero gain at the frequency which you wish to use as a reference (often 1 kHz). Use the Save As menu command to save this curve as an EQ file (.ADQ). Select this file as the generator EQ Curve and select Sine EQ Sine as the waveform. A response curve should now show almost perfectly flat response, with the generator equalization compensating for the total measured non-flatness of the system (including cables) at each frequency. Most exact equalization will result if the Sweep Settling tolerance for the measured parameter is tightened beyond normal and if the reading rate is forced to 4/sec to get the best possible resolution.

User-made EQ files may be viewed in tabular format and edited by use of the Edit button in the “EQ Curve” dialog, as discussed above. To view an EQ file graphically, use the File Open EQ command. Note that opening an EQ file does not change the Sweep panel settings, so it may be necessary to change Data 1 units and Optimize the graph for the desired display.

## IMD (Intermodulation Distortion) Test Waveforms

The Audio Precision IMD (Intermodulation Distortion) option permits IMD testing to three standards—SMPTE/DIN, CCIF/DFD/Twin tone, and DIM/TIM. The IMD

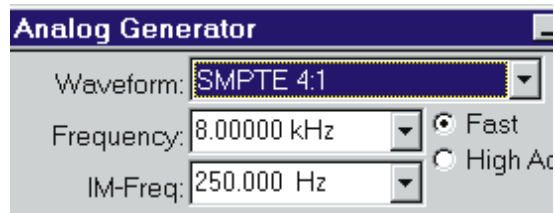


Figure 4-6 IM Frequency Field

option is made up of two circuit boards, an intermodulation distortion test signal generator and an intermodulation distortion analyzer. The generator circuit board generates one of several selected signals which is combined with the sinewave from the main analog oscillator and fed to the output stage. The Generator option is controlled from the Analog Generator panel of APWIN. Clicking the down arrow at the right of the Waveform will display a list of waveforms. On this list, the intermodulation test signal waveform selections are SMPTE 1:1, SMPTE 4:1, CCIF, DIM30, DIM B, and DIM 100. Click the desired signal choice. The waveforms are described below.

Amplitude of IMD test signals is controlled from the Analog Generator Amplitude field. All the IMD waveforms are calibrated to produce the same peak-to-peak signal amplitude that would be produced by a sinewave of the entered value. Thus, the Vp and Vpp units will exactly describe the IMD waveform peak or peak-to-peak amplitude. With all other units, the IMD signal peaks will be 1.414 times the indicated amplitude (peak-to-peak, 2.828 times the indicated amplitude). The measured RMS value of all IMD waveforms will be less than the RMS value of a sinewave of the same peak amplitude since all IMD waveforms have higher crest factors than a sinewave. The applications advantage is that a device under test will not go into clipping when the signal is changed from a sinewave to an IMD signal at the same generator panel Amplitude value, but could clip if the true RMS value were constant between all waveforms.

In the SMPTE waveform mode (which also generates DIN-compatible signals), the IMD generator board generates a sinewave at one of six frequencies—40, 50, 60, 100, 125, 250, or 500 Hz. This low-frequency sinewave is linearly combined in either a 4:1

amplitude ratio (Waveform SMPTE 4:1) or 1:1 amplitude ratio (Waveform SMPTE 1:1) with the variable-frequency sinewave from the main oscillator. For compatibility with the IMD analyzer module, the main oscillator frequency must be 2.5 kHz or higher. The amplitude of the combined IMD test signal is calibrated at the same peak-to-peak value as a sinewave of the value set in the Generator Amplitude field.

In the CCIF (twin-tone, difference tone, DFD) mode, the signal from the main sinewave oscillator is fed to one input of a balanced modulator and the selectable low-frequency signal generated on the IMD generator board is fed to the other input of the balanced modulator. The output from the balanced modulator is a double sideband suppressed carrier signal, where the main oscillator is the suppressed carrier, with equal-amplitude sidebands spaced above and below the suppressed carrier frequency. The IM-Freq field selects the spacing between the upper and lower sidebands. The main generator Frequency field determines the frequency of the suppressed carrier, which is thus the center frequency between the two IMD signals. The amplitude of the combined IMD test signal is calibrated at the same peak-to-peak value as a sinewave of the value set in the Generator Amplitude field.

DIM stands for Dynamic Intermodulation. An alternate frequently-used term is TIM, for Transient Intermodulation. In all DIM waveforms (DIM 30, DIM 100, and DIM B), a squarewave at a frequency near 3 kHz is generated on the IMD generator board and is linearly combined with the sinewave from the main generator. The peak-to-peak amplitude ratio of the squarewave to the sinewave is 4:1. The amplitude of the combined IMD test signal is calibrated at the same peak-to-peak value as a sinewave of the value set in the Generator Amplitude field.

For the DIM 30 and DIM 100 waveforms, the squarewave frequency is 3.15 kHz and the main oscillator must be set to a frequency of 15 kHz in order for analysis to be made by the IMD analyzer board. The squarewave is passed through a single pole 30 kHz low pass filter in the DIM 30 waveform and through a single pole 100 kHz low pass filter in the DIM 100 waveform. For the DIM B waveform (B standing for broadcast), the 30 kHz low pass filter is also used, the squarewave frequency is set to 2.96 kHz, and the main

oscillator sinewave frequency must be set to 14 kHz for proper analysis by the IMD analyzer.

## Squarewave Generator

Selecting Square in the Waveform field will produce squarewaves from the generator output when the BUR-GEN module is installed. The squarewave is calibrated in peak equivalent sinewave terms. Thus, selecting a squarewave with an Amplitude value of 1.000 Vrms on the analog generator panel will produce a 2.828 V peak-to-peak signal. The maximum available amplitude of the squarewave is limited to half the amplitude selectable in Sine waveform. The frequency range for squarewaves is from 20 Hz to 20 kHz. Note that the generator configuration balanced-unbalanced selection must match the external load. Connecting to an unbalanced load from the balanced output configuration will produce distorted squarewaves.

## Sinewave Bursts—Overview

The tone burst capability of the BUR-GEN module provides sinewave bursts which switch between normal, calibrated generator sinewave amplitude and a lower amplitude. Switching between the higher and lower levels always occurs at a positive-going zero crossing, thus there will always be an integer number of complete cycles in the burst. The duration and repetition rate of the burst are controllable, as is the lower amplitude. The main sinewave oscillator produces the sinewave. Thus, the normal frequency and amplitude settings on the analog generator panel determine its frequency and the higher amplitude. In addition to the free-running, repetitive mode of Burst Normal waveform, bursts may be either externally triggered (Burst Triggered) or the signal may be gated on and off (Burst Gated) by an external signal. The tone burst parameters may be swept.

### Burst Control

The burst control area of the Analog Generator panel consists of up to three fields labeled Burst On, Interval, and Low Level. All three of these fields will be visible only when Burst Normal mode is selected in the Waveform field. Only the Burst On and Low Level lines will be displayed when Burst Triggered waveform is selected, and only the

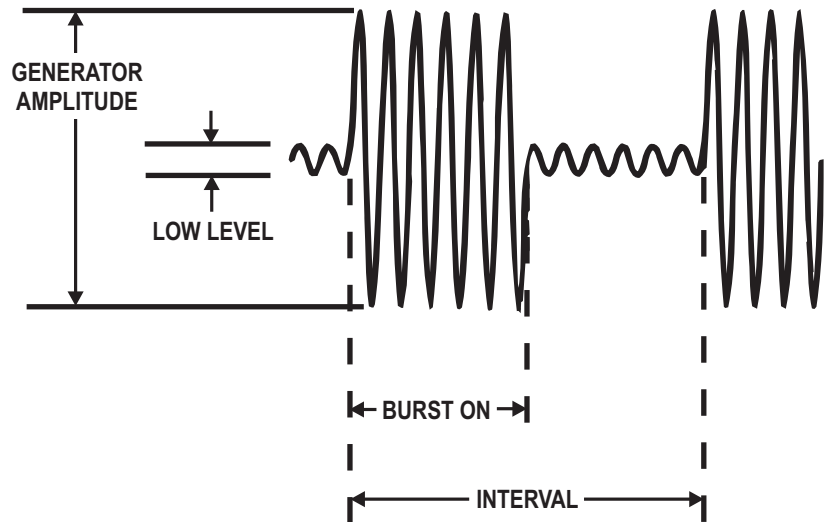


Figure 4-7 Tone Burst Definitions, Burst Normal Waveform

Low Level line will be displayed when Burst Gated waveform is selected.

### Burst Length (Duration)

The maximum Burst length is 65,535 cycles of sinewave. The burst “on” time may be set in cycles.

### Burst Interval

Interval is the time from the beginning of one burst to the beginning of the next burst, not the time between the end of one burst and the beginning of the next. The maximum burst Interval is 65,536 cycles of sinewave.

### Low Level

The amplitude of signal between bursts is controlled by the **Low Level** parameter. The Low Level value is relative to the Burst “on” level, which is the calibrated steady-state level controlled by the Amplitude field. The Low Level (low level) line controls the amplitude of the generator output between bursts. In triggered and gated mode, the generator output will be at the Low Level amplitude between



triggered bursts or gated-on portions. The Low Level amplitude may be set in any of four units.

**%** lower level expressed as a percentage of calibrated level

**dB** lower level in decibels relative to calibrated level

**X/Y** ratio of lower level (X) to calibrated level (Y)

**PPM** lower level in Parts Per Million of calibrated level value

The lower level amplitude may equal the upper level amplitude, in which case no burst, gated, or triggered effect will be visible. At large amplitude ratios, the amplitude resolution of the lower level becomes poorer. The display will show the actual available resolution steps, rather than the entered value. The lower level is limited to -80 dB (0.01%).

The Interval line is blanked in Burst Triggered waveform, since the burst interval will be determined by the external trigger source.

Both the Burst (duration) and the Interval lines are blanked in Burst Gated waveform, since both these parameters are determined by the external gating signal.

### Triggered Bursts

In triggered operation (Burst Triggered waveform mode), one burst of the specified Burst duration will be generated for each trigger presented to the TRIGGER/GATE input. This input is a BNC connector on the generator auxiliary connector panel. The input is LSTTL compatible. In Burst Triggered mode, it is intended to be driven by a signal that is at a logic high level but pulses low for at least one microsecond to trigger a burst. The figure shows the relationships between the trigger signal and the generator output for an example

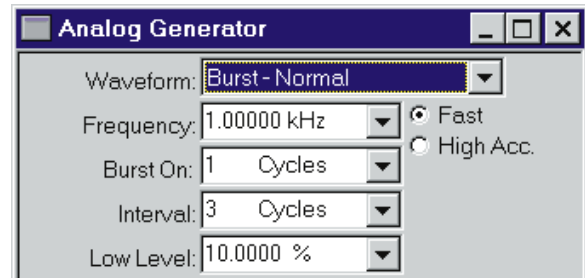


Figure 4-8 Tone Burst Control Fields

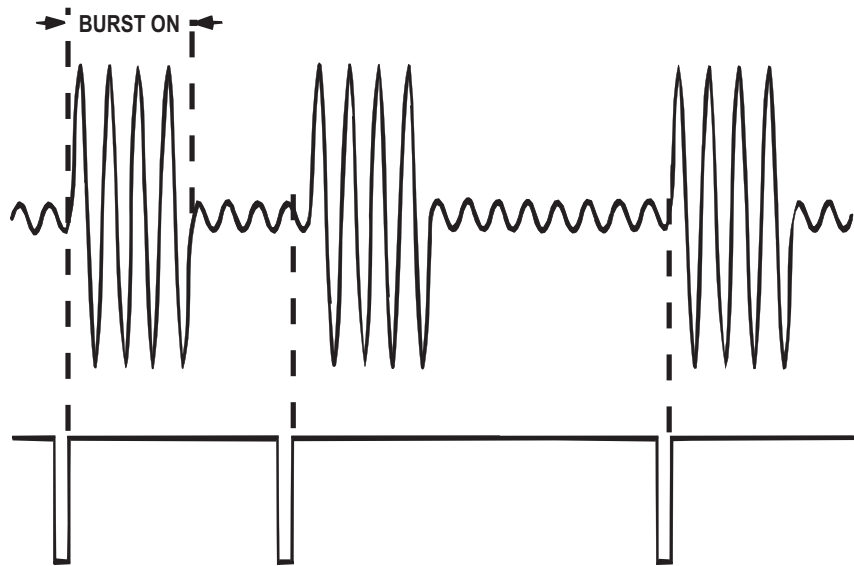


Figure 4-9 Burst Triggered Mode, Definitions

burst of four cycles. The burst will be triggered by the positive-going (trailing) edge of such a signal. The burst will begin at the first positive-going zero crossing of the sinewave signal that follows the positive-going edge of the trigger signal. If the TRIGGER/GATE input signal remains at the logic high level after triggering the burst and between bursts, the burst duration will be as set on the panel and the generator output amplitude will remain at the Low Level value until the next trigger. If the TRIGGER/GATE input drops to the low logic level during a burst, it will gate the signal off (terminate the burst prematurely) at the next positive-going zero crossing of the sinewave.

Bursts will always consist of an integral number of cycles, beginning and ending at positive-going zero crossings. Thus, there may be a delay of up to one sinewave period between the positive-going trigger pulse at the external connector and the beginning of the burst.

### Gated Bursts

Gated operation (Burst Gated mode) allows an external signal at the TRIGGER/GATE connector to control whether the generator output amplitude is at the upper value (Analog Generator panel

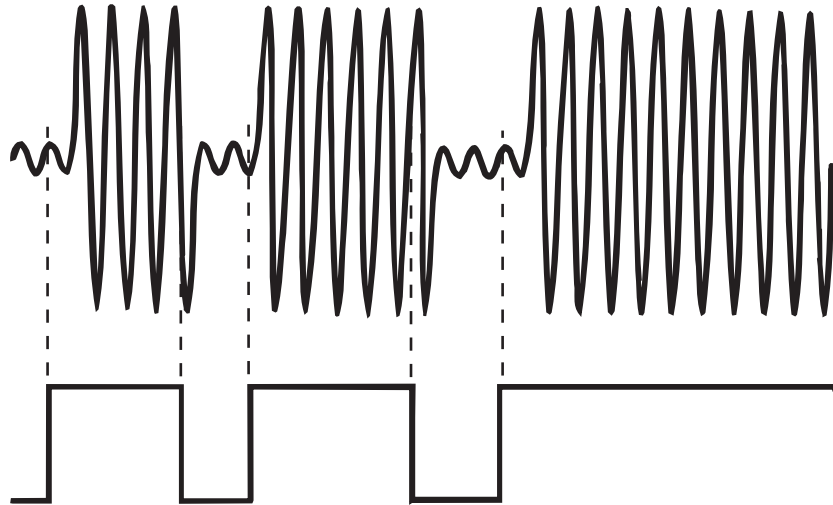


Figure 4-10 Burst Gated Mode, Definitions

Amplitude setting) or the Low Level value. Positive, LSTTL-compatible logic conventions are used. Thus, the sinewave amplitude will be at the upper level when the trigger/gate input is high and at the Low Level when the input is low. When no external control device is connected to the TRIGGER/GATE input, it is pulled high by an internal resistor and the generator output will be at the high level. The actual output gating always takes place at positive-going zero crossings, so there can be up to a one-period delay at both the gate-on and gate-off transitions. The figure shows the timing and logic relationships in Burst Gated mode. Note that the TRIGGER/GATE input connector is functional in the Burst Normal, Burst Gated, and Burst Triggered modes. If this control input is pulled to a logic low condition, it will gate the signal to the Low Level amplitude even during internally-controlled burst mode or during an externally-triggered burst.

## Noise Waveforms

Several varieties of noise waveforms are available from the BUR-GEN option. Both the Noise and Pseudo Noise selections produce noise waveforms. Pseudo Noise synchronizes with the 4/second reading rate of the analog analyzer to produce stable,

repeatable measurements. Noise (true random) waveforms have a continuous spectrum and produce measurements that do not repeat from reading to reading. Spectral distribution in either mode may be chosen between pink noise, white noise, or 1/3 octave bandpass filtered noise. The white noise selections are bandwidth-limited to 22 kHz.

### Pseudo and Random Noise

The Noise Pseudo waveforms produces noise that is random during a 262 millisecond period, but which then repeats every 262 milliseconds. This repetition cycle synchronizes with the nominal 4/sec reading rate of the analyzer, producing stable displays. The Noise waveforms are truly random and never repeat. Readings made using Noise (not Pseudo Noise) waveforms will not be stable at any reading rate. The noise spectrum in the Noise waveforms will have energy at all frequencies within its specified bandwidth; that is, the spectral lines will be infinitesimally closely spaced. In Pseudo Noise waveforms, the spectral lines will be spaced at the repetition rate of the pseudorandom cycle, or approximately every 4 Hz from 4 Hz to the upper bandwidth limit. This signal may not be acceptable for certain applications, particularly at very low frequencies. However, if the measurement interval is limited, it does no good to have a noise signal which repeats less often.

Amplitude calibration in the Pseudo Noise waveforms is in terms of equivalent sinewave peak, as with other complex waveforms such as intermodulation test signals and squarewaves. The Pseudo Noise waveforms of the BUR-GEN have a crest factor (ratio of peak to RMS) of 4:1 (12 dB). Since a sinewave has a crest factor of 1.414:1 (3 dB), the RMS value of the Pseudo Noise waveforms at any given Amplitude setting will be 9 dB less than that of a sinewave at the same Amplitude. In the Noise waveforms (not Pseudo), amplitude calibration is to approximately the same RMS value as in Pseudo Noise waveforms. The result is that in random Noise waveforms, occasional noise peaks may exceed the peak-to-peak value of a sinewave of the same Amplitude. While the theoretical crest factor in (random) Noise waveforms is infinite, a 4:1 crest factor will be exceeded only 0.01% of the time. The maximum available Amplitude setting in the noise modes is half that of the sinewave modes.

In either Pseudo Noise or (random) Noise waveforms, additional selections of the noise spectral distribution may be made among Pink, White, or BP (Bandpassed).

## White Noise

The Noise White and Pseudo Noise White waveforms select white noise mode. This waveform is bandwidth limited to 22 kHz to maximize the noise energy falling within the audio band. The spectral distribution of white noise is characterized by equal noise energy per Hz of bandwidth. The spectral range between 100 Hz and 200 Hz will thus have the same energy as the range between 10,000 Hz and 10,100 Hz. If analyzed by a constant bandwidth spectrum analyzer such as a superheterodyne or FFT analyzer, white noise will show a flat energy characteristic versus frequency (up to the bandwidth limitation). Analysis with a constant-percentage-bandwidth (constant Q) filter such as in bandpass mode (or most real-time analyzers) will show a rising characteristic versus frequency, at the rate of 3 dB per octave.

## Pink Noise

The Noise Pink and Pseudo Noise Pink waveforms select pink noise mode. Pink noise is characterized by equal noise energy per fractional octave, fractional decade, or equal percentage bandwidth. Thus, the octave of pink noise between 5 kHz and 10 kHz will contain the same energy as the octave between 300 Hz and 600 Hz. A constant-percentage-bandwidth analyzer such as bandpass mode and most real-time audio analyzers will show a flat characteristic with frequency. A constant bandwidth spectrum analyzer such as a superheterodyne or FFT analyzer will display pink noise as having a fall-off with increasing frequency, at the rate of 3 dB per octave. Filtering the basic white noise source through a 3 dB per octave filter generates the BUR-GEN's pink noise. Pink noise will sound subjectively flat to the ear because the sounds are perceived on a constant bandwidth basis. When checking response of devices such as multiway loudspeaker systems, pink noise will supply more equal levels to low, midrange, and high frequency drivers than will white noise.

## Noise Bandpassed Waveform


Noise BP (Bandpassed) and Pseudo Noise BP (Bandpassed) waveforms select the pink noise waveform, but further process the noise by passing it through a 1/3 octave tunable bandpass filter whose center frequency is controlled by the Frequency field of the Analog Generator panel. This filter is, in fact, the basic state-variable oscillator circuit of generator, used as a bandpass filter. The filter center frequency can thus be tuned from 10 Hz to 204 kHz.

The center frequency can be controlled from the Generator Frequency field and can be swept at either Source-1 or Source-2 on the Sweep and Advanced Sweep panels. Units may be Hz and kHz or any of the generator relative frequency units.


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## Frequency Control and Units

The Frequency field controls the frequency of most waveforms. The frequency of digitally-generated waveforms (DGen waveform selection) is controlled from the digital generator section of the Digital Analyzer (DSP) panel, so the Analog Generator Frequency field disappears when DGen is selected. Except for the Bandpassed Noise selections, noise waveforms have no controllable frequency parameter so the Frequency field disappears. In the SMPTE-DIN and CCIF IMD waveforms, a second frequency parameter may be selected so an IM Frequency field will be displayed just below the main frequency field.

Note that there are interactions between frequency, maximum amplitude, and waveforms. Below 10 Hz and above 50 kHz, maximum available sinewave amplitude is half that at normal audio frequencies. Burst and square waveforms are not available across the full sinewave frequency range. Frequency may be expressed in absolute units (Hz), or in a variety of units relative to the Reference Frequency value near the bottom of the Analog Generator panel. The Frequency box is a numeric entry field. New values may be typed or existing values edited from the keyboard, followed by the  key.

Note that changing units does not change the frequency of the generator. The existing frequency will simply be re-stated in the new

units. Note also that the value displayed after pressing  may differ slightly from the value entered, since APWIN software will display the nearest frequency which the instrument can actually generate.

The IM Frequency field is selected in the same manner as the main frequency field. However, the IM generator option has only seven fixed choices for IM frequency. If a value is entered into this field from the keyboard, APWIN software will select the nearest IM frequency from the available choices.

### Fast-High Accuracy Modes

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Fast mode produces the most rapid frequency settling along with frequency accuracy and resolution suitable for nearly all audio tests. High accuracy mode provides greater accuracy and resolution, but requires from 150 milliseconds (above 50 Hz) to 750 milliseconds (at 10 Hz) for complete settling each time the frequency is changed.

## Output On/Mute

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The On-Off button turns on or off the output channels whose individual output control buttons (CH A and CH B) are on, as indicated by the green color. The On-Off button is a “toggle” function, reversing output conditions each time it is pressed. When off, outputs are reverse-terminated in the selected source impedance.

## Channel Selection


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The CH A and CH B channel selection buttons determine whether output channel A, B, both, or neither will have signal present when the On-Off button is On. The lowest noise and signal feed-through condition is obtained by turning the On-Off button off rather than turning an individual channel off. The System One Analog Generator has only one power amplifier and output transformer, so the same signal will be present at both outputs when they are both selected.


## Channel B Invert

The Channel B output may be phase-inverted (180 degrees) with respect to Channel A by checking the Invert box near the B On-Off button. Phase inversion of one channel is used to create subcarrier rather than main carrier output in stereo multiplex systems. When there is no check mark in the box, the two outputs are in phase.

## Amplitude Control and Units

Generator output amplitude is determined by the setting in the Amplitude field. Since both channels are driven from the same power amplifier and transformer, the same amplitude will appear at both outputs if they are driving equal load impedances. Amplitude may be stated in a variety of practical units. A new amplitude value may be entered or the existing value edited from the keyboard, followed by the  key.

Note that there are interactions between maximum available amplitude and waveform, frequency, and output configuration. Maximum available sinewave amplitude at the frequency extremes (below 20 Hz, above 50 kHz) is half that at standard audio frequencies. Maximum amplitude in unbalanced configurations is half that in balanced mode. Maximum amplitude with burst, noise, and square waveforms is half that with sinewaves.

Note that the actual output amplitude does not change when units selections are changed, but the existing amplitude is simply re-stated in the new unit. Note also that the value displayed after pressing  may differ slightly from the value entered, since APWIN software will display the nearest frequency which the instrument can actually generate.

## Output Configuration

The generator outputs may be configured as balanced or unbalanced and floating or grounded. Additionally, a common mode rejection ratio test configuration is available to simplify measurement of



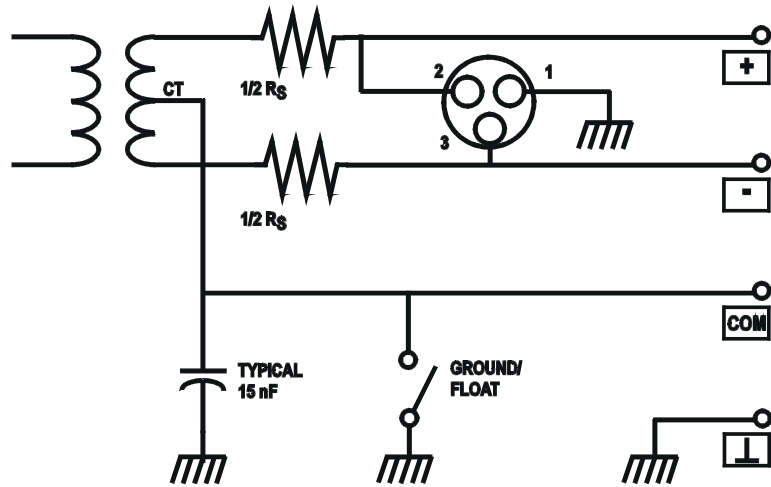


Figure 4-11 System One Analog Generator, Simplified Output Circuit in Balanced Configuration

CMRR of a device input. Configurations of both channels A and B are slaved together.

Note the interaction between Configuration and Z-Out (output impedance). The medium output impedance is not available in unbalanced configurations. The lowest output impedance is half the value in unbalanced mode.

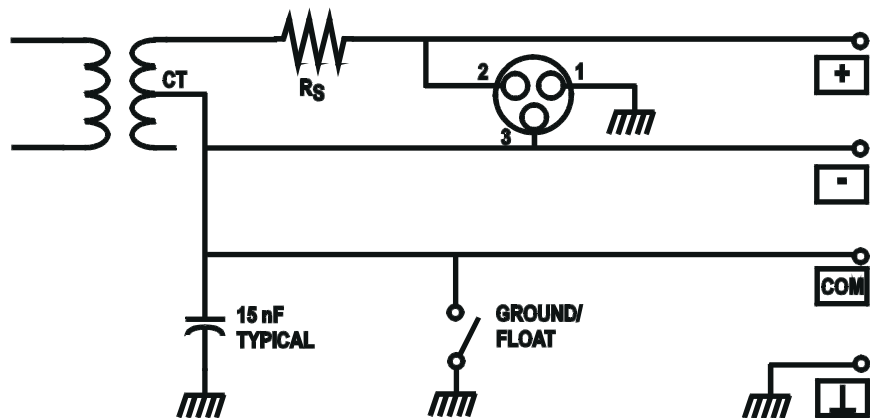


Figure 4-12 System One Analog Generator, Simplified Output Circuit in Unbalanced Configuration

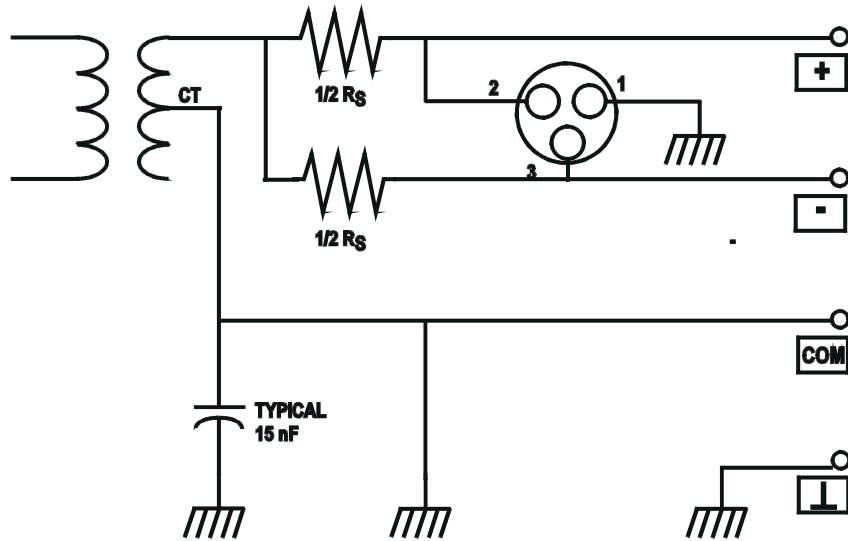


Figure 4-13 System One Analog Generator, Simplified Output Circuit in CMTST (Common Mode Test) Configuration

In the **balanced** output configuration, the two ends of the output transformer drive pins 2 and 3 or the XLR connector and the + and - banana jacks. Half of the selected output impedance is connected in series with each leg. The common banana jack connects to the transformer center tap. This common connection may be selected on the panel as grounded or floating.

In the **unbalanced** output configuration, only half the transformer secondary is used. Pin 2 of the XLR and the + banana jack connect to the top end of the output transformer through the selected output impedance. The center tap of the transformer connects directly to pin 3 of the XLR and to the minus and common banana jacks. If the output configuration is selected as ground, that common conductor is also connected to ground. Since only half the transformer secondary is used in unbalanced configuration, the maximum available amplitude is half that (6.02 dB less) of balanced configuration.

In the **CMTST** (common mode test) configuration, only half the transformer secondary is used so maximum available amplitude is half that of the balanced configuration. The top of the transformer secondary connects to two resistors, each half of the selected output impedance value. One of those resistors drives pin 2 of the XLR and

the + banana jack and the other resistor drives pin 3 of the XLR and the minus banana jack. The transformer center tap connects to the common banana jack and to ground. Thus, XLR pins 2 and 3 (or the + and - banana jacks) are driven with equal amplitude, in-phase signals with respect to ground. This type of signal will be largely rejected by well-designed balanced (differential) inputs.

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## Impedance Selection

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Generator output impedance may be selected as 50, 150, or 600 Ohms in balanced configurations; if the Euro-Z option is present, these values are <40, 200, and 600 Ohms (see the European Broadcast Impedances section on page 3-37 for information on setting APWIN software to display the European impedances when the option is installed). In unbalanced configurations, the medium value is not available and the low value is cut in half. The output impedance control of channels A and B are slaved together.

Except when dBm or Watts units are selected for generator amplitude, changing the output impedance does not change the open-circuit voltage (emf) behind that output impedance. With a finite load impedance, the actual terminal voltage will change as output impedance changes since the effective voltage divider ratio between source and load also changes. When dBm or Watts units are selected, APWIN computes the required open circuit voltage necessary to produce the specified power (Amplitude field) across the specified external load value (Reference dBm or Reference Watts fields). Since this open circuit voltage value also depends on generator output impedance, open-circuit voltage will change when source impedance is changed since power in the load is held constant.


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## dBm Reference

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See the AC Amplitude Units section starting on page 14-1 for a discussion and definition of dBm units.



When dBm units are selected in the generator Amplitude field, APWIN software computes the necessary emf (open-circuit voltage) which will cause the specified power (Amplitude field) to be delivered

to the specified load impedance (dBm Reference field). Thus, the user must enter the actual value of device input impedance into the dBm Reference field for output power to be properly calibrated. A new value may be entered or the existing value edited from the keyboard, followed by the  key.

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## dBr Reference

See the AC Amplitude Units section starting on page 14-1 for a discussion and definition of dBr units.

When dBr units are selected in the generator Amplitude field, the actual output amplitude is computed based on the decibel value entered in the Amplitude field and the dBr reference value. The present value of generator amplitude may be automatically written into the dBr Reference field by pressing the  function key or selecting the Edit Set Gen dBr Ref menu item. A new value may be entered or the existing dBr Reference value edited from the keyboard, followed by . The dBr Reference value may be stated in any of a number of amplitude units.

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## Frequency Reference

The Analog Generator has a number of relative frequency units in addition to absolute frequency in Hz. When any relative frequency unit is selected, APWIN computes and sets the actual frequency based on the relative value in the generator Frequency field and the Frequency Reference value. The present value of Analog Generator frequency may be written into the Frequency Reference field by pressing the   keys or selecting Edit Set Generator Freq Ref menu item. A new value may be entered or the existing value edited from the keyboard, followed by .

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## Watts Reference

See the AC Amplitude Units section starting on page 14-1 for a discussion and definition of Watts units..

When Watts units are selected in the generator Amplitude field, APWIN software computes the necessary emf (open-circuit voltage) which will cause the specified power (Amplitude field) to be delivered to the specified load impedance (Watts Reference field). Thus, the user must enter the actual value of load impedance into the Watts Reference field for output power to be properly calibrated. A new value may be entered or the existing value edited from the keyboard, followed by .

## Hardware Overview and Concepts

### Analog Generator Overview

The System One analog generator consists of an oscillator section and an output stage section. This block diagram shows the generator in simplified form. The oscillator section always includes a variable frequency low-distortion sinewave oscillator and a signal multiplexer to allow selection of other optional waveforms. The IMD option adds generation of three standard intermodulation distortion (IMD) test signals. The BUR option adds a squarewave, sine burst, and noise generator. If DSP modules are present, DSP-generated signals may be

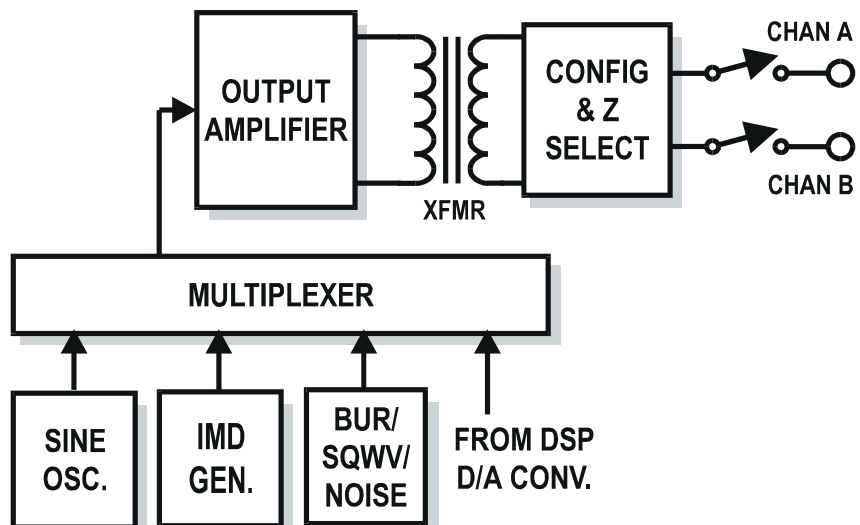


Figure 4-14 System One Analog Generator, Simplified Diagram

routed from a D/A converter in the DSP module through the multiplexer to the generator output stage.

The sinewave oscillator uses state-variable topology to optimize low distortion and fast settling time. Capacitors are switched to select one of four frequency ranges: 10-204 Hz, 204 Hz-2.04 kHz, 2.04 kHz-20.4 kHz, or 20.4 kHz-204 kHz. Within each range, a 13-bit MDAC serves as a variable resistor to set the oscillation frequency anywhere across the 10:1 range (20:1 on the lowest range). A leveling circuit (not shown) operates continuously to provide output amplitude that is independent of frequency. The sinewave output feeds a multiplexer that selects among the various waveforms that may be present. The multiplexer output goes to a high-resolution (fine) amplitude control block shown in the output stage block diagram.

When the BUR (squarewave-sine burst-noise) option is present, the main sinewave oscillator output may be fed through the BUR circuit block where it can be synchronously gated or triggered. See the Sine Burst topic for more details. The sinewave from the main oscillator also serves as the frequency-determining source for the squarewave signal generated on the BUR circuit board. The BUR board can also generate pseudorandom or random noise of white or pink frequency distribution. In BP NOISE function, the pink noise output may be fed through the sinewave oscillator frequency-determining circuitry functioning as a 1/3-octave bandpass filter to produce narrowband noise at any center frequency from 10 Hz to 204 kHz.

When the IMD option is present, the main sinewave oscillator provides one of the two IMD test signals in the case of the SMPTE/DIN or DIM/TIM standards. The main sinewave is combined in the IMD module in the proper amplitude relationship with either a low-frequency sinewave selected among several frequencies (SMPTE/DIN) or with a fixed-frequency squarewave (DIM/TIM), and sent to the multiplexer. For the “CCIF” (difference tone) IMD standard, the main sinewave oscillator provides the “carrier frequency” into a balanced modulator. The low-frequency oscillator in the IMD module supplies the modulating frequency to the balanced modulator. The resulting output from the balanced modulator is a double-sideband suppressed-carrier signal, consisting of the two equal-amplitude test

tones (“sidebands”) equally spaced in frequency above and below the main oscillator frequency.

When DSP modules are present, a number of digitally-generated waveforms can be converted to the analog domain via a D/A converter on the DSP module and routed through the analog generator amplitude control and output stage via the DGen waveform selection.

### Analog Generator Output Stage

The System One analog generator power amplifier section includes amplitude control and output configuration control functions. Signal from the output of the oscillator section feeds the high-resolution (fine) amplitude control. This control uses a 13-bit multiplying digital-to-analog converter (MDAC) as the variable element of an attenuator, producing amplitude steps of 1.27 microvolts. In the top 12 dB (lowest attenuation portion) of this attenuator’s range, these amplitude steps result in resolution of 0.001 dB to 0.005 dB. An attenuator that may be set to zero dB (unity gain) or 24 dB loss follows this high-resolution amplitude control circuitry. The output of this

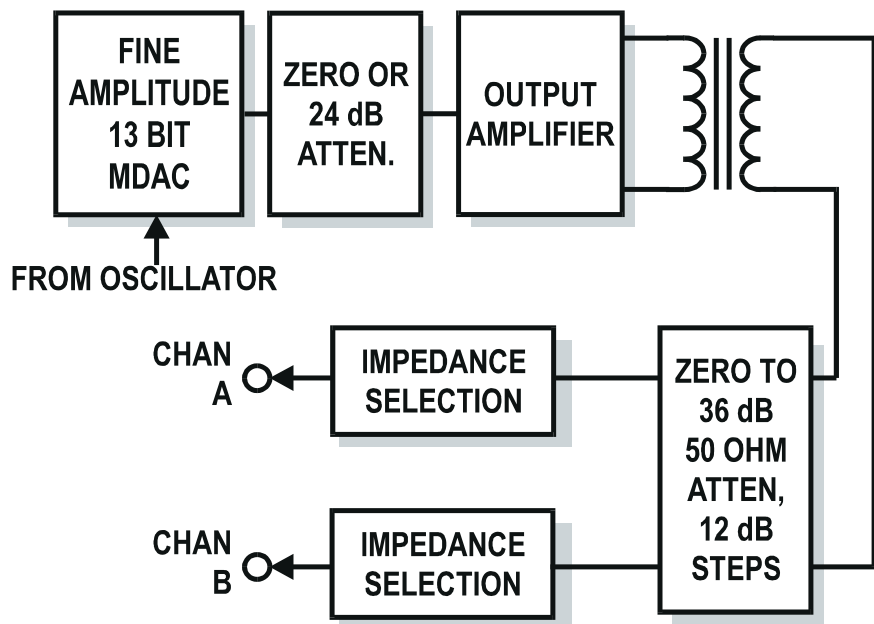


Figure 4-15 System One Analog Generator, Simplified Diagram of Output Section

block drives the generator power amplifier. The power amplifier is transformer coupled with its balanced output connected to a passive attenuator consisting of three 12-dB steps. At the output of the attenuator, the signal is resistively split and fed to output impedance-determining (“build-out”) resistors for output channels A and B. Selectable resistors provide for three values of output impedance. The net resolution of the various amplitude control sections results in 0.01 dB or better resolution from maximum output down to approximately one millivolt, with the 1.27 microvolt steps then amounting to a progressively larger dB value as the output signal is reduced below the one millivolt amplitude.

Control of these various attenuators is managed by APWIN software so as to be essentially transparent to the user. The user enters the desired value of output amplitude and the correct settings of each attenuator are automatically selected to provide the highest quality output signal.




# Analog Analyzer

## APWIN Operation and Panels

The System One Analog Analyzer contains four measuring instruments—the Level Meter, Phase Meter, Frequency Counter, and the Reading Meter. The first three meters are single-function instruments, while the Reading Meter is flexible with many different measurement functions. All selectable or tunable filtering and the selectable detector types are in the Reading Meter section.

To bring the Analog Analyzer panel to the screen:

Mouse: Click on the Analog Analyzer icon  on the toolbar or select Panels Analog Analyzer via the menu bar

Keyboard: Type  or  (for Panels analog Analyzer).

The normal default start-up mode of APWIN software has the Analog Analyzer panel on page one.

The Analog Analyzer panel may be displayed in two sizes. The smaller version (displayed at start-up) shows only the most important controls and displays. Displays on the small panel are the Level, Frequency, Phase, and Reading meters with unit selections. Controls on the small version include input connector selection and the A-B channel selection for the Reading meter. The large version adds input termination impedance selection controls, auto-fixed range check

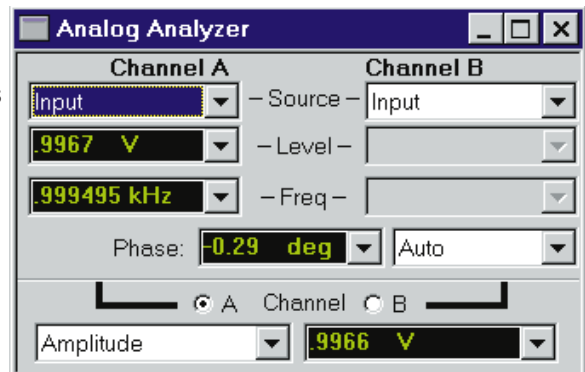




Figure 5-1 System One Analog Analyzer Control Panel, Small Version

boxes and manual range selection controls for the input ranging sections. It also displays the auto-fixed range check box and manual range selection controls for the Reading meter range amplifier and controls for meter reading rate, detector type, high pass filters, low pass filters, plug-in option filters, bandpass-bandreject filter steering mode and frequency control field when fixed. At the bottom are reference value fields for dBm, dBr, Watts, and frequency. Clicking on the panel icon on the Panels Toolbar brings up the small version panel. Holding down the **SHIFT** key while clicking on the toolbar icon brings up the large version. To switch between the small and large panels:

Mouse, Windows 3.11: Click on the top right corner icon  on the panel, in the control bar.

Mouse, Windows 95: Click on the middle  of the three icons at the top right of the panel , in the control bar.

Keyboard: Press the **CTRL**  or **CTRL**  keys.

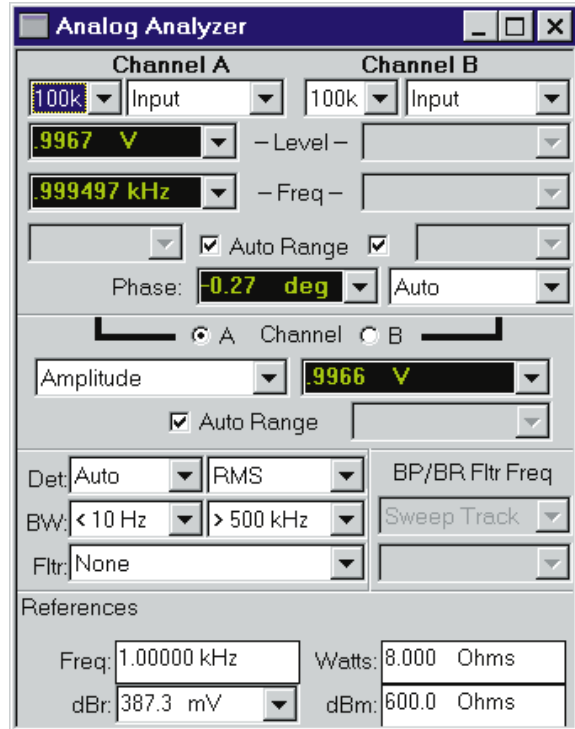


Figure 5-2 System One Analog Analyzer Control Panel, Large Version

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## Input Termination Impedance

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Input termination impedance at the System One analog analyzer XLR or banana input connectors is selectable via a field at the top left corners of the channel A and B sections of the large version of the analyzer panel.

The BNC “AUX INPUT” connector input impedance is always high impedance.

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## Input Source Selection

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The typical System One Analog Analyzer input panel contains XLR and double banana connectors for both channels plus a BNC auxiliary input for channel A. The double banana jacks are hard-wired to pins 2 and 3 on the XLR connectors on each channel. System One also has internal cables installed from the analog generator outputs to selector relays at the analyzer inputs. The source selection field on the Source line at the top of the analyzer panel selects between the XLR/banana jack inputs and the generator monitor facility (or the Auxiliary BNC connector on Channel A).

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## Level Meter

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The Level Meter measures amplitude of either Channel A or Channel B. In all Reading Meter functions except the “2-Channel” functions, the Level Meter measures the same channel as the Reading Meter. In the 2-Channel functions, the Level Meter measures the opposite channel from that measured by the Reading Meter. Amplitude measurements made by the Level Meter may be expressed in a number of different units.

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## Frequency Counter

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The Frequency Counter measures frequency of either Channel A or Channel B. In all Reading Meter functions except the “2-Channel” functions, the Frequency Counter measures the same channel as the Reading Meter. In the 2-Channel functions, the Frequency Counter measures the opposite channel from that measured by the Reading

Meter. Frequency measurements made by the Frequency Counter may be expressed in absolute units (Hz) and a number of different relative units referred to the Reference Frequency field near the bottom of the panel.

## Input Ranging Control and Manual Selection

Each input channel (A and B) includes attenuators and selectable gain amplifiers designed to bring input signals across a wide amplitude range to the optimum level for System One's detectors. The two channels are independent. The autorange circuit detectors are peak responding to prevent overload even with high crest factor signals. For most applications, it is most convenient (and least error-prone) to allow the Autoranging circuitry to select the proper range. For certain applications where measurements must be made on signals whose amplitude constantly changes, such as tone burst signals or program

material, it may be necessary to disable autoranging and manually select a range which will handle the expected peak signal amplitude. When Autoranging is selected

on either channel (Auto box is checked), the manual range selection field is gray and non-functional. When the box is unchecked to disable autoranging, the range field will display the full-scale value of the range which had been automatically selected. The range may be expressed in a number of different units, selectable via the down arrow at the right of the field. A different range may be manually selected.


To select a new manual range, enter the value of the largest expected signal amplitude into the range field. A new value may be entered or the present range setting edited from the keyboard after selecting



Figure 5-3 Analyzer Input Ranges Automatically Controlled



Figure 5-4 Analyzer Input Ranges Fixed at User-Entered Values

the range field of the desired channel. APWIN software will choose the most sensitive range that will not be overloaded by a signal of the specified amplitude. It will display the full-scale value of the selected range when  is pressed after making or editing an entry.

## Phase Meter Units

The field at the right of the Phase meter display permits selection of four phase ranges: Auto,  $-180+180$ ,  $0+360$ , or  $-90+270$ .

When **Auto** is selected, the Phase meter ranges automatically. When phase is graphed versus frequency in a sweep and the phase shift or delay through the device under test is large enough to go through more than one complete revolution, the plotted phase will be “unwrapped”. That is, phase will plot continuously through thousands of degrees if necessary, assuming that the size of the frequency steps is small enough that measured phase does not change by too large an amount from point to point.

When  **$-180+180$**  is selected, the panel display and graph plots are constrained to a range between minus 180 degrees and plus 180 degrees. Plots of phase will jump abruptly from minus 180 to plus 180 (or vice-versa) on adjacent frequency steps if the measured phase crosses the end of the range.

When  **$0+360$**  is selected, the panel display and graph plots are constrained to a range between 0 degrees and plus 360 degrees. Plots of phase will jump abruptly from 0 to plus 360 (or vice-versa) on adjacent frequency steps if the measured phase crosses the end of the range.

When  **$-90+270$**  is selected, the panel display and graph plots are constrained to a range between minus 90 degrees and plus 270 degrees. Plots of phase will jump abruptly from minus 90 to plus 270 (or vice-versa) on adjacent frequency steps if the measured phase crosses the end of the range. This range is particularly useful when measuring devices with low values of phase shift for polarity. In this case the readings are likely to be within a few degrees of zero or within a few degrees of 180 and it is simple to put limits on the results of a measurement for Pass/Fail testing.

## Input Channel Selection

The buttons above the Reading Meter display determine whether the System One Reading Meter measures Channel A or B. In most functions of the Reading Meter, the Level Meter and Frequency Counter are also switched to the same channel by these buttons. In the “2-Channel” functions of the Reading Meter, the Level Meter and Frequency Counter measure the opposite channel from the Reading Meter.



Figure 5-5 Reading Meter Channel A/B Selector

## Reading Meter Function Selection

The flexible Reading Meter can measure a number of different audio functions by virtue of its tunable bandpass-bandreject filter, selectable bandwidth-limiting and weighting filters, and selectable detectors. The IMD analyzer option and wow and flutter option also feed their processed signals to the Reading Meter for final display. The complete list of measurement functions available, if the IMD analyzer and wow and flutter options are present, is shown in Figure 5-6.

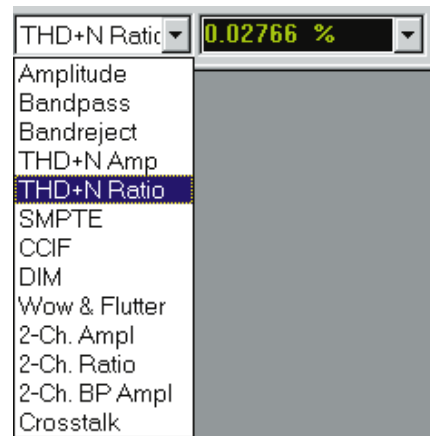


Figure 5-6 Reading Meter Function Selection List

### Reading Meter Function Amplitude

In the Amplitude function, the amplitude of the selected channel signal is measured and displayed. Detector type, reading rate, high-pass filter, low-pass filter, and plug-in option filter may be selected.

## Reading Meter Function Bandpass

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In the Bandpass function, the selected channel signal is passed through a four-pole 1/3 octave tunable filter before its amplitude is measured and displayed in order to accomplish a frequency-selective amplitude measurement. The center frequency of the filter may be set anywhere in the range from 10 Hz to approximately 204 kHz. The filter center frequency may be made to automatically track: the Source 1 or Source 2 sweeping parameter (Sweep Track), the measurement of the Frequency counter on the same channel (Counter Tuned), the analog generator (AGen Track), or to be fixed at a user-entered frequency value (Fixed) by the selection of the BP/BR Filter Freq field. Detector type, reading rate, high-pass filter, low-pass filter, and plug-in option filter may be selected.

## Reading Meter Function Bandreject

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In the Bandreject function, the selected channel signal is passed through a tunable bandreject (notch) filter before its amplitude is measured and displayed. The center frequency of the filter may be set anywhere in the range from 10 Hz to approximately 204 kHz. The filter center frequency may be made to automatically track the Source 1 or Source 2 sweeping parameter (Sweep Track), the measurement of the Frequency counter on the same channel (Counter Tuned), the analog generator (AGen Track), or to be fixed at a user-entered frequency value (Fixed) by the selection of the BP/BR Filter Freq field. Detector type, reading rate, high-pass filter, low-pass filter, and plug-in option filter may be selected.

## Reading Meter Function THD + N Amplitude

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In the THD+N Amplitude function, the selected channel signal is passed through a tunable bandreject (notch) filter in order to reject the fundamental component of an applied single sinewave test signal before the remaining distortion and noise components amplitude is measured and displayed. An electronic servo continuously adjusts the filter frequency for maximum rejection of the highest-amplitude frequency component of the signal. The center frequency of the filter may be set anywhere in the range from 10 Hz to approximately 204 kHz. The filter center frequency may be made to automatically track

the Source 1 or Source 2 sweeping parameter (Sweep Track), the measurement of the Frequency counter on the same channel (Counter Tuned), the analog generator (AGen Track), or to be fixed at a user-entered frequency value (Fixed) by the selection of the BP/BR Filter Freq field.

The resulting harmonic distortion and noise amplitude measurement is displayed in a selection of absolute units (Volts, dBV, dBu, etc.). This THD+N Amplitude function is particularly useful for measuring distortion during amplitude sweeps. Detector type, reading rate, high-pass filter, low-pass filter, and plug-in option filter may be selected.

### **Reading Meter Function THD + N Ratio**

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In the THD+N Ratio function, the selected channel signal is passed through a tunable bandreject (notch) filter in order to reject the fundamental component of an applied single sinewave test signal before the remaining distortion and noise components amplitude is measured and displayed. An electronic servo continuously adjusts the filter frequency for maximum rejection of the highest-amplitude frequency component of the signal. The center frequency of the filter may be set anywhere in the range from 10 Hz to approximately 204 kHz. The filter center frequency may be made to automatically track the Source 1 or Source 2 sweeping parameter (Sweep Track), the measurement of the Frequency counter on the same channel (Counter Tuned), the analog generator (AGen Track), or to be fixed at a user-entered frequency value (Fixed) by the selection of the BP/BR Filter Freq field.

The resulting harmonic distortion and noise amplitude measurement is displayed in a selection of relative or ratio units (% , dB, parts per million, X/Y, etc.). The THD+N Ratio function is the most commonly used distortion measurement function, particularly when the signal amplitude is constant as during a frequency sweep. Detector type, reading rate, high-pass filter, low-pass filter, and plug-in option filter may be selected.



## Reading Meter Function SMPTE/DIN (Intermodulation Distortion)

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The IMD analyzer option must be installed for this function to operate. In the SMPTE IMD analysis mode (which also analyzes DIN IMD signals), the expected test signal consists of two sinewaves plus intermodulation distortion products. The lower-frequency sinewave may be anywhere in the range from 40 Hz to 500 Hz. The higher-frequency sinewave may be anywhere in the range from 2 kHz to 200 kHz. The signal is fed through a 2 kHz high-pass filter, which strips off the low-frequency tone component of the signal. An amplitude modulation demodulator follows, providing output amplitude proportional to the sidebands (IMD products) located near the high-frequency “carrier” signal. This signal is further low-pass filtered to remove the “carrier” and is fed on to the bandpass/bandreject (BP/BR) filter block which is not used during SMPTE IMD analysis. The final result is measured and displayed in the main analyzer display field below the channel selection buttons. Only relative units (dB, %, PPM, X/Y) are available in the IMD functions since all IMD standards refer the distortion measurement to the amplitude of one of the signal components. The BW fields (high pass and low pass filter selection) are gray and unavailable in IMD modes since selection of these filters could cause erroneous readings.

## Reading Meter Functions CCIF/DFD

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The IMD analyzer option must be installed for this function to operate. In the CCIF and DFD (difference frequency distortion) modes, the expected signal consists of two equal-amplitude sinewaves, relatively closely spaced about a relatively high-frequency center value, plus intermodulation distortion products. Only the low frequency difference product ( $f_2 - f_1$ ) amplitude is measured by the analyzer. Spacing between the two sinewaves may be anywhere between 80 Hz and 2 kHz. The center frequency value may be anywhere from 4 kHz to 200 kHz. The selected channel signal is fed directly to the 2.45 kHz low-pass filter and 30 Hz high-pass filter and on to the BP/BR filter. The BP filter, in bandpass mode, is automatically tuned to the difference frequency product ( $f_2 - f_1$ ) in CCIF and DFD modes. Thus, other IMD products and wide-band noise are rejected. The result is measured and displayed by the main analyzer. Only relative units (dB,

%, PPM, X/Y) are available in the IMD functions since all IMD standards refer the distortion measurement to the amplitude of the test signal components. The BW fields (high pass and low pass filter selection) are gray and unavailable in IMD modes since selection of these filters could cause erroneous readings. CCIF and DFD functions differ by a 6.02 dB calibration difference, with the DFD function providing the lower values according to the most recent revision of the IEC-268 specification.

### **Reading Meter Function DIM/TIM**

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The IMD analyzer option must be installed for this function to operate. In the DIM/TIM (dynamic intermodulation/transient intermodulation) mode, the expected signal consists of a squarewave at approximately 3 kHz combined with a sinewave at approximately 15 kHz, plus intermodulation distortion products. The selected channel signal is fed directly to the 2.45 kHz low-pass and 400 Hz high-pass filters. The DIM/TIM test signal is constructed so that a fifth order and a sixth order IMD product fall into the spectrum below 2.45 kHz. These are designated u4 and u5 according to IEC 268-3. The 16-pole 2.45 kHz filter rejects the squarewave fundamental but passes the fifth and sixth order products on to the detector. Only relative units (dB, %, PPM, X/Y) are available in the IMD functions since all IMD standards refer the distortion measurement to the amplitude of one of the signal components. The BW fields (high pass and low pass filter selection) are gray and unavailable in IMD modes since selection of these filters could cause erroneous readings.

### **Reading Meter Function Wow & Flutter Overview**

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The optional wow and flutter (W&F) analyzer is a circuit board mounted in the lower analog analyzer compartment. Its input signal comes from either channel A or channel B as selected on the Analyzer panel buttons just below the Phase meter. Various standards define test frequency, detector characteristics, and measurement bandwidth. The wow and flutter analyzer meets these standards. See page 5-13 for additional information on standards. For measurements of normal (rotationally caused) wow and flutter, the signal is passed through a 2-4 kHz bandpass filter which accommodates the standard 3.0 and 3.15 kHz test signals. For measurements of “high band” or “scrape” flutter,

a test tone of 12.5 kHz is recommended and the signal is fed through a 4 kHz to 20 kHz bandpass filter. Signal from the selected bandpass filter is fed to a frequency modulation (FM) discriminator. The instantaneous output voltage of the FM discriminator is proportional to the instantaneous peak frequency deviation of the test signal. For weighted wow and flutter measurements, the signal passes through a W&F weighting filter; the filter is bypassed for unweighted measurements. The final AC wow and flutter signal is fed to the W&F detector and the measured value displayed on the Analyzer panel. A quasi-peak detector is used in IEC (DIN) specified W&F measurements. An average-responding detector is used in NAB and JIS specified W&F measurements, with a longer time constant selected for JIS than for NAB.

### Wow and flutter control

The wow and flutter measurement function is selected as “Wow & Flutter” in the field which selects functions of the principal (reading) meter on the Analog Analyzer panel. Two other fields change and two additional fields are “grayed out” when the Wow & Flutter function is selected. The field that normally offers detector choices of RMS, AVG, Q-PK, etc., is replaced with a field offering selection of the three rotational wow and flutter standards IEC, NAB, and JIS. Selecting IEC enables a quasi-peak detector with dynamics (“ballistics”) conforming to the IEC and DIN standards. Selecting NAB or JIS enables an average detector with approximate VU ballistics; JIS also enables a software algorithm that imparts a 4-5 second integration time constant to measurements for a heavily damped response. Measurements may take up to 20 seconds to settle in JIS mode.

The field which normally offers selection among plug-in option filters is replaced with a field which permits selection among weighted or unweighted rotational wow and flutter (3 or 3.15 kHz test tones), weighted or unweighted-high band flutter, wide-high band, or scrape-high band flutter. The latter four choices are normally used with a 12.5 kHz test tone. When Weighted is selected, the weighting filter is inserted between the discriminator and detector stages. When Unweighted selection is made, the measurement bandwidth is flat, extending from approximately 0.5 Hz to 200 Hz.

The four selections ending with “High Band” instruct the analyzer to assume a nominal test tone or “carrier” frequency of 12.5 kHz. **Wide-High Band** selects the full bandwidth of the analyzer, extending from 0.5 Hz to approximately 5 kHz (typically -3 dB at 4.5 kHz). The exact response is significantly influenced by the test tone frequency and has been optimized for 12.5 kHz tones. Lower frequency test tones will exhibit degraded bandwidth and aliasing for FM products above half frequency. The **Scrape-High Band** selection chooses a 200 Hz to 5 kHz analysis bandwidth. **Unweighted-High Band** selects a 0.5 Hz to 200 Hz bandwidth allowing relative comparisons between the scrape flutter and/or servo harmonic products versus rotational products. **Weighted-High Band** selects the same weighting filter used with 3 kHz-3.15 kHz test tones, permitting both weighted wow and flutter and scrape flutter measurements with the same test frequency.

When making wow and flutter measurements, the normal input autoranging feature of the instrument should be disabled. Tapes can exhibit momentary dropouts that might trigger ranging, causing a severe transient in the wow and flutter measurement. Fixing the input range is done with the large version of the Analog Analyzer panel displayed, by unchecking the Auto Range checkbox that corresponds to the input channel in use. Then, enter the maximum expected input level into the field next to the checkbox. The analyzer has been designed to handle signals over a 30-dB window, giving ample margin with a fixed input range.

Wow and flutter measurements are typically observed in real time via a bargraph display or on the Reading meter numeric display on the Analog Analyzer panel, or by making a chart-recorder-style “sweep” of wow and flutter versus time. For the time sweep, select Time as the chassis and instrument in the Source-1 Browser. Check the “Show Readings” checkbox, since Time is a reading, and click OK to close the browser. Typical start and stop times might be 5 seconds Start and 25 or 35 seconds Stop. The first five seconds of data is generally not useful since the long time constant detector required for accurate wow measurements will take about five seconds to recover from the initial transient when the wow and flutter option is first enabled by opening a test or changing the Reading meter from another function. For this

reason, limits for Pass/Fail testing are not normally applied to the first several seconds. Sweep settling should be turned off (Algorithm None on the Sweep Settling panel) for wow and flutter measurements. The Compute 2-Sigma utility is sometimes used following a wow and flutter versus time measurement to produce the 2-sigma value. See the Compute Menu section of the Menus chapter for more information.

## Wow and flutter theory of operation

Wow and flutter is the undesirable frequency modulation of an audio signal due to instantaneous speed variations caused by mechanical imperfections in a recording and playback mechanism such as a tape recorder or turntable. Wow and flutter measurements are usually made with a test tape or disk having a pre-recorded tone that is assumed to contain very little residual FM. The reproduced tone is bandpass filtered to limit potential wideband interference and is fed into an FM discriminator. The output of the discriminator is an ac signal whose amplitude is proportional to the instantaneous frequency deviation of the test tone. For most measurements this signal is passed, before detection, through a selectable weighting filter whose peak response is centered at about 4 Hz. The purpose of the weighting filter is to produce numerical results that parallel the human ear's sensitivity to different frequency components of wow and flutter. Unweighted measurements simply bypass the weighting filter.

## Measurement Standards

Four major standards exist for the measurement of wow and flutter: IEC, DIN, NAB, and JIS. All recommend the measurement of weighted frequency modulation of a test tone, but differ in specific test tone frequency, detector type, and/or “ballistics” (the dynamic response of the detector). The IEC and DIN standards are identical and recommend a 3.15 kHz test tone with a quasi-peak detection characteristic. Both NAB and JIS recommend a 3.0 kHz test tone but differ in detector type: NAB specifies an average response (RMS calibrated), JIS specifies “effective” response which is similar to NAB detection but with a much longer integration time constant.

The flat or unweighted bandwidth of the NAB and JIS recommendations extends from 0.5 Hz to 200 Hz, covering the portion

of the spectrum where frequency modulation is normally caused by imperfect rotating components such as idlers, wheels, capstans, pulleys, or motors. IEC/DIN recommends a 0.2 Hz lower cutoff; however, little energy is normally present below 0.5 Hz. Because the settling time of a practical 0.2 Hz system cutoff would be 5 to 10 seconds, the low frequency cutoff of the wow and flutter measurement option is designed to be 0.5 Hz regardless of the standard selection. Weighted measurements are not compromised because the weighting filter response includes the effects of the 0.5 Hz low frequency rolloff.

### Scrape flutter theory of operation

Frequency modulation in tape recorders can also be caused by frictional effects of the tape sliding over guides or the tape heads themselves. This form of imperfection is called “scrape” flutter and is characterized by FM products extending to 5 kHz, but often peaking near 3 kHz. Motors with servo speed regulation can also exhibit FM products substantially above the 200 Hz cutoff of normal unweighted wow and flutter measurements. Higher frequency FM products are perceived more as added noise, “grit”, or “harshness” instead of as frequency modulation.

To measure this form of flutter, it is necessary to use a higher test tone or “carrier” frequency to permit FM discrimination of products to 5 kHz without aliasing. To achieve this extended performance, the wow and flutter analyzer utilizes the HIGH BAND flutter measurement technique developed by Dale Manquen of Altair Electronics, Inc. The recommended HIGH BAND test tone frequency of 12.5 kHz yields the desired measurement bandwidth of 5 kHz on recorders that have frequency response to 18 kHz. The wow and flutter option also permits operation at test tone frequencies down to 10 kHz with some increase in alias errors, for recorders such as consumer VCRs that have more limited high-frequency response.

Scrape flutter is normally measured with average detection-RMS calibrated (NAB) characteristics. Measurement bandwidth in HIGH BAND mode is selectable over four bandwidths: 4 Hz bandpass (WTD), 0.5-200 Hz, 200 Hz-5 kHz, and 0.5 Hz-5 kHz. For a typical professional recorder which incorporates a scrape flutter idler, the

below-200 Hz and above-200 Hz contributions will be approximately equal.

Despite the difference in test tone frequencies, the conventional and HIGH BAND modes yield near-identical readings if both are weighted or both are unweighted. The only change with the HIGH BAND mode is the extended measurement capability. The HIGH BAND mode can therefore be used for all measurements unless either a pre-recorded 3 or 3.15 kHz test tape is being used, or if the recorder has such limited frequency response that only the lower frequency test tone will pass through the machine.

### Spectrum analysis of wow and flutter

With a DSP-based System, FFT spectrum analysis can be performed directly on the 3 kHz or 3.15 kHz tone, or on the wow and flutter discriminator output. Select the Spectrum analyzer/Generator DSP program (fftgen) in the Analyzer field at the top of the Digital Analyzer panel. To acquire signal at the discriminator output, select **A/D** as **Input** and **Anlr Rdg Ratio** (analyzer reading ratio) as **Source** on the Digital Analyzer panel. Choose **Maximum** (16384 samples with full memory) as the **Transform Length** for the greatest frequency resolution, or a shorter transform length for faster operating speed. On the Digital Input-Output (DIO) panel, select a **Rate** of **1k** (1 kHz sampling rate at A/D converter). This combination of settings provides FFT resolution of approximately 0.06 Hz for excellent separation of flutter-frequency components. However, the acquisition time required to fill a 16384-sample buffer at a 1 kHz sample rate is nearly 17 seconds. Selecting a shorter Transform Length, or selecting the 8k Rate on the DIO panel will make tradeoffs for shorter acquisition time at the cost of poorer frequency resolution.

If the flutter is being caused by defective rotating components, the circumference and diameter of the faulty component may be calculated from the measured flutter frequency and knowledge of the tape speed. For example, a dominant flutter frequency of 7.5 Hz at a tape speed of 7.5 inches per second means that the flutter could be caused by a defective idler or capstan with a circumference of 1.00 inches (diameter of 0.318 inches). This is a powerful diagnostic technique for locating defective rotating components.



## **Reading Meter Function 2-Channel Amplitude**

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2-Channel Amplitude function provides simultaneous amplitude measurements of both channels of a stereo device or other two-channel system. The Reading meter displays the amplitude of the channel selected by the Channel select buttons just above the Reading meter display. Only absolute units (Volts, dBu, dBm, dBr, etc.) are available. The Level meter displays the amplitude of the alternate channel. The Reading meter measurement is influenced by the selection of detector, high and low pass filters, and plug-in option filters but the bandpass-bandreject filter is not used. Since the Level meter has an RMS detector and no filters, it is good practice to be sure that the Reading meter RMS detector is used and no filters are selected when in 2-Channel Amplitude function, in order to make the measurement characteristics of the two channels as identical as possible.

## **Reading Meter Function 2-Channel Ratio**

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In the 2-channel ratio function, the reading meter displays the calculated ratio of the Reading meter on the selected channel signal to the Level meter on the opposite meter. Only ratio units (dB, %, X/Y, etc.) are available. The Reading meter measurement is influenced by the specific selection of detector, high and low pass filters, and plug-in option filters but the bandpass-bandreject filter is not used. If the two analyzer channels are connected to the input and output of the device under test, this measurement becomes the voltage gain through the device under test. Since the Reading meter has a wider dynamic range, it should always be connected to the lower-level signal point.

## **Reading Meter Function 2-Channel Bandpass (BP) Amplitude**

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2-Channel BP Amplitude function provides simultaneous amplitude measurements of both channels of a stereo device or other two channel system, using a tunable selective filter in the Reading meter. The Reading meter, effectively in Bandpass function, displays the amplitude of the channel selected by the Channel select buttons just above the Reading meter display. Only absolute units (Volts, dBu, dBm, dBr, etc.) are available. The Level meter displays the amplitude



of the alternate channel. The Reading meter measurement is influenced by the selection of detector, high and low pass filters, and plug-in option filters in addition to the bandpass filter.

### Reading Meter Function Crosstalk

In the Crosstalk function, the reading meter displays the calculated ratio of a frequency-selective Reading meter measurement on the selected channel signal to the Level meter on the opposite meter. The bandpass filter is engaged in the Reading meter in Crosstalk mode, permitting accurate measurements of signals ten to twenty dB below the wide-band noise level. Automatic steering selections of the bandpass filter are as described above in the Bandpass function section. The Reading meter measurement is also influenced by the specific selection of detector, high and low pass filters, and plug-in option filters.

## Reading Meter Ranging Control

The Reading Meter functional block contains switchable-gain amplifiers with up to 1024x gain, depending upon function and signal level. For most applications, it is most convenient (and least error-prone) to allow the Reading Meter Autoranging circuitry to select the proper range. Note that Reading Meter Autoranging is separate and independent from Input Autoranging. For certain applications where measurements must be made on signals whose amplitude constantly changes, such as tone burst signals or program material, it may be necessary to disable autoranging and manually select a range which will handle the expected peak signal amplitude. When Reading Meter Autoranging is selected (Auto box above and to the right of the Reading Meter display is checked), the manual range selection

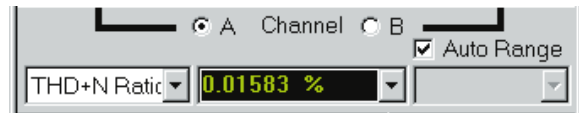


Figure 5-7 Reading Meter Auto-ranged

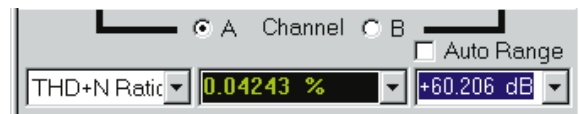



Figure 5-8 Reading Meter Range Fixed at User-Entered Gain Value

field is gray and non-functional. When the Auto box is unchecked to disable autoranging, the range field will display the gain of the range that had been automatically selected. The gain may be expressed in decibels or as a multiplication factor, selectable via the down arrow at the right of the field. A different gain range may be manually selected. The ranges are in 4x (12.04 dB) steps from 4x to 256x (12.04 dB to 48.16 dB) for functions not using the bandpass-bandreject filter, with an additional 1024x (60.21 dB) range in the Reading Meter functions, which use the bandpass-bandreject filter.

To select a new manual gain range, enter the value of gain desired into the range field. A new value may be entered or the present range setting edited from the keyboard after selecting the gain range field. APWIN software will choose the next lowest gain range to avoid overload, and will display the gain range selected when  is pressed after making or editing an entry.

## Detector Reading Rate Control

All four Analog Analyzer meters may make readings at the approximate rates of 4, 8, 16, or 32 readings per second. Slower reading rates provide more integration of noise and other variations and better resolution, while faster rates provide shorter test times. Reading rate is normally managed as a function of signal frequency by an automatic algorithm in APWIN software. It may be sometimes desirable to force a faster reading rate to obtain greater testing speed at some cost in accuracy, or to force a slower rate to obtain greater resolution at a cost in speed. Generally speaking the 32/second rate will provide full specified accuracy for signal frequencies above 65 Hz, with 16/second valid above 30 Hz, 8/second valid above 20 Hz, and 4/second valid above 10 Hz. With complex signals, the frequency to be concerned with is the smallest spacing between individual frequency components if that is less than the lowest absolute frequency.

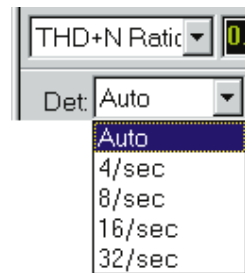


Figure 5-9 Meter Reading Rate (Integration Time) Control. Level, Frequency, Phase Meters are Controlled Simultaneously by this Field

## Detector Control

Five different detector responses are available in the Reading Meter: RMS, Average, Peak, Quasi-Peak (Q-Peak), and Sine Scaled Peak (S-Peak). The Level Meter is always RMS responding.

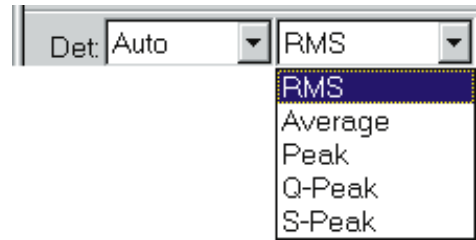


Figure 5-10 Reading Meter Detector Response Selection List

## High Pass Filter Control

The Reading Meter includes selectable three-pole (18 dB/octave) high-pass filters at turnover frequencies of 22 Hz, 100 Hz, and 400 Hz in addition to flat condition. These filters do not affect any of the other meters.

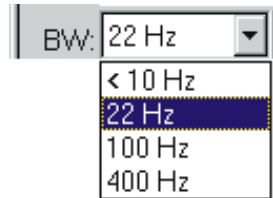


Figure 5-11 Reading Meter High Pass Filter Selections

## Low Pass Filter Control

The Reading Meter includes selectable three-pole (18 dB/octave) low-pass filters at turnover frequencies of 80 kHz and 30 kHz in addition to a 22 kHz selection that cascades the three-pole filter at 30 kHz with a three-pole 22 kHz low-pass. A flat condition is also available. These filters do not affect the Level Meter, but selecting any one of them does connect a single-pole 100 kHz low-pass filter before the Frequency Counter.

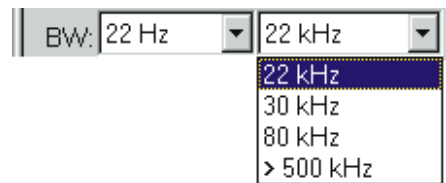


Figure 5-12 Reading Meter Low Pass Filter Selections

## Optional Filter Control

The Reading Meter includes five sockets (slots) for optional plug-in filters, plus front-panel BNC connectors via which an external, user-furnished filter may be selected. These filters do not affect the Level Meter or Frequency Counter.

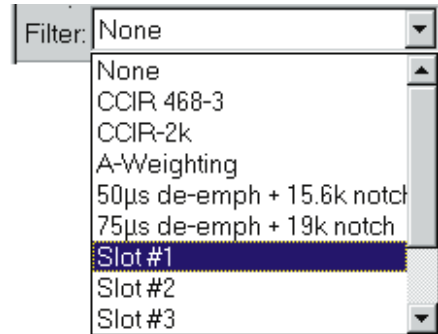


Figure 5-13 Reading Meter Plug-in Option Filter Selection List

A wide range of weighting, bandwidth-limiting, and bandpass filters are available from Audio Precision to be plugged into those sockets. Once installed, a filter slot (socket) may be selected in the list displayed when the down arrow at the right end of the “Filter” control is clicked. The Utilities Filter menu command may also be used to specify what filter is plugged into which socket. Filters may then be selected by name instead of slot (socket) number.

The Utilities Filters menu command may be used to designate which filter is in which socket (slot). Filters may then be selected by name instead of (or in addition to) slot number. The external filter capability, via the front-panel “Ext Filter” BNC connectors, may also be selected by the Filter control. See the Utilities Filters information starting on page 3-42.

The Reading Meter will not function correctly if an empty socket is selected, or the Extern choice is made with no signal path at the front panel BNC connectors.

## BP/BR Steering

The Reading Meter includes a four-pole bandpass or bandreject filter tunable from 10 Hz to 204 kHz. The filter is not used in Amplitude function, 2-Channel functions, and Wow and Flutter function. In the three IMD functions, the filter is configured to improve measurement performance and is not independently steerable. In

Bandpass, Bandreject, THD+N, and Crosstalk functions, the filter frequency may be steered in one of four fashions. The filter center frequency is normally automatically steered (Counter Tuned) to the correct frequency for the particular Reading Meter measurement function and signal frequency in use, but in some cases may need to be Fixed at a specific value, forced to track the analog generator frequency (AGen Track), or forced to track the frequency of the device selected at Source 1 or Source 2 on the Sweep panel (Sweep Track).




Figure 5-14 Reading Meter BP/BR Steering Controls, Fixed Mode

The fixed mode frequency entry field will be gray and non-functional in either Auto or Gen Track steering modes. In Fixed steering mode, this field is active and new entries may be made or existing entries edited from the keyboard, in absolute Hz or a variety of relative frequency units referred to the Reference Frequency value at the bottom of the Analog Analyzer panel.

## Frequency Reference






See also the Frequency Units section of the Units chapter, starting on page 14-6.

When any relative frequency units (anything other than Hz) are selected at the Analog Analyzer Frequency Counter, APWIN software computes the displayed value by use of the Frequency Reference value. A new Frequency Reference value may be entered or the present value edited from the keyboard, followed by pressing the  key.

## dBr Reference

See the AC Amplitude units section starting on page 14-1 for a discussion and definition of dBr units.

When dBr units are selected at the Analog Analyzer Level and/or Reading Meters, APWIN software computes the displayed value by use of the dBr Reference value. The present signal amplitude may be


entered into the dBr Reference field (causing the meter reading to become 0 dBr) by pressing the  function key or selecting Edit Analyzer dBr Reference (  , for Edit analyzer dBr reference) from the menu. A new dBr Reference value may be entered or the present value edited from the keyboard, followed by pressing the  key.

## dBm Reference

See the AC Amplitude units section starting on page 14-1 for a discussion and definition of dBm units.

When dBm units are selected at the Analog Analyzer Level and/or Reading Meters, APWIN software converts the actual measured value

in Volts to dBm by use of the  $P = \frac{V^2}{R}$  and  $dB = 10 \log \frac{P_1}{P_2}$

relationships. The value used for R is the circuit resistance or impedance across which the analyzer input is connected. This is the net parallel value of the analyzer's input impedance with the external circuit impedance. The user must enter this value in the dBm Reference field. A new value may be entered or the present value edited from the keyboard, followed by pressing the  key.


## Watts Reference

See the AC Amplitude units section starting on page 14-1 for a discussion and definition of Watts units.

When Watts units are selected at the Analog Analyzer Level and/or Reading Meters, APWIN software converts the actual measured value

in Volts to Watts by use of the  $P = \frac{V^2}{R}$  relationship. The value used

for R is the circuit resistance across which the analyzer input is connected. This is the net parallel value of the analyzer's input impedance with the external circuit resistance. The user must enter this value in the Watts Reference field. A new value may be entered or the

present value edited from the keyboard, followed by pressing the  key.

## Hardware overview

The analog analyzer consists primarily of two identical input channels and four meters. Each input channel (A and B) has independent peak-sensitive autoranging circuitry and independent control of input impedance. The Phase meter compares the phase of the two input channels, immediately following the input ranging circuitry. A signal multiplexer routes the channel A or B signal to the Level, Frequency, and Reading meters. The Level meter measures signal amplitude on the selected channel. The Frequency meter measures signal frequency on the selected channel. The Level meter, Frequency meter, and Phase meter are all single-function meters with relatively little flexibility. The Reading meter is highly flexible, with a selection of high-pass, low-pass, and optional filters selectable, a tunable bandpass/bandreject filter, and additional autoranging gain

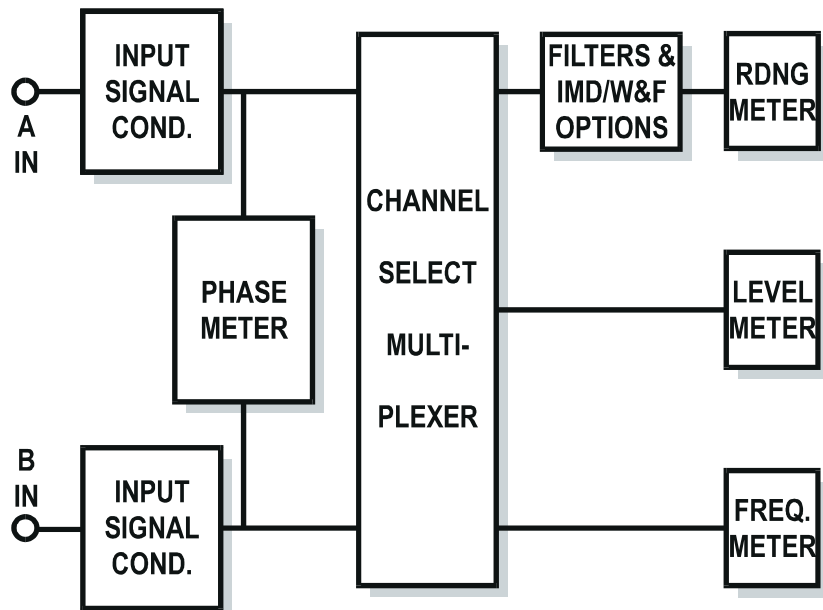


Figure 5-15 System One Analog Analyzer, Simplified Block Diagram

stages. The Reading meter also has a selection of detector responses available. Optional intermodulation distortion (IMD) and wow and flutter (W&F) circuit boards may be added to the analog analyzer. When present, their measurements may be selected as alternate functions of the Reading meter. For more details on the major blocks, place the mouse cursor on the appropriate portion of the block diagram and click.

## Input configuration

Each analog analyzer input channel functionally consists of a balanced high-impedance input, autoranging attenuators and/or gain stages, and balanced-to-unbalanced conversion circuitry. The signal is first presented to AC coupling capacitors and selectable termination impedances. Input termination resistances of either 600 Ohms or 150 Ohms (600 or 300 Ohms with the EURO-Z option) may be independently switched across either input, or either input may be operated terminated only by the 100 kilohm input impedance of the

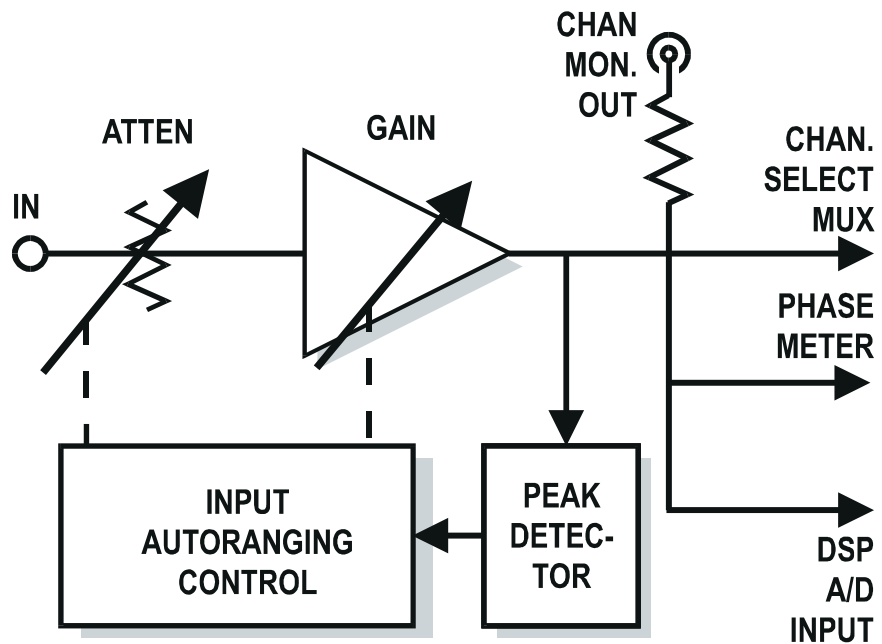


Figure 5-16 System One Analog Analyzer, Simplified Diagram of Input Channel



first stage. Balanced (differential) switched attenuators and amplifiers follow. The attenuators and amplifiers may be fixed on a specific range but typically operate under control of peak-sensitive autoranging circuits. The autoranging detectors are peak-sensitive and independent between channels. Each channel will select the most sensitive range that will not clip the input signal. The autoranging function (or a fixed range properly selected by the user) assures that the signal amplitude from the conditioning section is optimum for accuracy, resolution, and noise performance of the remainder of the analyzer.

Channels A and B are identical except for a BNC connector selectable as the AUX INPUT on Channel A. The shell of the BNC connector is connected to chassis ground. In addition to the Aux selection at Channel A, either channel may be independently connected to front panel input connectors or an analog generator monitor facility. The front panel connectors on each channel are typically an XLR connector and a pair of banana jacks hard-wired in parallel. The plus banana jack connects to pin 2 of the XLR and the minus jack to pin 3. The generator monitor facility consists of an internal cable for each channel, connecting directly to pins 2 and 3 of the corresponding analog generator output connector (A to A, B to B).

## Level Meter

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The Level meter is a single-function, precise, true RMS amplitude-measuring instrument. Unlike the Reading meter, the level meter has no selectable filters. The Level meter has a smaller dynamic range than the Reading meter, with full specified accuracy and resolution extending down only to ten millivolts. The Level meter is useful far below ten millivolts, but with deteriorating accuracy and resolution. Bandwidth and frequency response flatness of the Level meter are somewhat superior to those of the Reading meter.

In all Reading meter functions other than 2-Channel and Crosstalk, the Level and Reading meters (and Frequency counter) are driven by the signal from the selected input channel. In 2-Channel and Crosstalk functions, the Reading meter is driven from the selected input channel while the Level and Frequency meters are driven from the alternate channel.

## Phase Meter

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The Phase meter is connected across the two input channels of the analog analyzer, immediately following the input autoranging circuitry. The phase meter functions by measuring both the period of the reference channel and the time delay between the signal zero crossing on the selected channel versus the reference channel. Time delay to both positive-going and negative-going zero crossings are measured and averaged to avoid errors when measuring non-time-symmetric signals.

Full specified accuracy for the phase meter requires the signal amplitude on both channels to be between ten millivolts and eight Volts. Accuracy is slightly reduced if either channel's amplitude is above eight Volts.

## Frequency Meter

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The System One Frequency meter or counter is a reciprocating, period-averaging design for maximum accuracy and resolution in a minimum measurement interval at audio frequencies. A period-measuring counter uses the input signal to control a gate into counter registers, with the counter clock (time base) signal flowing through the gate into the registers to be counted when the gate is open. Since quartz crystal clock frequencies are much higher than any audio frequency, a period-measuring architecture always provides better resolution in any given measurement interval than a traditional frequency counter architecture. The final count in the registers represents time per cycle rather than cycles per time, however, so the reciprocal ( $F = 1/T$ ) must be computed in order to display the signal frequency rather than its period. A period-averaging counter adds the feature of automatically maximizing resolution for any permissible measurement interval. The first zero crossing of the signal opens the gate and clock pulses begin to be accumulated in the registers. The gate remains open if the permissible measurement time has not been exceeded at the next zero crossing of the signal. A second counter keeps track of the number of signal cycles during which clock pulses are being accumulated. When the permissible measurement interval is finally reached, the gate is then closed at the next signal zero crossing. The number of pulses in the counter register is then divided by the

number of signal cycles for which the gate was open and the reciprocal is calculated.

The minimum signal amplitude for fully-specified operation is 10 millivolts.

## Reading Meter

The Reading meter is the flexible meter in the analog analyzer. In its basic (no options) configuration, it consists of a tunable bandpass/bandreject (BP/BR) filter followed by three selectable fixed low-pass filters, five sockets for plug-in option filters, connections to an external user-supplied filter, and three selectable fixed high-pass filters. The filter section output feeds the inputs of the reading meter detectors and the READING MONITOR OUTPUT BNC connector. When the intermodulation distortion (IMD) option is present and an IMD function is selected, the signal is processed through the IMD circuit board before being presented to the BP/BR section input. When the wow and flutter (W&F) option is present, the signal is processed by the W&F circuitry including an extended low-frequency detector on the W&F board, then presented directly to the voltage-to-frequency converter of the Reading meter. For block diagrams of the IMD, W&F, and filters sections of the

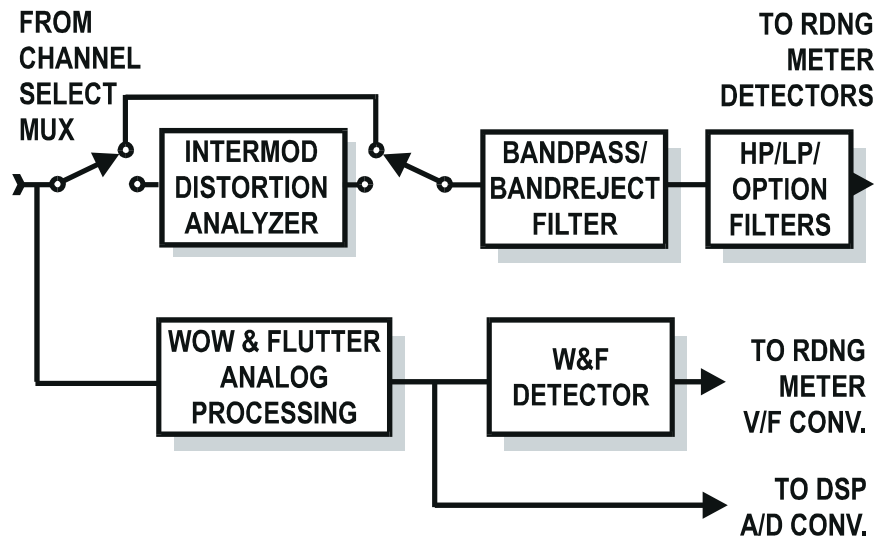


Figure 5-17 System One Analog Analyzer, Simplified Block Diagram of Reading

meter, place the mouse cursor on the appropriate section of the block diagram and click.

### BP/BR Filter

The Reading meter filter section consists of a tunable bandpass/bandreject filter, an autoranging gain stage, and a selection of three each fixed built-in high-pass and low-pass filters plus sockets for optional plug-in filters and provisions for connections to an external, user-furnished filter. The four pole bandreject filter has excellent ability to attenuate a band of frequencies, including close-in sidebands as are caused by flutter, while still attenuating the second harmonic by very small amounts. Maximum performance is obtained in THD+N function rather than Bandreject function of the Reading meter, since this engages a servo system that continuously re-tunes the filter for maximum rejection. In Bandreject function, both sections of the filter are nominally tuned to the same frequency but typically attenuate the signal less than in THD+N function due to filter component tolerances. Signal to the fixed filter section passes through the bandpass-bandreject (BP/BR) filter block, whether or not the

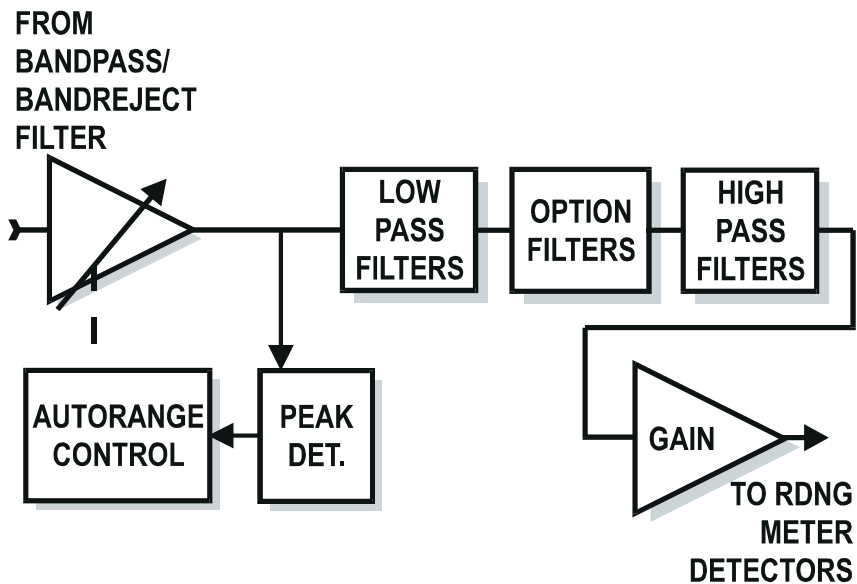


Figure 5-18 System One Analog Analyzer, Simplified Block Diagram of Fixed Filter Section

BP/BR filter is actually selected at the time.

Signal amplitude from the BP/BR filter block is adjusted by a switchable gain amplifier before being applied to the fixed filters. The gain of this amplifier is normally controlled by a peak-sensitive autoranging circuit which operates to assure that the signal amplitude is as high as possible for best signal-to-noise ratio while still not overloading the filters and detectors. The autoranging function may be disabled and a fixed gain range selected by the user. The gain range selected when autoranging is determined by signal amplitude before filtering. A high-amplitude signal component at an extreme frequency can cause a low gain range to be selected even though that signal component is then filtered off by one of the following filters. Only one of the high-pass, one of the low-pass, and one of the option filter sockets (or external) may be selected at the same time. The output of the filter section connects to the detector section of the READING meter.

## Reading Meter Detector

See also the dBm vs. dBu concepts discussion on page 14-4 and the Detector Concepts section on page 5-32.

The signal output from the reading meter filter section is presented to the inputs of three detectors—true root-mean-square (RMS), average-reading RMS-calibrated (Avg), and the peak/quasi-peak detector (PK /QPK). Each detector produces a DC output amplitude proportional to that parameter of the input AC signal. The DC signal from the selected detector is connected to a voltage-to-frequency (V/F) converter, whose output is a pulse train at a rate proportional to the DC input voltage. The pulse train from the V/F converter feeds a counter. There are four selections of gate time into the counter, nominally 1/32, 1/16, 1/8, or 1/4 second. At any given pulse rate (corresponding to a DC voltage from the detector), doubling the gate time doubles the number of pulses accumulated into the counter and therefore doubles the resolution of the reading. The value sent from the counter to computer for display or graphing is corrected for the gate time used. Thus, the final value is accurate regardless of gate time, but the resolution of the reading and integration of short-term variations in the signal increase with longer gate times.

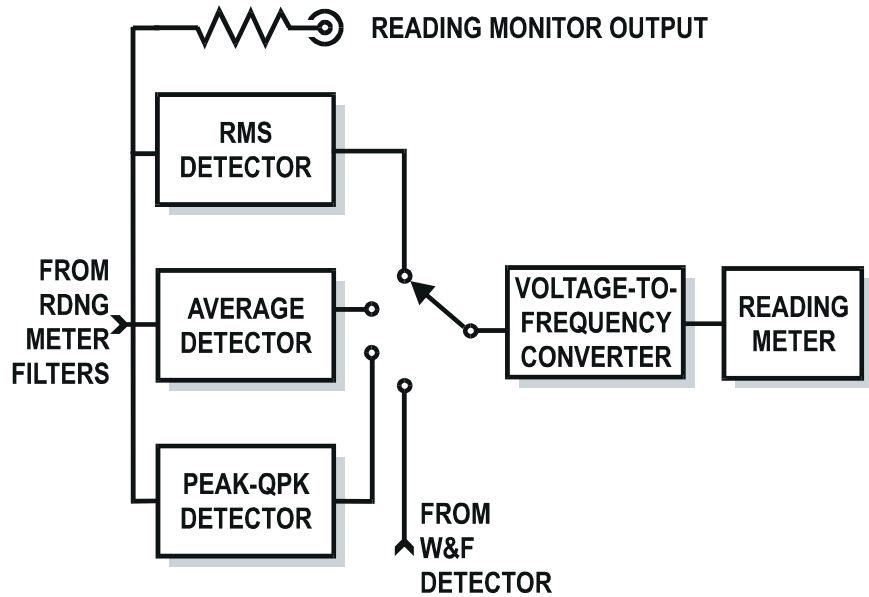


Figure 5-19 System One Analog Analyzer, Simplified Diagram of Detector Section

Since the W&F option requires response below 1 Hz, a separate detector is incorporated into the W&F circuit board. The output of this W&F detector connects to the V/F (voltage to frequency) converter input for W&F function.

The input signal to the three detectors is the final AC signal in the READING meter. For analog analyzer measurements, this comes from the output of the Reading Meter Filters section. It is also fed to the READING MONITOR OUTPUT BNC connector for oscilloscope monitoring. If DSP modules are present, this same signal is the circuit point available as input to the A/D converters of the DSP module. In W&F function, the READING MONITOR OUTPUT connector and reading meter selection to the DSP A/D converters are fed from the same signal that drives the W&F detector.

### Autorange vs. Fixed Range Concepts

A competent audio analyzer must measure signals across a wide dynamic range. Modern high-powered audio amplifiers produce output voltages on the order of 100 Volts, while output levels from

microphones during acoustic measurements may be hundreds of microvolts or less. No precision level detector can function accurately across such a high dynamic range. Therefore, an analyzer must contain circuitry to attenuate high level signals or amplify low level signals to bring them to the optimum level for precision detection and indication.

Older audio analyzers and lower-cost instruments use manual range switching. The operator is responsible for selecting an input range that brings the signal into the proper range. Such instruments normally used analog meters, with the proper range selection indicated by the meter pointer being above 1/3 full scale but not beyond full scale. Sophisticated modern analyzers normally manage range selection automatically for maximum accuracy and resolution without operator intervention.

The System One analyzer has automatic range selection near the inputs and again in the “Reading” meter circuitry. Since there are two input channels, the input range selection is duplicated in both channels. See the Analyzer input signal conditioning topic for a block diagram of the analyzer including input ranging; see the Reading meter filters topic for a diagram of the Reading meter, showing the location of the ranging circuitry there.

In all three locations, the autoranging function is controlled by peak-sensitive detectors so that high crest factor signals will cause the proper range to be chosen. Instruments with RMS or average responding detectors controlling autoranging will select an excessively-sensitive range with high crest factor signals, causing clipping within the instrument with resulting errors.

For certain applications, it is desirable or necessary to fix ranges. Low duty cycle signals, such as a tone burst repeated at the rate of a few bursts per second, will not be reliably measured by autoranging circuits and require the operator to set a fixed range based on a steady sinewave of the same amplitude. Program material such as music and voice cannot be successfully handled by autoranging, so the user must make an independent determination of the expected peak amplitudes and select a fixed range that will handle them. Random noise and certain other signals may have extremely high crest factors which

exceed the maximum crest factor rating of the detectors in the autorange circuits, so it may be prudent to fix the analyzer range on the next less-sensitive range above the one chosen automatically. Note that for measurements with the Level meter, Phase meter, Frequency counter, or DSP-based measurements where the signal is obtained immediately following the input ranging circuits, only those input ranges must be fixed. For all measurements made by the Reading meter, it may also be necessary to fix the range in the Reading meter circuitry.

When a fixed range is selected, it represents an engineering compromise between two problems. If too sensitive a range is chosen, signal peaks may exceed the linear signal-handling capability of the analyzer and clipping with measurement errors will result. If the fixed range is selected too conservatively and the signal varies downwards, increasing measurement errors will result due to the limited dynamic range of the precision detectors in the Level and Reading meters. The full dynamic range of these detectors is on the order of 50 dB, with signals more than 50 dB below full scale not being measured at all. The full specified accuracy of the detectors is met over a much narrower dynamic range, with increasing errors resulting from signal levels more than about 12 dB below full scale.

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## Detector Concepts

The detector is an audio measurement instrument is the circuit block which converts the analog audio signal to an varying DC voltage which is some function of the amplitude of the analog audio signal. The DC voltage is typically then converted in some manner to numbers that may be graphed or otherwise used by a computer or digital display.

There are several detector functions that produce DC voltage output proportional to different parameters of the audio signal's amplitude. They include true RMS, average responding-RMS calibrated, peak, and quasi-peak. System One also includes an S-peak (Sinewave scaled peak) selection which is a combination of peak detection hardware and computer scaling of the result.



True RMS detectors have a DC voltage output proportional to the RMS (root-mean-square) value of the audio signal input. This is proportional to the heating value (power, assuming constant load impedance) of the signal. For most applications, the true RMS detector is the best way to compare the amplitude of non-sinusoidal signals, especially those with significant amounts of noise.

Average responding RMS calibrated (usually just called Average) produce an output DC voltage proportional to the average value of the absolute value of the input signal, which is the same as saying they respond to the average value of a full-wave-rectified version of the input signal. There is little to recommend average responding detectors at this time, but for many years they were much less expensive to implement and much more robust than true RMS detectors. Therefore, many modern analyzer still include average-responding detectors for purposes of correlating to measurements made with older instruments, or to comply with older specifications such as Dolby CCIR-ARM or the SINAD method of specifying communications receiver sensitivity, both of which were written in the days when only average-responding detectors were readily available. Average responding detectors are normally calibrated with a sinusoidal input signal and are adjusted to read the same as a true RMS signal on that waveform. Therefore, they will provide identical measurements on single sinewaves to an RMS detector, but will differ from an RMS detector for all complex waveforms such as noise and multitone signals.

A Peak detector provides a DC output voltage proportional to the peak voltage of the input signal. Since clipping effects on peak excursions set the maximum signal handling capabilities of many audio devices, a peak detector is useful. Peak detectors are also used for measurements of some types of noise signals such as impulses or other narrow, low duty factor pulses. The theoretically ideal peak detector would have an infinitesimally-short attack time and an infinite decay time, which is not practical in an instrument. Practical peak detectors have some minimum signal repetition period at which they meet specified accuracy, with decay of the previous impulse detracting from accuracy beyond that value.

A Quasi-peak detector is the specific modification of a true peak detector required to meet the specifications of CCIR Recommendation 468. This specification grew from the need to measure the effects of impulsive noise which commonly coupled into broadcast transmission circuits when they passed through a telephone switching center in the days when relay switching was common. CCIR 468 quasi-peak detectors are normally used in conjunction with the CCIR 468 weighting filter.

System One also includes an S-Peak (Sinewave scaled peak) detector. The peak detector hardware is used, with APWIN software multiplying the peak detector output by 0.707 before displaying it. Thus, the S-Peak detector is calibrated to show the amplitude of a sinewave that would have the same peak value.

## DSP Introduction

System One DSP versions always include a main DSP module and may also contain either an additional memory module or a digital input/output module with additional memory. All functions of the DSP modules are defined by selecting a Digital Analyzer function, which then downloads the selected DSP program from the computer; no functions are available without loading a program.

The DSP programs differ from one another by the type of analysis capabilities offered. Some of the DSP programs also contain signal generation capability. Certain DSP programs, such as HARMONIC, are designed only for analog domain signals. Others, such as GENANLR and BITTEST, are designed only for use in the digital domain and thus are useful only when the digital i/o module is present. The remaining programs are useful with both analog and digital domain signals.

All System One DSP versions include dual A/D converters to permit analog domain signals to be measured by DSP techniques. Analog signals may be fed to System One's front panel balanced ac-coupled inputs and gain/attenuator stages before connection to these A/D converters, for convenient operation across a wide dynamic range. The Reading meter signal, following all analog hardware processing such as the bandreject filter, wow and flutter discriminator, or IMD detection and filtering, may also be fed to one of the A/D converters. Alternately, BNC connectors provide unbalanced, dc-coupled, 2 Volts rms full scale inputs to both A/D converters.

All DSP-based instruments include a single D/A converter to provide analog domain output when any of the signal-generating DSP programs are in use. This D/A output signal may be routed through the analog generator amplitude control circuitry, power amplifier, and output transformer via the DGen waveform selection. An unbalanced, dc-coupled, 2 Volts rms full scale output from the D/A converter is also available at a BNC connector.

The digital i/o module in System One Dual Domain units (SYS-300 series) includes input and output in the AES/EBU format and SPDIF/EIAJ (consumer) format at front-panel XLR, coaxial (phono connector labeled SPDIF), and optical connectors, and in parallel and general-purpose serial formats at rear-panel connectors. Via the external SIA-322 Serial Interface Adapter accessory, the parallel i/o can be converted to a wide variety of serial formats for connection to digital devices at the component and circuit board level.

The DSP module can operate at sample rates as high as 192 ksamples to provide 80 kHz bandwidth for analog domain measurements. For digital signal generation and measurements in the AES/EBU and consumer formats, the three standard rates of 48, 44.1, and 32 ksamples are provided.

## Digital I/O Panel

The APWIN Digital I/O Panel controls input and output formats, output sample rate, output word width (resolution), output dither, displays sync lock indicators for the rear-panel digital reference input and digital signal input, and has a button to display the Status Bits panel. The Digital I/O panel can be brought to the screen by:

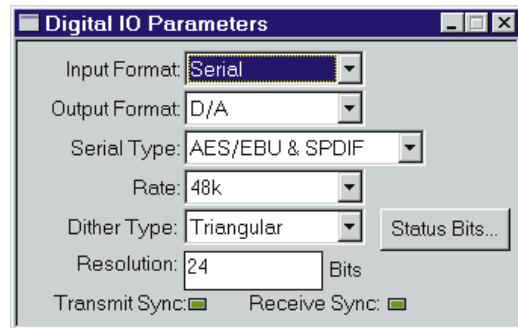


Figure 6-1 Digital I/O Panel

clicking on the Digital I/O icon  on the toolbar

or

selecting Panels Digital I/O from the menu bar

or

from the keyboard, pressing    (for Panels digiTal i/o)

or

from the keyboard, typing  

## Input Format

---

The System One digital input format control provides selections of Serial, Parallel, and DGen. “Serial” must be selected on the DIO panel to accept digital input from the AES/EBU (XLR), SPDIF (phono), or optical front panel connectors or from the rear panel general-purpose serial connector. The Serial Type field below the input format field must also be properly set as described below. The “Parallel” selection accepts signal from the rear panel parallel input connector. “DGen” internally routes the imbedded digital audio signal from the digital generator to the digital analyzer. The digital generator ON/MUTE switch or channel selection switches do not affect the DGen selection. The DGen selection does not re-route Status Bit monitoring, which still comes from the external device connected to the XLR, phono, or optical connectors. To monitor the internally-generated Status Bits, an external cable is required from digital generator output to digital analyzer input.

## Output Format

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The System One digital output format control provides selections of Serial, Parallel, and D/A. Serial must be selected to provide digital output to the AES/EBU (XLR), SPDIF (phono), or optical front panel output connectors or to the rear panel general-purpose serial output connector. The Serial Type field must also be properly set as described in the following section. The Parallel selection feeds the rear panel parallel output connector. D/A selects an internal single-channel D/A converter whose output drives both the DGen waveform selection of the analog generator and the DSP D/A Output BNC connector on the rear of SYS-322 models and the front of SYS-222 models.

## Serial Type

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The Serial Type field selects both input and output connectors and formats. It must be set to “AES/EBU & SPDIF” to enable the front-panel XLR, phono, or optical input and output connectors. It

must be set to “GP Serial” to enable the rear-panel 15-pin D-sub input-output connector.

## Sample rate

---

The “Rate” field on the System One Digital I/O Panel permits selection of the digital sample rate. This value becomes the sample rate of the A/D converters in the DSP module. If the DSP program presently in use contains signal generation capability and if the DSP is not locked to an external digital reference signal, the Rate field directly controls the digital generator output sample rate and the clock rate to the D/A converter. When a digital input signal is being analyzed, the DSP always locks to the sample rate of the incoming signal. However, System One DSP hardware does not measure the incoming sample rate and APWIN software assumes that the incoming sample rate is the same as the value in the Rate field. Therefore, the frequency of the imbedded audio signal input will be measured and displayed with an error equal to the ratio of the actual incoming sample rate to the value in the Rate field. All rates are derived from the internal crystal clock. If an acceptable digital reference signal is provided at the rear panel sync input connector as indicated by the Transmit Sync indicator lighting, the generator output sample rate will be locked to the sync input rate rather than to the value in the Rate field.

## Dither Type

---

Dither of the digital output signal may be turned off or selected from three choices of probability distribution function and frequency spectrum in the Dither Type field of the DIO panel. The selections are:

**Triangular** probability function dither has no noise modulation effect but produces a slightly worse output signal to noise ratio. This is normally the preferred choice.

**Rectangular** probability function dither provides the best signal to noise, but suffers from modulation noise effects.

**Shaped** dither is triangular probability distribution noise with a rising 6 dB/octave slope with zero dB effect at 1/2 the sample rate, thus placing most of the dither power at higher frequencies where some falls

out of band of most devices and where the human hearing system is less sensitive.

**None** selection turns dither off.

The dither amplitude is automatically determined by the value entered into the Resolution field. Rectangular probability dither amplitude is set at 1/2 LSB and triangular dither amplitude is set at one LSB for the Resolution (word width) value presently set. See the Dither section later in this chapter for more information.

### **Resolution (word width)**

---

The value in the Resolution field determines the output word width. Digital output signals are internally generated at a full 24 bit resolution in System One, then rounded to the bit value specified in this field. If dither is turned on, the dither will also be added at the LSB (least significant bit) level of the specified resolution.

### **Transmit sync indicator**

---

The Transmit Sync indicator of System One shows whether an acceptable signal has been connected to the sync input on the rear of the chassis. When this indicator is “lighted”, the digital output sample rate will be locked to the sync input frequency rather than being controlled by the Rate field.

### **Receive sync indicator**

---

The Receive Sync indicator will “light” if an acceptable digital signal is connected to the input connector selected in the Input Format field.

### **Button for Status Bits Panel**

---

Clicking on this button causes the Status Bits panel to display.

## **AES/EBU & Consumer Interfaces**

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The most popular format for digital audio is commonly called the AES/EBU format. It is defined in the AES3 standard published by the Audio Engineering Society. Consumer digital audio equipment

commonly is encoded according to standards often called SPDIF (for Sony-Philips Digital Interface) or simply the “Consumer” standard. Defining documents include IEC958 and a document from the EIAJ. The principal differences between the Professional AES/EBU standard and the Consumer standard are in the definitions of the Channel Status Bytes and in the fact that the Consumer standard is often implemented as an unbalanced (coaxial) signal of about one Volt p-p amplitude, where the AES/EBU format is typically a balanced signal on XLR connectors, with about five Volts p-p amplitude.

## Status Bits Panel

The AES/EBU professional and the consumer standards (sometimes known as SPDIF or EIAJ) for serial audio transmission include “administrative” data bits in addition to the bits representing the audio signal and synchronization bits. One

of the most important administrative bits is the Channel Status bit (C bit). Status bits for both the A and B subframes (channels) are organized across blocks of 192 frames into 24 status bytes. System One digital i/o units always send the same status bits on Channels A and B and display received status bits only from Channel A. With most digital devices, status bits are identical on the two channels. The use and meaning of these status bytes is generally different between professional and consumer standards. Byte 0 defines, among other things, whether the bytes are to be interpreted according to the professional or consumer standard. The Status Bits panel permits setting the desired status information to be transmitted along with the audio data and displaying the incoming status bit information. Status information may be set and displayed in both high-level English and as hexadecimal numbers. The high-level and hex settings and displays are both continuously functional, regardless of which format is used for output control.

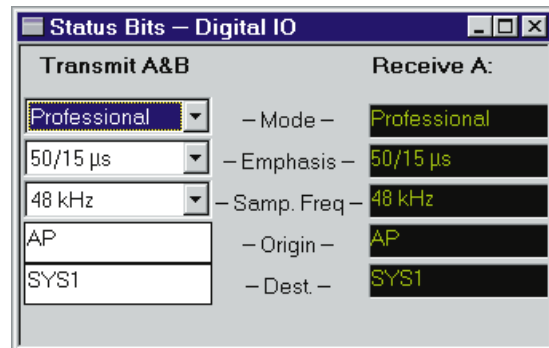


Figure 6-2 Status Bits Panel, Small Version



The System One Status Bits panel can be brought to the screen by:

clicking on the Status Bits icon  on the Panels Toolbar

or

selecting Panels Status Bits from the menu bar

or


clicking on the Status Bits . . . button on the Digital I/O Panel

or

from the keyboard, pressing    (for Panels statUs bits)

or

from the keyboard, pressing   (for Bits/Bytes)

The Status Bits Panel may be displayed in small and large sizes. The small size version of the panel includes the Mode, Copyright, Emphasis, Sample Rate, and Category setting and reading fields when Consumer format status bytes are being transmitted. When Professional status bytes are transmitted, the small version Status Bits panel includes settings and reading fields for Mode, Emphasis, Sample Freq, Origin, and Destination. The large version of the panel includes settings and readings fields for all defined features in the Professional or Consumer standard. Clicking on the panel icon on the Panels Toolbar brings up the small version panel. Holding down the  key while clicking on the toolbar icon brings up the large version. To switch between the small and large panels, double-click anywhere in the Title Bar.

The Status Bits panel in the digital i/o versions of System One will not function completely unless a Digital Analyzer (DSP program) with digital input and output has been loaded and digital i/o selected on the Digital I/O panel. Thus, the status bits feature is not available when HARMONIC is loaded. Only the Received Status Bits features will function with FFTSLIDE, and then only if digital input is selected. Both Transmit and Receive sections can function with any of the other

**Status Bits – Digital IO**

**Transmit A&B**

Consumer     Professional  
 Normal     Non-Audio

Emphasis: 50/15  $\mu$ s

Freq Mode: Locked

Sample Freq: 48 kHz

Channel Mode: Not Indicated

User Bits: None

Auxiliary Bits: 20-bit not define

Audio Word Len: Not indicated

Ref Signal: Not a ref. signal

Origin: AP    Dest: SYS1

Time-of-day: 0

Reliability Flags:  0-5     14-17  
 (set = unreliable)  6-13     18-21  
 CRC Enable

**Receive A:**

Mode: Professional

Audio Mode: Normal

Emphasis: 50/15  $\mu$ s

Freq Mode: Locked

Sample Freq: 48 kHz

Channel Mode: Not Indicated

User Bits: None

Auxiliary Bits: 20-bit not defined

Audio Word Len: Not indicated

Ref Signal: Not a ref. signal

Origin: AP

Dest: SYS1

Local Address: 7270609

Time-of-day: 0

Reliability Flags:  0-5     14-17  
 (On = unreliable)  6-13     18-21  
 CRC Valid:

	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23
Transmit A:	8D	00	00	00	00	00	41	50	20	20	53	59	53	31	00	00	00	00	00	00	00	00	00	00
Receive A:	8D	00	00	00	00	00	41	50	20	20	53	59	53	31	D1	F0	6E	00	00	00	00	00	00	38

Figure 6-3 Status Bits Panel, Large Version

DSP programs if digital input and output are selected on the Digital I/O panel.

The Status Bits panel is organized into three sections: Transmitted status bits (high level display), Received status bits (high level display), and Hexadecimal display transmitted and received status bits.

### Transmitting Status Bits

The Professional versus Consumer setting is the most fundamental, since the definition of most of the remaining Channel Status codes

(and thus the layout of the settings and display fields on the Status Bits panel) changes as a result.

The remainder of the high-level Transmit control section of the panel will be re-drawn when the Consumer-Professional selection is changed. Any of the remaining transmitted status bytes may then be changed by standard Windows control techniques which include buttons, fields which display a list of choices via the down arrow at the right, check boxes, and free-form entry fields.

## Received Status Bytes

---

The center section of the System One Status Bits panel displays, in high-level English terminology, the received status bytes from Channel A. The displays are up-dated approximately four times per second. If the received status byte code changes between Consumer and Professional, the remainder of the receive section of the panel will be re-drawn since most definitions are different between the two standards.


## Hexadecimal Transmit and Receive

---

The bottom section of the System One Status Bits panel consists of hexadecimal transmit and receive registers. The receive registers display the incoming subframe A status bytes only. The hexadecimal transmit indicator will indicate the codes being transmitted (on both subframes A and B simultaneously), whether they were set by the high level fields above or are set in the hexadecimal fields themselves. The values in the transmit register may be changed by over-typing the existing values and pressing Enter. The controls in the high-level Transmit section above will then change to follow changes in the hex settings. To change the hexadecimal Transmit register settings:

**Mouse:** Click on the transmit field. A vertical character cursor will appear. Use the horizontal arrow keys to move the cursor to the desired location and overwrite the sections of the entry to be changed. Press Enter to make the change.

**Keyboard:** Shift the focus to the Status Bits panel, if necessary, with the **CTRL** **TAB** keys. Select the Transmit field with the **TAB** key (moves selection downwards) or **SHIFT** **TAB** keys (moves selection upwards). Use the horizontal arrow keys to move the cursor to the

desired location and overwrite the sections of the entry to be changed. Press  to make the change.

## Digital Analyzer Panel

A variety of digital analysis (and in many cases, digital signal generation) programs are available for DSP versions of System One. Some are useful for digital domain signals only, some for analog signals only (via built-in A/D converters), and some may be used with signals of both domains. To bring the Digital Analyzer panel to the screen:

Mouse: click on the Digital Analyzer icon  on the Panels Toolbar, or select Panels Digital Analyzer via the menu bar

or

from the keyboard, press    (for Panels digital analyZer)

or

Press  .

The System One Digital Analyzer panel is virtually blank until an analysis function (downloadable DSP program) is selected. To load a DSP program (select an analysis function), click on the down arrow at the right of the Analyzer control and select one of the function description-program names which are displayed. The Digital Analyzer panel will then be configured for the features of the particular DSP program selected and can then be displayed in both small and large versions.

APWIN and System One DSP versions are shipped with nine DSP programs. Three of these programs are “real time” DSP programs that perform selected analysis functions on a continuous basis, and may also provide signal generation. Their measurement results appear on the panels in the same way as analog hardware analyzer measurements and may be graphed. Their key settings may be swept. One of these real time programs, which is a selective voltmeter (HARMONIC) operates only with analog domain signals. See the

HARMONIC chapter starting on page 7-1 for details. The other two real time programs, GENANLR and BITTEST, operate only with digital domain signals. GENANLR emulates most standard audio measurement techniques such as level and frequency measurements, selective amplitude measurements, THD+N, and weighted and unweighted noise measurements in the digital domain. See the GENANLR chapter beginning on page 8-1 for details. BITTEST is designed for bit error measurements of the imbedded audio path with a variety of waveforms. See the BITTEST chapter starting on page 9-1 for details.

The remaining six DSP programs are FFT (Fast Fourier Transform)-based programs. These programs are FFTSLIDE, FFTGEN, FASTTEST, FASTTRIG, CODEC, and MLS. The FFT Overview chapter beginning on page 10-1 provides information on common characteristics of all these programs. The two general-purpose programs, FFTSLIDE and FFTGEN, are described in the Spectrum Analyzers chapter. The three programs designed for multitone generation and analysis are described in the Multitone Analyzers chapter. MLS is discussed in the MLS chapter, starting on page 13-1.

## DSP Hardware and Architecture

The Digital Signal Processor modules include two or three DSP chips, memory, 16-bit A/D and D/A converters (2 Volts RMS full scale), and supporting circuitry. Memory may be of two sizes, with the memory size controlling the duration of signal that can be acquired and thus the ultimate FFT resolution available. A standard SYS-200 series contains 8 k samples of memory per channel and permits FFT spectrum analysis of up to 2k lines. Adding the MEM option to an SYS-200 or moving to the SYS-300 series increases memory to 30,720 samples per channel and permits FFT spectrum analysis up to 8k lines (bins). For the various types of enhanced measurements on signals in the analog domain, analog signals are converted into the digital domain by two A/D converters. All remaining processing and measurement is then digital. The points at which the analog signal may be acquired include (in effect) the Channel A, Channel B, and Reading monitor connectors of the analyzer, the MONITOR connector

of the generator, and two additional BNC input connectors dedicated to the DSP.

The SYS-300 (Dual Domain) series can additionally acquire and generate 24-bit digital audio signals in the AES/EBU and SPDIF serial formats at front panel connectors, and general-purpose serial and parallel-format signals at connectors at the rear of the enclosure. Note that the BNC connectors labeled ANALOG INPUTS A and B are unbalanced, DC-coupled inputs of 2 Volts RMS full scale. This allows measurements below the low frequency limit of the normal analog analyzer AC coupled inputs.

### **Rate vs. Bandwidth**

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The Rate choices available in the hardware are 1 kHz, 8 kHz, 32 kHz, 44.1 kHz, 48 kHz, 176.4 kHz, and 192 kHz. Not all rates are available in all DSP programs. The AES/EBU, SPDIF/EIAJ, and optical digital input and/or output capability functions only with the 32 kHz, 44.1 kHz, and 48 kHz rates. The remaining rates, if available in a particular program, function through the parallel or general purpose serial ports and as sampling rates of the A/D converters for acquiring analog signals.

The bandwidth available with any rate cannot exceed half the sample rate, as originally shown by Nyquist. In practice, the useful bandwidth is somewhat less than half the sample rate. System One's DSP module, when acquiring analog signals, actually functions with the A/D converters always operating at a 192 kHz or 176.4 kHz sample rate and with an anti-alias low-pass filter bandwidth from zero to 80 kHz. When a lower sample rate is selected, a second DSP chip functions as a decimator, effectively scaling down both the sample rate and the anti-alias filter corner frequency to lower values. Typical -3 dB points are 375 Hz at the 1 kHz rate, 3.94 kHz at the 8 kHz rate, 15.7 kHz at the 32 kHz rate, 21.6 kHz at the 44.1 kHz rate, 23.5 kHz at the 48 kHz rate, and 85.3 kHz at the 176.4 and 192 kHz rates. Linear phase acquisition of analog signals is only available at sample rates of 48 kHz and lower. The 1 kHz sample rate is intended only for acquisition and spectral analysis of very low frequency signals such as wow and flutter.

The bandwidth of the signal being acquired may be further limited by the processing selected in the System One analog hardware. For example, when acquiring through the Reading signal path, the lowpass, highpass, tunable bandpass, and any optional filters selected will all affect the signal. When measuring Intermodulation signals from the Reading meter, the measurement bandwidth will be limited by the IMD mode selected.

## DSP Input Signal Selection

Both System One + DSP (SYS-200 series) and System One Dual Domain (SYS-300 series) can acquire and perform analysis of analog signals. System One Dual Domain can furthermore acquire digital audio signals directly in the digital domain.

The DSP is a full two-channel device, with two digital input channels and two separate A/D converters and anti-alias filters for two analog signals. Note that the DSP channels have been named CH-1 and CH-2, to avoid confusion with the analog analyzer's A and B input channels.

In System One Dual Domain models (SYS-300 series), if Serial, Parallel, or DGen are chosen as the source on the Input Format line, the choices at CH-1 and CH-2 become **A**, **B**, and **None**. The parallel input connector is located on the instrument rear panel. Serial Input Format includes two serial data formats, **AES/EBU & SPDIF** or **GP Serial**. AES/EBU & SPDIF refer to the front panel female XLR connector, the front-panel RCA phono type connector, and on later versions (after serial number SYS1-32214) a front panel optical input connector. All are functionally in parallel. GP Serial refers to a general-purpose serial format at a rear-panel 15-pin D-sub connector. On units before S/N SYS1-32214, the XLR (labeled AES/EBU) and phono (labeled SPDIF) inputs are effectively in parallel and cables should not be connected to both at the same time. On units S/N SYS1-32214 and later, the optical, phono, and XLR inputs may all have cables connected at the same time. System One Dual Domain will sense the presence of signal and select one of these inputs according to the following priority if two or more have signal present simultaneously: XLR first, optical second, phono last.

## SIA-322 Serial Interface Adapter

An accessory unit, the SIA-322, is available to greatly simplify connection of serial output digital devices other than AES/EBU and SPDIF/EIAJ to System One Dual Domain. The SIA-322 permits selection of a wide variety of serial formats of both one and two channels with from 8 to 24 bits to be interfaced simply by setting front-panel switches on the SIA-322. MSB-first and LSB-first data can be used with a variety of word-strobe positions, in either TTL or CMOS logic families. Contact Audio Precision or your Audio Precision International Distributor for more information on the SIA-322.

## AES/EBU Input Termination

With Dual Domain units before serial number SYS1-32214, the XLR and phono input terminating impedance is 240 Ohms as called for in the original AES/EBU standard. If an optical panel option retrofit has been installed, these older units will function as described in the following paragraph.

On Dual Domain units with serial number SYS1-32214 and higher, a push-push switch between the XLR (labeled AES/EBU) connectors permits selection of a 110 Ohm termination or high impedance (bridging) on the XLR and phono inputs, as specified in the current revision of the AES/EBU standard. The 110 Ohm termination (switch pushed in) should be selected when the System One Dual Domain input is the only receiving device on an AES/EBU cable. If the Dual Domain unit is to be connected across a cable between an AES/EBU transmitter and receiver where the receiver already properly terminates the line, the switch should be in the out (bridging) position.

## DSP Output Operation

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Some of the .DSP programs include signal generation capability. Signals generated by DSP techniques can be converted to the analog domain. In a Dual Domain unit, they can also be furnished directly in the digital domain. Some DSP programs may use the analog output capability simultaneously with the digital output. Consult the chapters for the DSP program being used for specific information.



## Analog Outputs

When the Output Format D/A selection is made on the DIO panel, analog output from the DSP-implemented generator is available via an internal 16-bit D/A converter. Distortion of the D/A converter is typical of 16-bit converters at about -85 to -90 dB, considerably inferior to the System One analog generator. Either the A or B output “buttons” (or both) in the digital generator section at the bottom of the Digital Analyzer panel must be “ON” when routing digitally-generated signals through the D/A converter. This analog output signal may be obtained at the ANALOG OUTPUT D/A connector of the DSP module, ground-referenced, with a typical two Volt RMS amplitude at 0 dBFS (100% FS) amplitude. Amplitude control at this connector is available only by varying the DGen Ampl field on the DSP panel. Thus, fewer bits of the output D/A are in use at lower amplitudes. Distortion as a percentage of the signal amplitude will increase and resolution will decrease at lower amplitudes. This output is DC coupled and the signal is inverted relative to the digital outputs and to the analog generator output.

## Analog Output via Generator Output

The D/A output signal may also be routed through System One’s analog generator amplitude control circuitry, power amplifier and output transformer, and circuitry for selection of source impedance, balanced vs. unbalanced connections, and floating vs. ground-referenced connection. To route the D/A signal through the analog generator outputs, select Waveform DGen on the Analog Generator panel. The digital generator output DGen Ampl field on the DSP panel will then be over-ridden (set to maximum) so that all control of amplitude comes from the Amplitude field on the Analog Generator panel. Frequency is controlled from the DGen Freq field on the DSP panel. In this mode, the digital signal constantly operates at the full 16-bit resolution of the D/A. Thus, the output waveform distortion is essentially unaffected by the Amplitude selected on the Analog Generator panel. Resolution is constant at all amplitudes. When the D/A output is routed through the analog generator output stage by the Waveform DGen selection, the Analog Generator A and B output buttons provide control of which output channel is driven.

## Digital Audio Outputs

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Serial or Parallel may also be selected on the Output Format line if a Dual Domain (SYS-300) system is in use. The general-purpose parallel output connector is located on the rear panel. The Serial Type field of the DIO panel permits selection among the serial formats. The general-purpose serial output and input share a 15-pin D-sub connector on the rear panel. The AES/EBU & SPDIF outputs are a male XLR connector and an RCA phono connector (labeled SPDIF) on the front panel. An optical connector was added on units after serial number SYS1-32214. On units before serial number SYS1-32214, the XLR and phono connectors are effectively in parallel and should not be used at the same time. On the units with optical connectors, the optical, phono, and XLR connectors are driven in parallel, are all buffered and may all be used simultaneously. All digital formats are multiplexed dual channel, 24-bit linear 2's complement format.

Since the digital interfaces are all two-channel multiplexed signals, the A and B buttons at the bottom of the Digital Analyzer panel permit driving either channel, both, or having no output.

### SIA-322 Serial Interface Adapter

An accessory unit, the SIA-322, is available to greatly simplify connection of serial input digital devices other than AES/EBU and SPDIF/EIAJ to System One Dual Domain. The SIA-322 permits selection of a wide variety of serial formats of both one and two channels with from 8 to 24 bits to be interfaced simply by setting front-panel switches on the SIA-322. MSB-first and LSB-first data can be used with a variety of word-strobe positions, in either TTL or CMOS logic families. Contact Audio Precision or your Audio Precision International Distributor for more information on the SIA-322.

### Resolution Field, Input vs. Output

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In all DSP programs except BITTEST the Resolution field on the DIO panel determines the word width that the generated output signal will be rounded to, and also controls the amplitude of any dither used (see below). With BITTEST, the Resolution field controls the input word width (see the BITTEST chapter starting on page 9-). If a

parallel-interfaced digital device under test has less than 24 bits of resolution, it should be connected to the most significant bits of System One Dual Domain's parallel output. Setting the Resolution field to the word width of the external device causes the DSP to round to that word length, producing a properly-symmetrical signal. When using the AES/EBU, optical or SPDIF outputs the bits below this point will be zeroed. If the Resolution field is set to a longer word width than the external device uses, the output word from the DSP generator is effectively truncated by its connection to the device with the shorter word width.

## Dither

Dither is noise combined with the signal to improve linearity, reduce distortion at low amplitudes, and extend the linear operating range below the theoretical minimum for undithered PCM signals of any particular resolution. The additional noise is introduced before quantizing and serves to randomize the quantization distortion and produce an undistorted signal with a slightly higher noise floor. If dither has not been turned off by use of the "None" selection in the Dither Type field of the DIO panel, the Resolution field on the DIO panel controls the amplitude of dither added to the digital generator output (except during use of BITTEST which has no dither available). For example, proper dither for a 16-bit system is obtained by entering 16 in the Resolution field. When System One's internal D/A converter is selected as the output port, the Resolution field is over-ridden and internally set to 16 bits.

Three choices of dither are available; Triangular, Rectangular, and Shaped (plus None). The selection between them is made via the Dither Type field on the DIO panel. Note that the dither selected also appears at the parallel digital output. Triangular dither is the default choice. Triangular and Rectangular refer to the shape of the probability functions of the two types of dither. Triangular dither can add or subtract up to one bit peak amplitude at the selected Resolution. Rectangular dither can add or subtract only 1/2 bit amplitude. Thus, Triangular dither increases the noise floor more than Rectangular. However, Rectangular dither will result in variations of the noise floor as a function of signal amplitude, while the noise floor is independent of signal with Triangular dither. Shaped dither is

Triangular probability function dither that has further been spectrally-shaped by decreasing its amplitude at low frequencies and increasing it at high frequencies. The audible effect of Shaped dither is thus significantly reduced, since the human ear is less sensitive at the higher frequencies where the dither is greater. Note that only one dither generator is used for the two channels so that a subtraction of the two channel signals by the device under test will produce zero.

For more information on dither, see the paper “Digital Dither: Signal Processing With Resolution Far Below The Least Significant Bit” by John Vanderkooy and Stanley P. Lipshitz, presented at the AES 7th International Conference “Audio in Digital Times” in Toronto on May 14-17, 1989.

## Parallel Digital Interface

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The general purpose parallel ports are multiplexed dual channel, 24-bit linear 2’s complement format. They are intended for connecting to A/D and D/A converters or for direct connection to DSP systems under test. They are also used to connect the SIA-322 Serial Interface Adapter to System One Dual Domain.

### Parallel Port General Information

The parallel ports are available on the rear panel of the system. The input and output each consist of a 34 pin dual row square pin connector. This is intended to mate with a multiconductor ribbon cable with a crimp-on insulation displacement connector. These cables may then be connected as desired at the other end. To reduce the risk of damage it is strongly recommended that all connections to the parallel ports be made with power to the System One turned off.

There are 24 data lines, a channel select line, a strobe line and a sample clock line. The remaining pins are grounds, plus one +5 V power line. The table below lists the pinout of both input and output connectors. All lines are independent between the two connectors except the sample clock output, which is shared. Figure 6-4 illustrates the physical orientation of the pins on the connectors when viewed from the rear of the instrument. Note the location of the key slot in the System One connector which prevents reverse installation of the

mating cable connector. The cable connector key is a small raised portion in the center of one of the wide surfaces of the cable connector. It is highly recommended that mating connectors be used which have the corresponding key. This will prevent mis-insertion and reduce the risk of damage to the system or to the device under test.

### Parallel Interface Connector Pin Assignments

Pin #	Function	Pin #	Function
1	Ground	18	Bit 8
2	Ground	19	Bit 7
3	Bit 23 (MSB)	20	Bit 6
4	Bit 22	21	Bit 5
5	Bit 21	22	Bit 4
6	Bit 20	23	Bit 3
7	Bit 19	24	Bit 2
8	Bit 18	25	Bit 1
9	Bit 17	26	Bit 0 (LSB)
10	Bit 16	27	Ground
11	Bit 15	28	Channel Select In
12	Bit 14	29	Ground
13	Bit 13	30	Data Strobe In
14	Bit 12	31	+5 Volts
15	Bit 11	32	Sample Clock Out
16	Bit 10	33	Ground
17	Bit 9	34	Ground

The +5V power line provided on the connectors is intended only to be used to tie unused data or control inputs high or to power termination networks if necessary. The power supply of the System One has not been designed to source current to customer-supplied devices.

All integrated circuits that connect directly to connector pins have been placed in sockets for easy replacement in

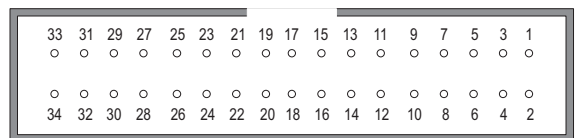


Figure 6-4 Parallel Input and Output Connectors

the event of damage. All input and output data bits are processed through 74ACT670 Dual Port Register Files. Control lines are buffered with 74AC244 Octal Buffers.

## Parallel Input

The parallel input accepts 24-bit data on the data lines and latches it when the strobe line makes a transition from low to high. The data is latched into the channel A input when the channel select line is held low and the strobe makes a transition from low to high. The data is latched into the channel B input when the channel select line is held high and the strobe makes a transition from low to high. Latching channel B data also generates an interrupt for the System One, causing it to read both channels of data just written to the latches. The channel select line must be held either high or low during the entire time the strobe line is low. A transition on the channel select line while the strobe line is low will cause an incorrect latch operation. Figure 6-5 illustrates the correct operation of the parallel input port and identifies

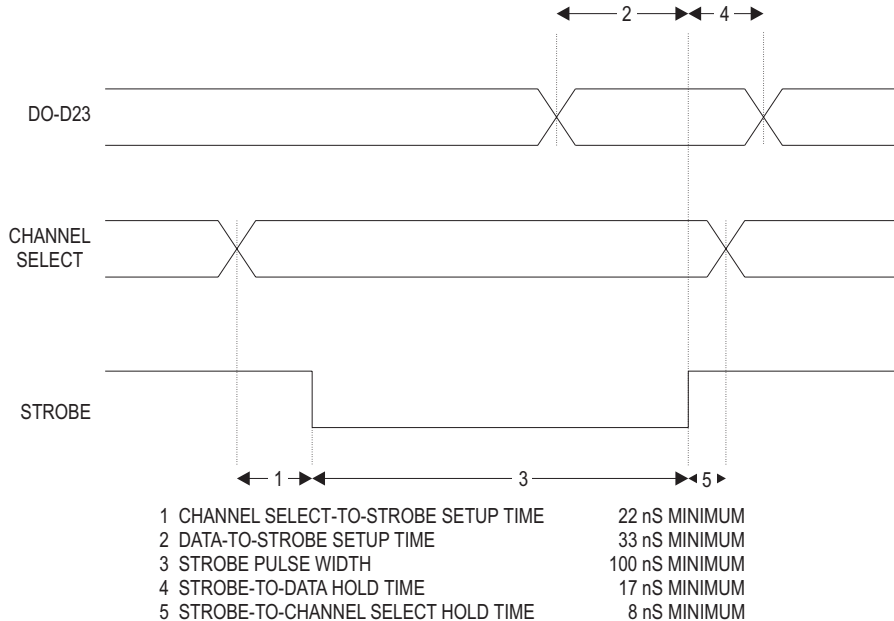


Figure 6-5 Parallel Input Port, Timing Relationships

the timing specifications for the operations. Note that data must always be supplied to Channel B since its latch operation triggers the DSP to read the data. If a single channel interface is desired, it must therefore be configured as Channel B and the channel select line connected to +5V.

All data inputs are TTL level-compatible high speed CMOS inputs. This allows connection to a wide variety of devices without loading concerns. The capacitance of each data line is approximately 10 picofarads. There are no termination or pull-up resistors. If less than 24 bits are applied to the input, the data must be applied to the most significant bits and the unused LSB input lines must be connected to ground. The strobe line and channel select line are TTL level high speed CMOS inputs.

The channel select and strobe inputs are internally pulled high with resistors. If the pins are not connected, this will suffice to maintain the signals high. However, if wires are attached to either of these lines and bundled into cables with the data lines, there may be sufficient capacitance and mutual inductance to couple transitions into one or both inputs, causing mis-operation of the interface. To reduce the magnitude of this coupling, the channel select and strobe lines should be driven from a low impedance source. Using shielded or ground-plane ribbon cable may reduce the coupling of data-line transitions into the strobe or channel-select inputs. However, the added load capacitance on the data lines may increase the noise created by data transitions. Placing resistors of from 22 Ohms to 100 Ohms in series with the source outputs will reduce the rise time of signals and reduce coupling into adjacent lines. On bundled cables there may be significant inductive coupling between the data lines and the strobe line. If this occurs, the strobe line may be removed from the bundle and run separately from the remaining lines. Terminating the strobe and channel select lines with a resistive pull-up/pull-down network at the connector may reduce the coupling and generally improve signal integrity. For most applications a 390 Ohm pull-up to +5V and a 220 Ohm pull-down to ground will present an adequate load for each line. These two networks may be powered by the +5V pin on the interface connector and should be mounted on the

connector itself. Proportionately lower resistances may be required when the cable impedance is significantly lower.

Logic noise on the System One ground connections may induce noise in the analog portions of the circuit being measured if care is not taken in the connections. This has resulted in some low level spurious noise observed when measuring oversampling A/D converters with switched capacitor input circuits. Oversampling A/D converters which use linear circuits for their loop filters have not shown similar sensitivity to ground noise, nor have conventional successive approximation converters. If ground noise effects are suspected, the grounds of the converter board under test and the System One may be isolated with ferrite beads in the ground conductors or by opto-isolators in the data link.

If some of the bits (especially the sign bit) are allowed to leak into the analog signal it will result in odd harmonic distortion (3rd, 5th, 7th, etc.) in the measured results. Excessive distortion is often a clue to examine the leakage paths in the device under test.

## Parallel Output

Note that dither is also fully operational for the parallel digital output signal. See the dither section on page 6-17 for details. The output bits below the dither point are not truncated by the dsp. If they need to be zeroed it should be done in the wiring from the parallel port to the device under test.

The output port provides 24-bit data on the data lines at all times. The channel select line determines which channel's data appears on these lines. The read strobe line causes both channels of data to be updated, and signals the System One processor that another pair of data words are required. The data is buffered with a two stage FIFO buffer that guarantees that data will be available when needed rather than after the processor's service delay. As long as the maximum sample rate restriction is obeyed the two-stage buffer will never empty. Figure 6-6 illustrates correct operation of the parallel output port and identifies timing specifications for the operations.



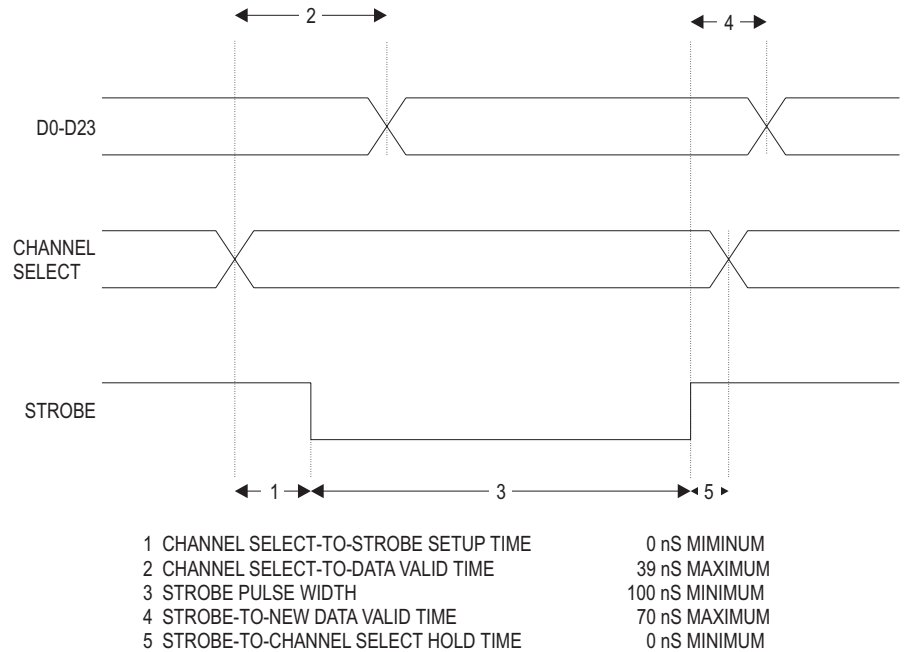


Figure 6-6 Parallel Output Port, Timing Relationships

Note that data may be read from one or both channels, as desired. However, the typical wiring connection to these ports provides an automatic strobe operation when the channels are read. If a single channel output is desired, the port should be configured as Channel B and the channel select line connected to +5V.

The parallel inputs and outputs may be operated at any sample rate selectable from the panel of the DSP program being run. The internal sample clock generator can produce a square wave sample clock at 32 kHz, 44.1 kHz, or 48 kHz. The DSP program in use may limit the choice of sample rates based on program processing time limitations. The internal sample clock generator also appears on pin 4 of the rear panel serial connector along with a 64x version of the sample clock. This higher frequency signal may be useful for clocking de-glitchers or similar circuits on the device under test. If an external sync signal is applied to the external sync input, the sample clock outputs will lock to it.

There is no limitation that the input and output sample rates be exactly the same or synchronous. However, it is assumed that the sample data pairs are read and written at a constant sample rate. The small FIFO buffers will allow sample jitter in the input and output operations of as much as  $3/4$  of a sample period.

The strobe line and channel select line are TTL level high speed CMOS inputs. All outputs are CMOS with 22 Ohm build-out resistances in series. This allows connection to a wide variety of devices without loading concerns. If less than 24 bits are needed, the most significant bits should be used and the unused LSB lines must be left unconnected.

The channel select and strobe inputs are internally pulled high with resistors. If the pins are not connected, this will suffice to maintain the signals high. However, if wires are attached to either of these lines and bundled into cables with the data lines, there may be sufficient capacitance and mutual inductance to couple transitions into one or both inputs, causing mis-operation of the interface. To reduce the magnitude of this coupling, the channel select and strobe lines should be driven from a low impedance source. Using shielded or ground-plane ribbon cable may reduce the coupling of data-line transitions into the strobe or channel-select inputs. However, the added load capacitance on the data lines may increase the noise created by data transitions. On bundled cables there may be significant inductive coupling between the data lines and the strobe line. If this occurs, the strobe line may be removed from the bundle and run separately from the remaining lines. Terminating the strobe and channel select lines with a resistive pull-up/pull-down network at the connector may reduce the coupling and generally improve signal integrity. For most applications a 390 Ohm pull-up to +5V and a 220 Ohm pull-down to ground will present an adequate load for each line. These two networks may be powered by the +5V pin on the interface connector and should be mounted on the connector itself. Proportionately lower resistances may be required when the cable impedance is significantly lower.

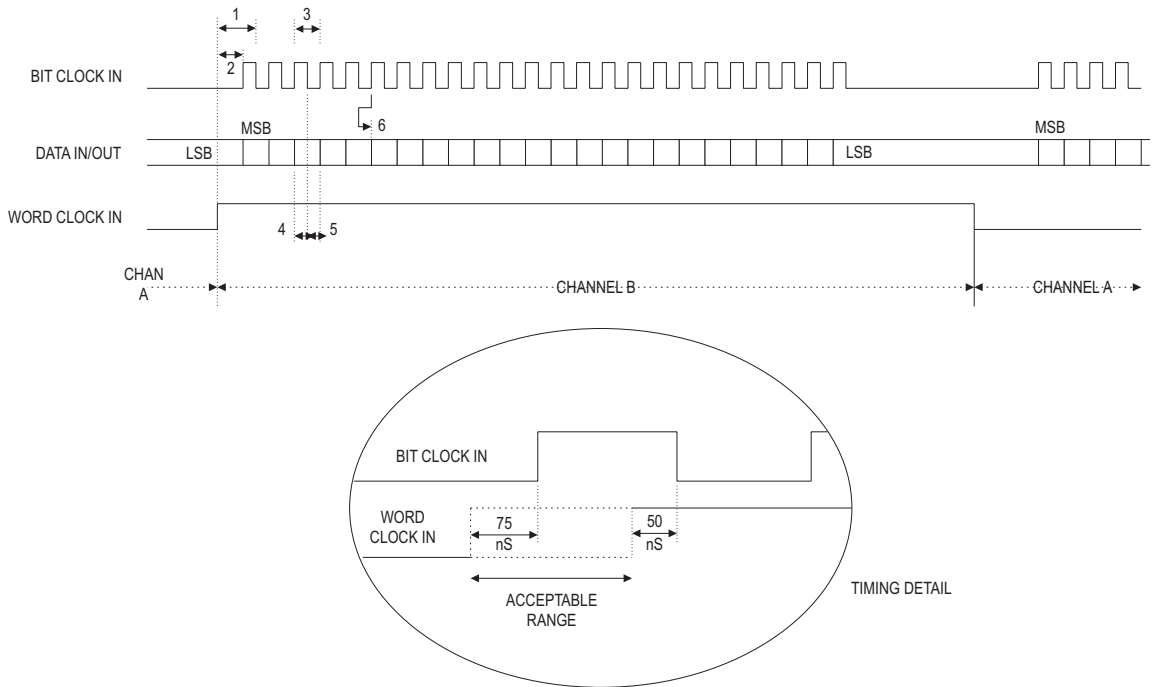
## Rear Panel Serial Connector

The 15-pin “D-sub” connector on the rear panel provides several auxiliary signals for the parallel and AES/EBU/SPDIF interfaces, and provides a general purpose serial input/output port. Clock outputs are provided to facilitate operation of the parallel port and to synchronize external devices with the AES/EBU signal. The pinout of the connector is detailed below. All inputs are TTL level compatible CMOS. All outputs are CMOS isolated by 24 Ohm series resistors.

Pin #	Function
1	Ground
2	+5 Volts (to tie unused inputs high)
3	Auxiliary input (DSP program-specific)
4	Internal sampling clock output
5	64x Internal sampling clock output
6	Serial Output data output
7	AES/EBU Input word strobe output
8	Ground
9	Unused
10	Serial Input bit clock input
11	Auxiliary output (DSP program specific)
12	Serial Output bit clock input
13	Serial Input data input
14	Serial Output word strobe input
15	Serial Input word strobe input

## General Purpose Serial I/O

The information in this section is provided for the user who plans to construct his own logic interface circuitry in order to interconnect System One Dual Domain and digital devices with serial interfaces. Alternatively, the SIA-322 Serial Interface Adapter is an accessory product to System One Dual Domain which permits interconnection to serial interface devices without the necessity of designing or constructing special interface circuitry. Switches on the SIA-322 front panel permit selection of word length from 8 to 24 bits, single or dual-channel operation, data word length from 8 to 24 bits with MSB or LSB first, right or left justification, and 0 or 1 padding, selectable



1 WORD CLOCK INPUT SETUP TIME (to bit clock falling edge)	50 nS MINIMUM
2 WORD CLOCK INPUT SETUP TIME (to bit clock rising edge)	75 nS MAXIMUM
3 BIT CLOCK PERIOD	312 nS MINIMUM
4 DATA INPUT SETUP TIME (to bit clock falling edge)	30 nS MINIMUM
5 DATA INPUT HOLD TIME (from bit clock falling edge)	50 nS MINIMUM
6 BIT CLOCK TO DATA OUTPUT DELAY (from bit clock rising edge)	75 nS MAXIMUM

Figure 6-7 General Purpose Serial Port, Timing Relationships

word clock transition within the data frame with bit-wide or word-wide clock, TTL or CMOS logic family support in either polarity and independent selection of bit clock and word clock source. For more information on the SIA-322, contact Audio Precision or your Audio Precision International Distributor.

The general purpose serial port consists of six signals, three for the input port and three for the output port. The serial port shares hardware with the AES/EBU-SPDIF/EIAJ interface, so the two may not be used simultaneously. The selection between the general purpose serial interface and the AES/EBU and SPDIF/EIAJ interface is made with the Serial Type field on the DIO panel. The hardware word width

of the general purpose serial port is always 24-bit. Unused bits fed to the DSP should be zeroed.

The serial input consists of three lines, all of which are inputs to the hardware. These inputs are TTL-level-compatible CMOS inputs. There is a data input, a bit clock input and a word strobe input. The user must generate both the bit clock and word strobe signals. The interface always operates in a dual channel mode with a word-width word-strobe input. If this port is used, the AES/EBU and SPDIF/EIAJ interface ports (both input and output) are not available.

See Figure 6-7 for timing diagrams of the general purpose serial port. Serial input data is clocked on the falling edge of the bit clock. The most significant bit is always first. The clock signal must be 48 times the data sample rate providing 24-bit data for each channel. This may be obtained with a gated clock at 64 times the sample rate, which lies idle for 8 bits of the 32-bit frame as illustrated in the figure. The maximum word strobe frequency is 48 kHz, the minimum is 32 kHz. The maximum bit clock frequency is 3.072 MHz (64 times the 48 kHz sample rate).

The serial output consists of three lines. Two are TTL level compatible CMOS inputs and one is a CMOS level output. The serial data line is an output. The bit clock and word strobe are both inputs. The user must generate both the bit clock and word strobe signals. If this port is used the AES/EBU and SPDIF/EIAJ interface ports (both input and output) are not available.

Serial output data is clocked out on the rising edge of the bit clock. The most significant bit is always first. The clock signal must be 48 times the data sample rate providing 24-bit data for each channel. This may be obtained with a gated clock at 64 times the sample rate, which lies idle for 8 bits of the 32-bit frame as illustrated in the figure. The maximum word strobe frequency is 48 kHz, the minimum is 32 kHz. The maximum bit clock frequency is 3.072 MHz (64 times the 48 kHz sample rate).

The signals required to clock data at the serial interface may be created from the 64x master clock on the serial connector. However, since two 24-bit intervals are to be obtained from a 64x clock there will

be idle periods in the resulting clocks. Care should be taken to insure noise-free and glitch-free signals on the cabling between the serial interface connector and the device being driven. The interface hardware is extremely fast and will respond to very short duration pulses.

### **AES/EBU Input Word Strobe Output**

This output is a square wave at the sample rate of the AES/EBU input. This allows triggering an oscilloscope off the received data words to view the waveform being received. Channel A or channel B of the interface may be selected by the trigger slope control of the oscilloscope.

### **Sample Clock Outputs**

The sample rate clock signal driving the AES/EBU output is available on one pin. This is the same as the parallel port sample frequency clock output. This allows triggering an oscilloscope off the transmitted data words to view the waveform being transmitted. Channel A or channel B of the interface may be selected by the trigger slope control of the oscilloscope. When the external sync input is used this signal will be slaved to the applied sync.

There is also a 64x sample rate clock output for triggering an oscilloscope at the bit rate of the interface. This clock may be used for eye pattern testing on the AES/EBU interface. It may also clock the serial input or output through external logic.

### **Auxiliary Inputs and Outputs**

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There is one auxiliary input and one auxiliary output on the 15-pin interface connector. These are DSP program-specific-signals whose functions will depend on the program loaded. The output may be used by the processor to signal events during the measurement or for auxiliary triggers during signal generation. The input may be used by the processor to sense that some external trigger event has occurred or to determine the state of an external process. If any functions are implemented on these signals they will be described in the particular DSP program chapter or separate DSP program manual.

The +5V power line provided on the connector is intended to tie unused data or control inputs high, if necessary. The power supply of System One has not been designed to source current to customer supplied devices. Momentary shorting of this supply to ground may cause the DSP to reset and require a Utilities Restore Hardware menu command to return to normal operation.

### External Sync Input

The 1/4 inch phone jack on the rear panel is designed to accept a house sync signal for driving the AES/EBU output. The clocks provided on the serial and parallel I/O connectors will also be slaved to this signal. The external sync signal may be an AES/EBU format signal (with or without data) or a square wave at twice the sample rate. If an AES/EBU signal is applied only the preamble information will be used to generate the internal clock signals. The data and control bits will be ignored. If a square wave is used it must be at twice the desired sample rate. Operation at a 48 kHz sample rate would thus require a 96 kHz square wave to be applied. When one of these two types of sync signals is applied the DSP will automatically determine the format and lock on. Lock may take up to two seconds after the signal is applied. Locked or unlocked status is explicitly indicated by the Receive Sync indicator on the Digital I/O panel.

The input will lock onto an AES/EBU sync signal at one of the three standard digital audio sample rates of 48 kHz, 44.1 kHz or 32 kHz or a square wave at twice these values. If a relatively clean signal is used the rate may be varied by up to 5 % high or low and the internal phase lock loop will still follow the signal. Thus, System One Dual Domain can be used with digital audio sample rates such as the 44.056 kHz used in certain video-related systems. When this input is used the internal crystal oscillators will not be used for the AES/EBU output functions. Frequency measurements made by FFT or the GENANLR DSP programs will be in error by the percentage deviation of the incoming sample rate from the DSP panel-selected rate.

For testing of interfaces at non-standard sample rates, an external square wave generator or the System One analog generator sync output may be routed to the external sync input. One effective setup is to use an external function generator with its VCF (voltage-controlled

frequency) input driven from one of the DC variable outputs of the DCX-127 multifunction unit. Sweeping the DCX DC Out as SOURCE-1 through the appropriate DC voltage range then results in sweeping the function generator squarewave frequency through a desired range, while measuring performance of the digital interface. If the System One analog generator is used, sharp and brief discontinuities may occur at 96 kHz and other integer values. This is due to the 16-bit frequency tuning DACs in the analog generator being programmed one byte (one-half word) at a time.

The external sync input is balanced and transformer coupled. The connector is a tip-ring-sleeve 1/4 inch phone jack. Normal connection is to the tip and ring with a balanced signal. If an unbalanced signal is connected the ring must be grounded. This will automatically be accomplished if a two conductor phone plug is used for the unbalanced connection. The ring should never be left unconnected or unreliable operation will result.

### External Trigger Output

The specific function of this connector will depend on the DSP program in use. If any function is implemented on this connector it will be described in the particular DSP program chapter or manual. This output is a 5 Volt CMOS/TTL compatible signal with a 22 Ohm source impedance.



# Narrow Bandpass Filter

To load the Narrow Bandpass Filter DSP program (HARMONIC), select “Narrow bandpass filter (harmonic)” at the top of the Digital Analyzer panel.

## Introduction

The Narrow bandpass filter program is designed for analog input only, **measuring signals from the Analog Analyzer Reading meter**. The program consists of a sharp bandpass filter whose center frequency may be set or swept across the audio spectrum and above. Two bandwidths are available at center frequencies from 20 Hz to 20 kHz (48 kHz sample rate), the wider of which is also tunable from 80 Hz to 80 kHz at the 192 kHz sample rate. The wider filter, available at both sample rates, has a Q (ratio of center frequency to 3 dB bandwidth) of about 12 (1/8 octave bandwidth). The sharper filter, available only at the 48 kHz sample rate, has a Q of about 15 (1/10 octave bandwidth).

HARMONIC features flexible steering of the filter center frequency. The user can select as the source of frequency information (to be tracked) the Analog Generator frequency, the Analog Analyzer hardware Bandpass/Bandreject filter frequency (which in turn can be steered by the Analog Analyzer frequency counter reading), or a user-entered value on the HARMONIC panel. The center frequency may then be made to track the source value on a 1:1 basis, or at any multiple of the source value from 1.00 to 9.00, or at any

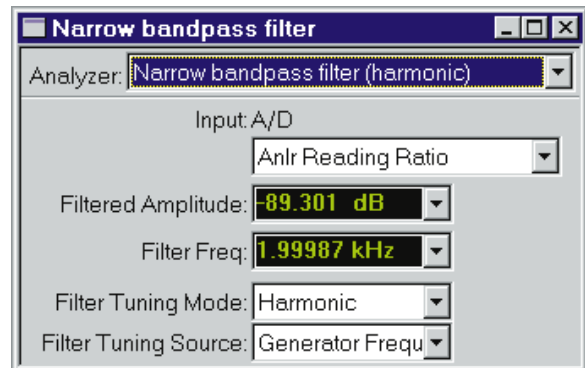


Figure 7-1 HARMONIC Panel, Small Version

user-entered value of frequency offset above or below the steering source frequency. When tracking the panel frequency value, the filter center frequency may be swept as Source 1 or Source 2 on the Sweep panel.

Although it is a DSP-implemented filter, HARMONIC is usable only from the analog analyzer Reading meter via a built-in A/D converter. Only analog domain units are available. If a tunable narrow bandpass filter

is needed in the digital domain, use the GENANLR program that includes a bandpass filter identical to the sharp mode of HARMONIC.

## Typical Applications

HARMONIC is used as a wave analyzer or selective voltmeter to make frequency-selective amplitude measurements. Common examples include making measurements of individual harmonic distortion product amplitudes following the analog analyzer's bandreject filter in THD+N function, measuring the amplitude of individual intermodulation distortion products, and making accurate amplitude measurements of sinewaves near or below the wideband noise level, as required in measurements such as analog tape print-through and crosstalk.

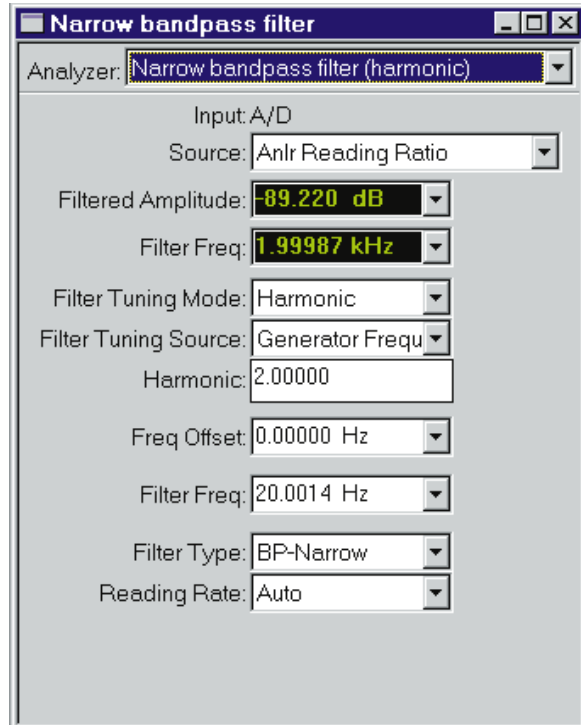


Figure 7-2 HARMONIC Panel, Large Version

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## Source

Only analog signals may be measured by HARMONIC, routed to the A/D converter from the main Reading meter. The Source field has two selections: **Anlr Reading Ampl** or **Anlr Reading Ratio**. The **Anlr Reading Ampl** selection provides signals with an absolute amplitude calibration in units such as Volts, dBV, dB<sub>r</sub>, dB<sub>u</sub>, and dB<sub>m</sub>. The **Anlr Reading Ratio** selection provides calibration of Reading meter signals referred to the amplitude of another signal. This includes individual harmonic amplitude measurements in % or dB below the fundamental when the Reading meter is in THD+N Ratio function, or measurements of individual IMD product amplitudes in % or dB when the Reading meter is in one of the IMD functions.

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## Filtered Amplitude Display

Amplitude measurements at the output of the bandpass filter are displayed in the “Filtered Amplitude” field at the top of the panel. The detector is a true RMS detector. Units of measurement may be changed by clicking on the down arrow at the right of the display field, then selecting the desired unit from the list that is displayed. Only analog domain units are available.

The input signal to HARMONIC always comes from the Analog Analyzer Reading meter. If the IMD option, wow and flutter option, or another Reading meter measurement function in a ratio display mode is in use (such as THD+N ratio), the % and dB units of HARMONIC are calibrated with respect to the definition of the parameter being measured.

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## Filter Frequency Display

The “Filter Freq” display (second row of green digits on black background) is not truly a measurement, but shows the center frequency to which the narrow bandpass filter is tuned. This frequency is the net result of the Filter Tuning Mode and Filter Tuning Source controls, Harmonic and Freq Offset fields, and the frequency to which the selected Tuning Source is currently set.

## Filter Tuning Mode

When a Tuning Source has been selected, one of three relationships between the bandpass filter center frequency and the frequency value of that Tuning Source may be selected in the Filter Tuning Mode field. To change the tuning mode, click on the down arrow at the right of the field and click on the desired selection. The three selections are Direct, Harmonic, and Offset.

When “**Direct**” is selected, the tunable filter center frequency will be exactly equal to the frequency value of the selected Tuning Source.

When “**Harmonic**” is selected, the tunable filter center frequency will be an integer multiple of the frequency value of the selected Tuning Source. The value of the integer is selectable from 2 through 9 in the Harmonic field.

When “**Offset**” is selected, the tunable filter center frequency will be higher than the frequency value of the selected Tuning Source by the value in the Freq Offset field if that value is positive, and will be lower than the source by the Freq Offset value if that value is negative.

## Filter Tuning Source

The selection in the Filter Tuning Source field determines which of three numbers is used as the information to steer the bandpass filter center frequency. The number from the selected source will be further modified by the Harmonic value in Harmonic Filter Tuning Mode and by the Freq Offset value in Offset Tuning Mode. The three Filter Tuning Source selections are Panel Setting, Analog Filter, and Generator Frequency.

When “**Panel Setting**” is selected, the value entered in the Filter Freq field (visible only on the large version of the Digital Analyzer panel) is the tuning source. This is a fixed value when no sweep is taking place. If “Harmonic” is the selected Instrument and “Filter Freq” is the selected parameter at Source 1 or Source 2 setting parameter on the Sweep panel, then the Panel value and thus the filter center frequency can be swept as part of test.

When “**Analog Filter**” is selected, the present center frequency of the Analog analyzer bandpass/bandreject filter is the tuning source. The Analog Analyzer BP/BR filter may in turn be tracking the Analog Generator frequency (Gen Track), be controlled by the Analog Analyzer Frequency Counter (Auto), or be Fixed at a value entered on the Analog Analyzer panel. If the Analog Analyzer BP/BR filter is in Auto mode, this selection permits selective measurements with HARMONIC which track the frequency of a signal from a test disc or test tape or which originate at some remotely located sweeping generator.

When “**Generator Frequency**” is selected, the present frequency of the main sinewave oscillator in the Analog Generator is the tuning source.

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## Harmonic Multiplier

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The “**Harmonic**” field permits entry of a numeric value which will be multiplied by the Filter Tuning Source value to determine the filter center frequency when the Filter Tuning Mode is set to “Harmonic”. It has no effect if the Filter Tuning Mode is not “Harmonic”. The Harmonic field is visible only on the large version of the Digital Analyzer panel. Values from 1.000 to 9.000 may be entered. For individual harmonic distortion measurements, the value must be an integer from 2.0 (2nd harmonic) through 9.0 (9th harmonic), but the entry need not be an integer. The Analog Analyzer Reading meter should be in THD+N function to reject the fundamental signal in order to achieve full performance from HARMONIC.

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### Frequency Limitations Due to Sample Rate

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It is important to remember that the tunable filter center frequency is limited to a maximum value of 20 kHz at the 48 kHz sample rate and to 80 kHz at the 192 kHz sample rate. High Harmonic values combined with high values of the Tuning Source presently selected can easily lead to filter center frequencies that cannot be achieved. For example, if the Filter Tuning Source is Generator Frequency and if the Analog Generator frequency is set or swept to 20 kHz, no Harmonic value above 4 could be used at the 192 kHz sample rate (80 kHz bandwidth limitation) and the 48 kHz sample rate would not support

Harmonic Tuning Mode at all. The maximum Harmonic value of 9 could be used only when the Tuning Source value is 8.88 kHz or less at the 192 kHz sample rate, or to Tuning Source values of 2.22 kHz or less at the 48 kHz sample rate. Recommended practice is to always use the 192 kHz sample rate when the Filter Tuning Mode is “Harmonic”, in order to provide operation across as much of the audio spectrum as possible.

## Frequency Offset from Tuning Source

The “**Freq Offset**” field permits user entry of a number which controls the frequency difference between the bandpass filter center frequency and the selected Filter Tuning Source when the Filter Tuning Mode is set to “**Offset**”. It has no effect if the Filter Tuning Mode is not “**Offset**”. The Freq Offset field is visible only on the large version of the Digital Analyzer panel. This is a direct numeric entry field that may be typed into from the keyboard or pasted into from the Windows clipboard. A positive value in this field causes the filter center frequency to be higher than the source by the specified value. A negative value causes the filter frequency to be lower than the source by the specified value.

## Frequency Limitations Due to Sample Rate

It is important to remember that the tunable filter center frequency is limited to the range of 20 Hz to 20 kHz at the 48 kHz sample rate and the 80 Hz to 80 kHz range at the 192 kHz sample rate. High positive Freq Offset values combined with high values of the Tuning Source presently selected, or high negative Freq Offset values combined with low values of the Tuning Source can lead to filter center frequencies that cannot be achieved. For example, if the Filter Tuning Source is Analog Generator and if the Analog Generator frequency is set or swept below 1 kHz, a Freq Offset value of -1 kHz will cause an error message.

## Panel Filter Frequency Entry

The **Filter Freq** entry field, visible only on the large version of the Digital Analyzer panel, permits user entry of a value which will control

the bandpass filter center frequency when the Filter Tuning Source is “**Panel Setting**”. This is a direct numeric entry field that may be typed into from the keyboard or pasted into from the Windows clipboard. When Harmonic is the selected instrument and Filter Frequency is selected as the Source 1 or Source 2 settings parameter on the Sweep panel, the “Filter Freq” value will be controlled by the Sweep panel during a sweep and will return to the value entered into the “Filter Freq” field at the end of the sweep. The value in the Filter Freq field will be further processed by the Freq Offset value if Filter Tuning Mode is “Offset”, or by the Harmonic value if Filter Tuning Mode is “Harmonic”.

### Frequency Limitations Due to Sample Rate

It is important to remember that the tunable filter center frequency is limited to the range of 20 Hz to 20 kHz at the 48 kHz sample rate and to the range of 80 Hz to 80 kHz at the 192 kHz sample rate. A combination of “Filter Freq” entry values and “Harmonic” or “Freq Offset” values can lead to commands to the filter which fall outside those ranges and error messages will result.

## Filter Bandwidth Selection

HARMONIC may be operated as a tunable bandpass filter, with two alternate selectivity values at the 48 kHz sample rate, or as simply a non-selective RMS detector (except for a high-pass filter to reject DC). Selection among these three filter shapes is made in the Filter Type field, which is visible only on the large version of the Digital Analyzer panel. To change the filter shape, click on the down arrow at the right of the field and click on the desired selection in the list which is displayed. The selections are **Highpass**, **BP-Wide**, and **BP-Narrow**.

When BP-Wide is selected, the Q of 12 (1/8 octave) filter is selected. This filter is available both at 48 kHz and 192 kHz sample rates (80 kHz maximum center frequency).

When BP-Narrow is selected, the Q of 15 (1/10 octave) filter is selected. This filter is available only at the 48 kHz sample rate (20 kHz maximum center frequency).

When Highpass is selected, the digital analyzer has flat frequency response across the range from 20 Hz to 21.75 kHz at the 48 kHz sample rate and from 80 Hz to 80 kHz at the 192 kHz sample rate.

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## Reading Rate

The Reading Rate field on the HARMONIC panel permits user control of the detector integration period and RMS detector reading rate. This field is visible only on the large version of the Digital Analyzer panel. To change the selection, click on the down arrow at the right of the field and click on the desired choice from the list which is displayed. The selections are Auto, 4/sec, 8/sec, 16/sec, 32/sec, and 64/sec.

When “Auto” is selected, APWIN software makes an automatic selection of the detector reading rate as a function of the present filter center frequency. During a sweep, the Auto selection chooses the fastest reading rate that does not compromise the specified accuracy of the detector. When a sweep involving the HARMONIC analyzer is not in progress, Auto sets the reading rate to 4/sec.

When any of the specific reading rates are selected, the detector reading rate remains fixed at that value regardless of the filter center frequency. This can result in measurement errors when using fast reading rates at low filter frequencies.



# Digital Domain Tester

The Digital Domain Tester (GENANLR.AZ1) consists of a sinewave generator and a variety of audio analysis functions for digital domain signals. To load the Digital Domain Audio Tester DSP program, display the Digital Analyzer panel. Click on the down arrow at the right end of the “Analyzer” field, at the top of the panel, and select “Digital domain audio tester (genanlr)”.

## Introduction

The Digital Domain Audio Tester (GENANLR) is a DSP-implemented digital-domain emulation of the most common analog domain types of audio measurement. Thus, it permits measurements on audio signals in the digital domain that can be readily compared to conventional analog audio measurements of other devices. GENANLR is not intended for measurements of analog-domain signals. GENANLR is a real-time program. Signal is continuously generated and measurements can be continuously observed in panel or bargraph mode for adjustment purposes.

The signal generation capability of this program provides flexible sinewave generation at any amplitude and frequency. Digital domain signal measurement capabilities include frequency measurements, simultaneous level measurements on both channels, A-weighted or CCIR-468-weighted noise measurements with RMS or quasi-peak detector, and measurements through a bandreject

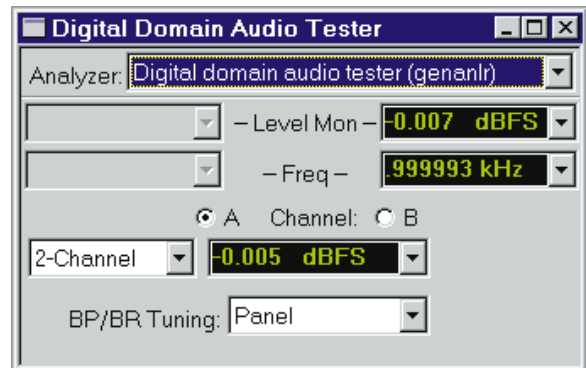


Figure 8-1 GENANLR Panel, Small Version

(notch) or bandpass filter to support THD+N and selective measurements. Since these same types of signal generation and measurement are part of the capabilities of the analog hardware sections of System One, all these measurements may be made in any combinations of domain including cross-domain measurements to test A/D and D/A converters.

## Sinewave generation

GENANLR can generate a 24-bit resolution sinewave at frequencies from approximately 5 Hz to 22.5 kHz at the 48 kHz rate, 20.67 kHz at the 44.1 kHz rate, and 15 kHz at the 32 kHz rate. The frequency resolution is  $(\text{rate})/224$  where “rate” is the digital sample rate selected. With a 48 kHz sample rate, for example, the resolution is  $48,000/224$  or approximately 0.0029 Hz.

The amplitude may be set over the range from 0 dBFS to  $-125$  dBFS at 24-bit resolution. Digital channels A, B, or both A and B may be driven. Caution should be exercised in not selecting output levels near maximum when significant amounts of dither are also selected. The sinewave signal and dither signal are independently generated. If the sum of the two exceeds digital full scale, clipping will occur.

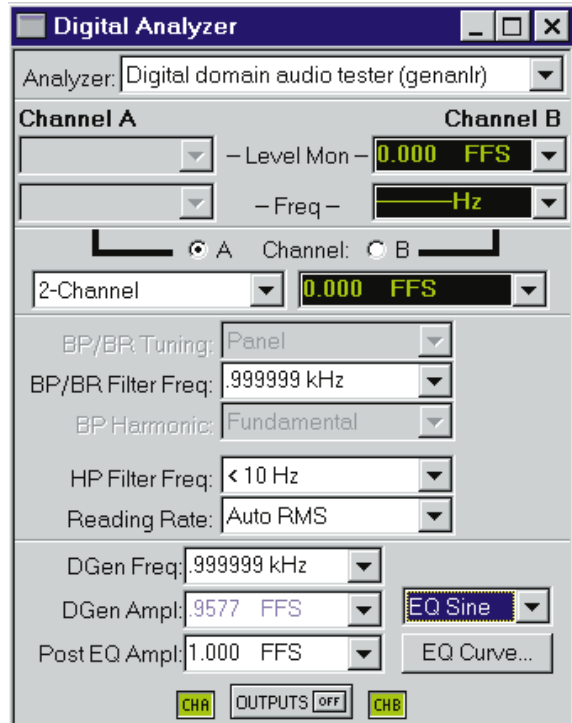


Figure 8-2 GENANLR Panel, Large Version

## EQ Sine—Generator Equalization

Real-time generator equalization of sinewave amplitude can be applied to the digital generator section of GENANLR.

Equalization works by specifying an amplitude vs. frequency curve in the

form of a specific filename and data column within the file. EQ Sine must be selected as the waveform. Equalization function starts with the user-specified amplitude value, looks up (interpolating when necessary) the correction factor in the equalization file for the generator frequency presently in use, and sends the corrected amplitude command (post-EQ amplitude) to the instrument. The Post-EQ field shows the actual output level from the generator circuitry delivered to the external load, after adjusting the user-requested (pre-EQ) value by the value from the EQ file for the generator frequency currently in use. User entries may be made into either normal (pre-EQ) or post-EQ fields and the other field will be adjusted automatically. If a frequency value outside the frequency range span of the EQ file is entered on the generator panel or occurs during a sweep, the post-EQ amplitude will go to zero.

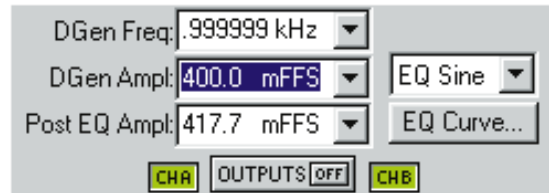


Figure 8-3 GENANLR Generator Section, EQ Sine Mode

### Specifying the EQ Curve

Equalization will not occur until a file and column have been specified and the EQ Sine waveform has been selected. The equalization file may be selected before or after selecting the EQ sine waveform. It is typically faster to first select the EQ Sine waveform, since that results in a message that no file has been selected and the opportunity (by clicking the Yes button in the dialog) to select one. A browser appears which permits navigating to any desired directory and selecting a .ADQ file name. More than a dozen .ADQ files are furnished with APWIN, all located in the C:\APWIN\EQ directory. It is also possible to create your own custom EQ files. After selecting the file, you must select a column that contains amplitude data. The furnished .ADQ files are all two-column files with frequency in column

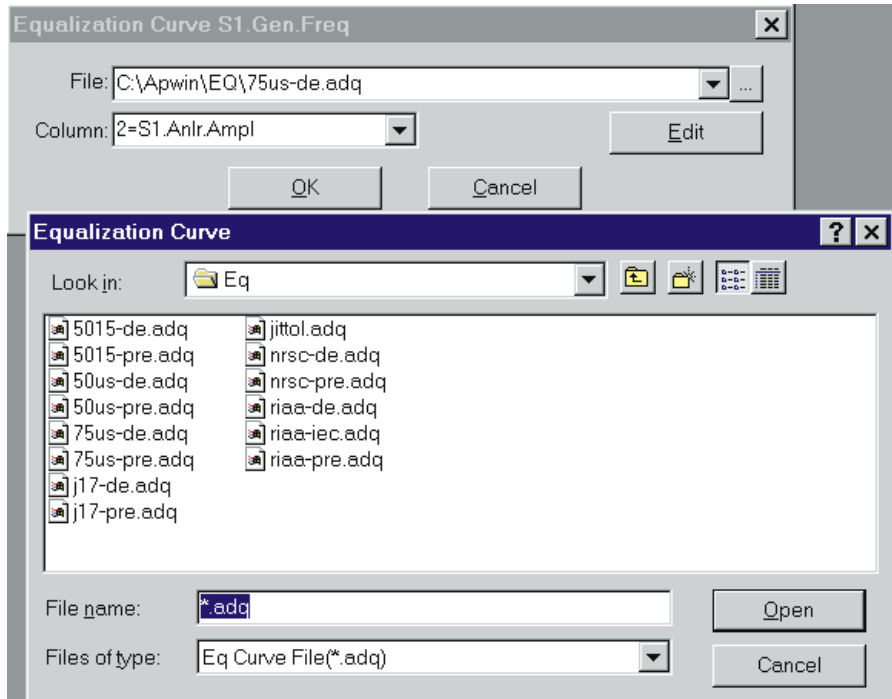


Figure 8-4 EQ File Browser

1 and amplitude in column 2. The “Edit” button on the browser may be clicked to display, and edit if desired, the EQ curve data in the Data Editor.

If you prefer to select the file and column before selecting EQ sine waveform, click on the “EQ Curve” button on the generator panel, which displays the same browser just, described. Actual equalization will not take place until the EQ Sine waveform is also selected. When the EQ Sine waveform is selected, the generator amplitude field or fields will display the setting in blue characters rather than the normal black. If EQ Sine waveform is selected without an equalization file and column having been specified, APWIN software considers the equalization factor to be unity at all frequencies, so no equalization takes place.

## Typical EQ Applications

The most common applications of generator equalization are to produce a nominally-flat output from a device that contains a known equalization function. Typical examples include fm and TV aural broadcast transmitters with 50 and 75 microsecond preemphasis curves, rf signal generators driving broadcast receivers with 50 and 75 microsecond deemphasis characteristics, phonograph preamplifiers with RIAA deemphasis networks, and the recording or playback section (individually) of a tape recorder with its equalization circuits. By connecting an equalization curve to the DSP-implemented generator that is the inverse of the nominal curve used in the device under test, the resulting measured output should be nominally flat. Small variations from flatness will be a measurement of the accuracy of equalization in the device, since the .ADQ curves furnished are accurate to better than 0.01 dB and the specified generator flatness is of the same order.

EQ files may be viewed in tabular format and edited by use of the Edit button in the “EQ Curve” dialog, as discussed above. To view an EQ file graphically, use the File Open EQ command. Note that opening an EQ file does not change the Sweep panel settings, so it may be necessary to change Data 1 units and Optimize the graph for the desired display.

## Resolution (Word Width)

The output word width (resolution) to which the generated 24-bit-wide word will be rounded is set on the Digital I/O panel. The dither on/off control and selection of dither type to be added are also made on the Digital I/O panel. Dither, if used, is automatically set to the correct amplitude for the output resolution setting.

## Measurement Section

The measurement portions of GENANLR consist of a Level Meter, Frequency Counter, and flexible main meter (Reading meter).

## Measurement channel selection

The Channel A and B “radio buttons” visible on both the small and large versions of the Digital Analyzer panel with GENANLR selected determine which channel the main, multi-function meter is connected. The Level meter and Frequency counter are connected to the same channel for some functions and to the opposite channel for other functions, as shown below. The indications of these two meters will be displayed at the left top of the panel when they are measuring the Channel A signal and to the right top when measuring Channel B.

Function	Level & Frequency Channel
2-channel	Opposite to main meter
Ratio	Opposite to main meter
Crosstalk	Opposite to main meter
All other	Same as main meter

## Level monitor

The Level monitor measurement is the top-most display on the Digital Analyzer panel when GENANLR is selected. The left-hand display is used when the Level monitor is measuring Channel A, and the right-hand display used when Channel B is measured. The Level monitor always uses true RMS detection, with a bandwidth extending from approximately 5 Hz to 0.4 times the sample rate (19.2 kHz at 48 kHz, 17.64 kHz at 44.1 kHz, and 12.8 kHz at 32 kHz). Measurement units are selected by clicking the down arrow at the right of the meter display. Only absolute digital domain amplitude units are available—%FS (percent of digital full scale), FFS (fraction of digital full scale), dBFS (decibels relative to digital full scale), and bits.

FS refers to digital full scale, a precisely defined quantity for digital audio signals. Positive digital full-scale is defined as 7FFFFFF Hex, negative full scale is 800000 Hex, and the zero signal baseline is 000000 Hex. Since digital domain amplitude calibration has been defined by the Audio Engineering Society in terms of the RMS value of a sinewave whose peaks just reach those maximum digital codes, it is possible to obtain readings greater than 100% FS (or 1.000 FFS or 0.00 dBFS or 0.00 bits) when measuring high-level signals whose crest factor is less than the crest factor of a sinewave. For example, since a

sinewave has a crest factor of 3.01 dB and a squarewave crest factor is 0.0 dB, a squarewave touching the maximum digital codes will be displayed as 141.4% FS (1.414 FFS, +3.01 dBFS, or +0.5 bits).

## Frequency counter

The Frequency measurement is the second-to-top display on the Digital Analyzer panel when GENANLR is selected. The left-hand display is used when the Frequency counter is measuring Channel A, and the right-hand display used when Channel B is measured. The Frequency counter bandwidth extends from 5 Hz to 0.40 times the sample rate.

## Main meter function selection

The main meter measurement is displayed centrally on the Digital Analyzer panel, just below the input channel A and B selection buttons on the Digital analyzer panel when GENANLR is selected. The units of measure may be selected by clicking on the down arrow at the right of the meter display. The main meter always measures the selected input channel.

The main meter has a flexible set of capabilities for both fixed and tunable filters and has both RMS and quasi-peak detectors. The main meter functions are 2-Channel, Ratio, Crosstalk, THD+N absolute, THD+N relative, Bandpass, Noise-Unweighted, Noise-A Weighting, and Noise-CCIR Weighting.

### 2-Channel

In 2-channel function, the main meter connects to the channel selected by the A or B radio buttons, while the Level monitor connects to the opposite channel. The bandpass/bandreject filter is not used. This function provides simultaneous readings of level on both stereo channels and thus permits frequency response measurements of digital stereo devices in a single sweep. The available units are the digital domain absolute units, %FS, FFS, dBFS, and bits.

## Ratio

In Ratio function, the main meter connects to the channel selected by the A or B radio buttons, while the Level monitor connects to the opposite channel. The bandpass/bandreject filter is not used. The display of the main meter is not the directly-measured amplitude of the selected channel, but is the amplitude ratio (in x/y or % units) or level difference (in dB units) of the two channels. Ratio function thus provides an interchannel balance or differential gain measurement, or a non-frequency-selective crosstalk measurement. Ratio mode works only when the selected channel amplitude is equal to or less than the amplitude on the opposite channel. If the amplitude on the selected channel is greater than the opposite channel, the display will be “clipped” at 0 dB and it will be necessary to select the other channel to obtain a reading.

## Crosstalk

In Crosstalk function, the main meter connects to the channel selected by the A or B radio buttons, while the Level monitor connects to the opposite channel. The bandpass filter processes the signal before the main meter measures it. The display of the main meter is not the directly-measured amplitude of the selected channel, but is the amplitude ratio (in x/y or % units) or level difference (in dB units) of the two channels. Crosstalk function thus provides a frequency-selective crosstalk measurement, permitting accurate crosstalk measurements even when the crosstalk signal is lower in amplitude than the wideband noise level. Crosstalk mode works only when the selected channel amplitude is equal to or less than the amplitude on the opposite channel. If the amplitude on the selected channel is greater than the opposite channel, the display will be “clipped” at 0 dB and it will be necessary to select the other channel to obtain a reading.

## THD + N relative

The THD+N relative function connects both the main meter and the Level monitor to the channel selected by the A or B radio buttons. The bandreject (notch) filter processes the signal to remove the fundamental signal before the main meter measures it. The display of



the main meter is not the directly-measured amplitude of the notch-filtered channel, but is the amplitude ratio (in x/y or % units) or level difference (in dB units) of the main meter with respect to the Level meter, which is measuring the unfiltered input signal. The result is THD+N relative to the input signal.

### THD + N absolute

The THD+N absolute function connects both the main meter and the Level monitor to the channel selected by the A or B radio buttons. The bandreject (notch) filter processes the signal to remove the fundamental signal before the main meter measures it. The display of the main meter is the directly-measured amplitude of the notch-filtered channel and thus does not depend upon the reading of the Level monitor. The available units (obtained by clicking on the down arrow at the right of the main meter display field) are the normal digital domain absolute amplitude units of %FS, FFS, dBFS, or bits. Digital domain THD+N in absolute units is particularly useful when performing amplitude sweeps on a digital I/O device or an A/D converter, where use of relative units obscures the basic noise-limited operation of many devices. An ideal device will exhibit constant THD+N at all amplitudes.

### Bandpass

The Bandpass function connects both the main meter and the Level monitor to the channel selected by the A or B radio buttons. The bandpass filter processes the signal to remove the fundamental signal before the main meter measures it. The display of the main meter is the directly-measured amplitude of the notch-filtered channel and does not depend upon the reading of the Level monitor. The available units (obtained by clicking on the down arrow at the right of the main meter display field) are the normal digital domain absolute amplitude units of %FS, FFS, dBFS, or bits. Digital domain bandpass function is useful when measuring very low-amplitude signals, approaching or below the wideband noise level, as is commonly done when making input/output linearity (amplitude) sweeps on a digital I/O device or an A/D converter.

## Noise Unweighted

The Noise Unweighted function connects both the main meter and the Level monitor to the channel selected by the A or B radio buttons. Neither the bandpass/bandreject filter nor either of the weighting filters is connected in the main meter circuit, but a high-pass filter may be selected. The display of the main meter is the directly-measured amplitude of the selected channel and does not depend upon the reading of the Level monitor. The available units (obtained by clicking on the down arrow at the right of the main meter display field) are the normal digital domain absolute amplitude units of %FS, FFS, dBFS, or bits. The Noise unweighted function in the digital domain corresponds to band-limited, unweighted noise measurements in the analog domain. The Noise unweighted function may also be used with the 400 Hz high-pass filter selected and stimulus from a low-frequency signal to make quantization noise and distortion measurements on digital signals.

## Noise A Weighting

The Noise “A” weighting function connects both the main meter and the Level monitor to the channel selected by the A or B radio buttons. The DSP-implemented A-weighting filter is connected in the main meter circuit. The DSP-implemented true RMS detector should be selected, typically at a low reading rate such as 4/second, in the Reading Rate field. The display of the main meter does not depend upon the reading of the Level monitor. The available units (obtained by clicking on the down arrow at the right of the main meter display field) are the normal digital domain absolute amplitude units of %FS, FFS, dBFS, or bits.

## Noise CCIR Weighting

The Noise CCIR Weighting function connects both the main meter and the Level monitor to the channel selected by the A or B radio buttons. The DSP-implemented CCIR-468 weighting filter is connected in the main meter circuit. For measurements conforming to the CCIR-468 standard, the DSP-implemented quasi-peak detector should be selected by the 4/sec QPK selection in the Reading Rate field. The CCIR filter maximum gain will then be 12.2 dB as specified in CCIR

Recommendation 468-4. This results in a 1 kHz unity gain frequency. If the RMS detector is selected, the unity gain frequency moves to 2 kHz as normally used in CCIR/ARM measurements. This is equivalent to the CCIR-2K selection on the analog analyzer.

The display of the main meter does not depend upon the reading of the Level monitor. The available units (obtained by clicking on the down arrow at the right of the main meter display field) are the normal digital domain absolute amplitude units of %FS, FFS, dBFS, or bits.

### **Bandpass/bandreject filter tuning**

The bandpass or bandreject (BP/BR) filter of GENANLR is tunable across the audio spectrum. In Bandpass or Crosstalk functions it may be tuned to any center frequency from 0.04% to 40% of the sample rate; for example, from 20 Hz to 19.2 kHz at the 48 kHz rate. The filter is a 10-pole design with a nominal 1/13 octave bandwidth (Q of 19, -3 dB bandwidth of approximately 5.2% of center frequency). The response will become non-symmetrical as the center frequency approaches 20 kHz. In THD+N modes, the filter tunes from 0.1% to 40% of the sample rate; for example, from 50 Hz to 19.2 kHz at the 48 kHz rate. Several modes are available for steering this filter frequency in order to simplify swept measurements. The five selections in the BP/BR Tuning field are Panel, DGen, Dig Freq, Anlr BP/BR, and Anlg Gen.

In **Panel** mode, the BP/BR filter is fixed-tuned at the frequency entered into the BP/BR Filter Freq field just below the BP/BR Tuning field. The Digital Analyzer panel must be displayed in its large version for the BP/BR Filter Freq field to be visible. This tuning mode is often used when performing linearity tests (wide range amplitude sweeps) of A/D converters, since the digital frequency counter of the GENANLR program will not be operational at extremely low signal levels.

In **DGen** tuning mode, the BP/BR filter center frequency is automatically tuned to the frequency of the sinewave generator of GENANLR, whether fixed in panel mode (at the bottom of the large version of the Digital Analyzer panel) or being swept as the Source 1 or Source 2 parameter. This tuning mode provides the most rapid and

reliable operation during frequency sweeps of digital input-digital output devices where GENANLR also provides the test signal.

In **Dig Freq** tuning mode, the BP/BR filter is tuned to the frequency being measured by the digital domain frequency counter as displayed near the top of the Digital Analyzer panel. This tuning mode is convenient for all types of digital domain measurements as long as the digital signal level is high enough to provide reliable, noise-free operation of the digital domain frequency counter, but will not be quite as fast in settling as the DGen mode.

In **Anlr BP/BR** tuning mode, the DSP-implemented digital domain BP/BR filter tracks the frequency of the Analog Analyzer hardware BP/BR filter.

In **Anlg Gen** tuning mode, the DSP BP/BR filter tracks the frequency of the Analog Generator. This is useful when measuring A/D converters.

In Bandpass function, the bandpass filter may also be automatically tuned to the 2nd, 3rd, 4th, or 5th Harmonic of the steering source instead of Fundamental in the BP Harmonic field. This permits automatic tracking of a specific harmonic from 2nd through 5th.

## Highpass filters

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The HP Filter Freq field is visible on the large version of the Digital Analyzer panel when GENANLR is selected. Clicking on the down arrow at the right of this field permits selection of four high-pass filter choices, at 10 Hz, 22 Hz, 100 Hz, and 400 Hz. High-pass filters, when selected, are connected in the main meter measurement path; they do not affect the Level monitor or the Frequency counter.

The 22 Hz, 200 Hz, and 400 Hz high-pass filters are two-pole (12 dB/octave) designs in either THD+N function. When used with the Noise A-weighted or Noise-CCIR weighted functions, they become four-pole maximally-flat designs.

In Noise Unweighted, Ratio, or 2-channel functions, the 22.4 Hz and 100 Hz filters remain four pole while the 400 Hz filter becomes an extremely sharp, high-rejection 400 Hz high-pass filter (ten-pole

elliptical). The 400Hz filter selection in these functions permits quantization noise and distortion measurements of A/D converters and digital systems. Quantization noise and distortion measurements are typically made by driving a digital system at its full input amplitude range with a low-frequency sinewave (typically below 50 Hz) whose frequency is not integrally-related to the sampling rate. This signal exercises the A/D converter while the filter attenuates the fundamental and all harmonics below approximately 220 Hz by at least 120 dB, but passes wideband noise above 400 Hz unattenuated.

## Reading Rate and Source

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The Reading Rate field, visible on the large version of the Digital Analyzer panel when GENANLR is the selected DSP program, selects the detector response and controls the update rate (integration period) of all three meters of this program. Selections include Auto with RMS detection, fixed rates from 4/sec through 64/sec with the RMS detector, and a 4/sec rate with a quasi-peak detector which complies with CCIR recommendation 468-4 and earlier.

The Auto RMS selection manages selection of the reading rate as a function of the frequency being measured and the instrument function so as to provide rapid testing speeds along with sufficient integration for accuracy at the present test frequency. Fixed rates may be selected in order to gain the maximum possible resolution and accuracy, even at a compromise in testing speed.

## Auto Reading Rate Source

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In order for the algorithm controlling the Auto rate selection to know what the present test frequency is, the Auto Read Rate Source field under it is provided. The selection in this field determines the information source to the Auto algorithm. The four selections are Dig Freq, DGen, Anlr BP/BR, and Anlg Gen.

The **Dig Freq** selection passes measurement values from the GENANLR Frequency counter to the Auto algorithm.

The **DGen** selection uses the GENANLR sinewave oscillator frequency as input to the algorithm.

**Anlr BP/BR** uses the present frequency of the Analog Analyzer bandpass/bandreject filter to control the Auto algorithm.

**Anlg Gen** uses the Analog Generator frequency.

Generally speaking, the parameter selected to be swept at Source 1 of the Sweep panel should also be selected as the Auto Read Rate Source.

# Digital Data Analyzer

The Digital Data Analyzer is designed for bit error measurements of imbedded audio signals.

To load the Digital data analyzer (BITTEST) program, select “Digital data analyzer (bittest)” at the top of the Digital Analyzer panel.

## Introduction

This program generates imbedded digital audio test signals and measures the returned imbedded digital audio signals for bit errors. It can also control transmission of the Data Valid/Invalid bit. BITTEST operates at 48 kHz, 44.1 kHz, and 32 kHz sample rates. Only these three sample rates may be used with BITTEST. Sample rate is selected on the Digital I/O Panel, not on the BITTEST panel. The imbedded audio test signal may be a pseudo-random sequence, constant valued samples (“digital dc”), a sinewave of selectable amplitude and frequency, or walking bit patterns. Generated word width in BITTEST is always 24 bits. Measurement word width is selectable for all signals, in the Resolution field of the Digital I/O panel. The measurement may display both real-time received data and errors in the received data sequence. Any amount of delay between transmitted and received signals is permissible, allowing testing of devices and transmission links with large amounts of delay or even recorder-reproducers. No dither is added to any of the signals.

The extensive signal generation and error measurement capability of BITTEST is useful for investigating the integrity of digital audio data links, recorders, etc. It is also invaluable for design test of digital interfaces. Each waveform in the program has a specific testing application. BITTEST operates only with digital domain input and output.

**BITTEST**

constantly generates signals and measurements can be continuously observed in panel or bargraph modes. This is particularly useful for making adjustments on devices under test.

Some of the test signals use the rear-panel

TRIG OUT BNC connector to trigger an oscilloscope or logic analyzer. Signal parameters, where adjustable, may be swept as a Source 1 parameter, and all measured parameters may be plotted as Data n on X-Y graphs.

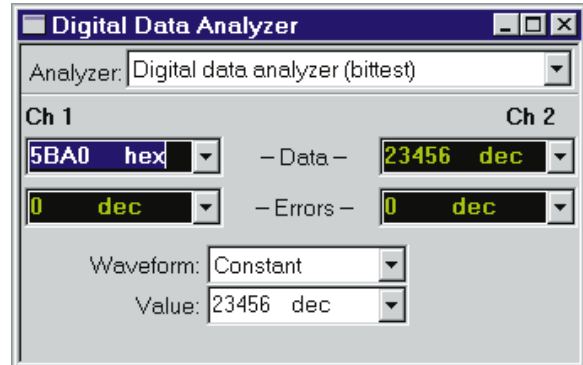


Figure 9-1 BITTEST Panel, Small Version

## Signal Generation

Digital-domain signals of five different waveforms may be generated by BITTEST. The output signal is always of 24-bit resolution. To change generated waveforms, click on the down arrow at the right end of the Waveform field and select the desired waveform. The selections are Constant, Random, Walking-1, Walking-0, and Sine. Note that the Waveform field also tells the BITTEST analyzer which waveform to expect for error analysis. If, for example, a Random waveform had been recorded on a digital recorder and the recording then played back for error analysis, the Waveform field must be set to Random even though the generator function of BITTEST is not required or used when analyzing a playback signal.

### Constant

The Constant mode generates a continuous stream of data samples at the same fixed value. This is the digital equivalent of a DC voltage source. The data word to be output is set by the VALUE field on the panel and may be the Source parameter in a sweep. The value may be entered in decimal or hexadecimal notation. Constant mode aids in the investigation of data-dependent errors in digital systems. No TRIG



OUT signal is generated in this mode. The DGen Freq and DGen Ampl fields have no effect with Constant waveform.

The Value entered in Constant mode is with respect to the word length entered in the Resolution field of the Digital I/O panel. Entering a Value of 1 with Resolution of 16 bits, for example, produces a constant stream of samples with the binary value:

0000 0000 0000 0001.

(Actually, a 24-bit word is generated with the remaining eight least bits set to zero, so the actual binary output word would be

0000 0000 0000 0001 0000 0000.

Only the 16 most significant bits (MSBs) would normally be connected to a 16-bit device under test.) If the Resolution field is changed to 24 with the Value remaining at decimal 1, the binary output word would now be

0000 0000 0000 0000 0000 0001.

## Random

The Random mode transmits a pseudo-random sequence uniformly distributed between plus and minus full scale. The amplitude is not adjustable. This signal is intended for error rate testing of communications links, AES/EBU interfaces, and digital recorders. No TRIG OUT signal is output in this mode. The DGen Freq, DGen Ampl, and Value fields have no effect with Random waveform.

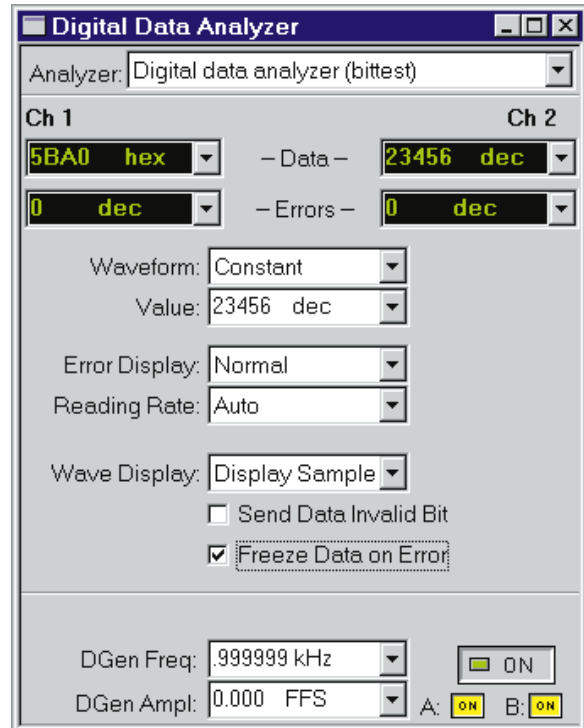


Figure 9-2 BITTEST Panel, Large Version

## Walking Bit

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There are two walking bit modes. The Walking-1 mode sets all bits to 0 except one bit, which is set to 1. This single high bit is continuously incremented from lower to upper bits. When it reaches the MSB it will wrap around to the LSB. The Walking-0 mode sets all bits to 1 except one bit, which is set to 0. This single low bit is continuously incremented from lower to upper bits. When it reaches the MSB it will wrap around to the LSB. The bit position is incremented each sample interval, exercising all bits in 24 samples. The amplitudes of the walking bit signals are therefore not adjustable. These signals help identify shorted data lines and crossed bits in digital audio circuits. Whenever the signal wraps around from the MSB to the LSB the rear panel TRIG OUT output will pulse. This allows triggering an oscilloscope on the pattern for viewing data in the device under test. The DGen Freq, DGen Ampl, and Value fields have no effect with walking bit waveforms.

## Sinewave

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The Sine waveform allows easy checking of gain and proper interface connection. Some digital audio devices apply gain or equalization to the signal, which makes use of bit-pattern-based test signals difficult. The sinewave will retain its shape passing through any linear digital device. Thus, only the sinewave is useful for error testing when the device under test has gain or equalization. The sinewave frequency is set with the DGen Freq field of the panel. The amplitude is set with the DGen Ampl field. The Value field has no effect in sinewave mode.

## Validity Bit

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The validity (V) bit on the AES/EBU and SPDIF/EIAJ interfaces may be set or cleared as desired to test the behavior of equipment in response to this bit. When the “Send Data Invalid Bit” checkbox is checked, the V bit is asserted (Invalid). When the box is not checked, the V bit is not asserted (Valid). Some equipment is designed to mute when the Invalid condition appears at its digital input. Digital recorders commonly set their output bit to Invalid when the tape is not playing

and to Valid when the tape is in motion. The selected validity bit will be generated in every sample until changed on the BITTEST panel.

## Signal Analysis

Received data is displayed on the Ch 1 and Ch 2 Data fields at the top of the panel, truncated to the word length set in the Resolution field of the Digital I/O panel. Ch 1 refers to digital Channel 1 (subframe A) and Ch 2 refers to digital Channel 2 (subframe B). BITTEST cannot be used with analog signals. The Data may be displayed in decimal or hexadecimal units, but decimal is recommended unless the Resolution field on the Digital I/O panel is set to 22 bits or less. The Data (and Errors) readings are updated at the rate selected in the Reading Rate field (Auto/Slow/Fast) until an error occurs. When an error occurs (see below), the display function depends upon the setting of the “Freeze Data on Error” checkbox. If the “Freeze” checkbox is checked, the Data fields will continue to display the value that was received when the first error occurred. If the “Freeze” checkbox is not checked, the Data fields will continue updating, independent of any errors detected.

Received data is also measured to determine if it matches the data transmitted. Only the number of bits selected in the Resolution field of the Digital I/O panel will be analyzed. This comparison is done with algorithms that are insensitive to delay between the send and receive sections. The number of errors in the received data per measurement interval are counted for each channel. The Error Display field selects the type of analysis to be performed. In the Normal mode, the number of errors detected during the last measurement interval (1/4 second in Slow or 1/32 second in Fast Reading Rate) are displayed directly in the Ch 1 and Ch 2 Errors fields of the panel. If Error Display is selecting as Maximum, the maximum error count during any measurement interval will be held in the display. Using the Totalize mode of the Error Display field will accumulate a running total of all errors. To reset the Totalize or Maximum error displays, press Enter with the cursor on the Error Display field.

All modes make error measurements based on the Waveform selection. For example, Random must be selected as the Waveform

when measuring a digital reproducer playing a recording previously made with the Random signal from BITTEST, Sine must be selected to measure errors from a sinewave recording, etc. For error checking with a Constant waveform signal, Constant waveform must be selected and the Value field must further have the specific value entered which was used when the recording was made. The random, sine, and walking bit modes compare each successively received sample with the algorithm used to generate the signal. Two samples are required before the error analysis synchronizes with the signal, so errors may be indicated during the initial measurement interval even though no error actually exists.

The Resolution field on the Digital I/O panel sets the number of input bits to be analyzed in all modes. It affects both data and error displays. The Resolution field also defines word length for the Value entry in Constant waveform.

### **Error Detection Flag**

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Whenever an error is detected the DSP will pulse the AUXILIARY OUTPUT (pin 11) on the rear panel 15-pin D-sub serial interface connector. This allows triggering an oscilloscope to observe received data errors. Since the error indication does not occur until after the end of the data word containing the error, a digital storage oscilloscope or logic analyzer is recommended. That instrument should be set in pre-trigger mode to acquire data before the trigger event.

## FFT Overview

Six of the DSP programs furnished with System One and APWIN are FFT-based programs. Selection of any of these programs is accomplished at the top of the Digital Analyzer panel. These FFT-based programs are FFTSLIDE, FFTGEN, FASTTEST, FASTTRIG, CODEC, and MLS. FFTSLIDE and FFTGEN are general-purpose spectrum analysis and waveform display programs. Their operational details are treated in the following chapter. FASTTEST, FASTTRIG, and CODEC are all designed for analysis via multitone techniques. They are described in detail in the Multitone Analysis chapter. MLS is significantly different from the other five in operation and application and is treated in the MLS chapter.

### FFT Fundamental Concepts

A Fast Fourier Transform (FFT) is a mathematical process for deriving the amplitude and phase spectrum (amplitude versus frequency and phase versus frequency) from a sampled waveform (amplitude versus time) of a signal. Certain simple, fundamental relationships exist between the number of samples (transform length) making up the acquired waveform, the sample rate, and the frequency resolution of the resulting amplitude and phase spectra. The fundamental bandwidth-versus-sampling rate relationship of any sampling process is also relevant.

**Bandwidth = 1/2 sample rate;** for example, with a 48 kHz sample rate, the maximum signal frequency is 24 kHz. When analog signals are sampled and converted by A/D converters, the anti-alias filters used with the converters may further reduce the effective bandwidth.

**Frequency resolution = Sample rate / record length;** for example, with a 16,384 sample waveform and a 48 kHz sample rate, the frequency resolution (also called **FFT bin width**, or **line spacing**) is approximately 2.93 Hz.

An FFT produces frequency-linear results; the resolution is constant across the bandwidth from DC (0 Hz) to 1/2 the sample rate. For example, with a 48 kHz rate and 16,384 samples, there will be 8,192 bins (lines), each 2.93 Hz wide, extending from 0 Hz to 24 kHz. The results of an FFT may be graphed on a logarithmic frequency scale if desired, but the actual FFT resolution is linear and thus equal across the spectrum.

The signal to be transformed is first acquired into a memory buffer called the acquisition buffer. The entire available acquisition buffer length is not necessarily filled each time signal is acquired. The amount of signal actually acquired into this buffer is called the record length.

<b>Program</b>	<b>Acq.buffer length, min mem</b>	<b>Acq. time at 48 kHz, min mem</b>	<b>Acq.buffer length, full mem</b>	<b>Acq. time at 48 kHz, full mem</b>
FFTSLIDE	8192	171 ms	30720	640 ms
FFTGEN	8192	171 ms	16384	340 ms
FASTTEST	8192	171 ms	16384	340 ms
FASTTRIG	N/A	N/A	16384	340 ms
CODEC	N/A	N/A	16384	340 ms

In time domain (oscilloscope mode) analysis, any portion up to and including the full acquisition buffer length may be displayed. For FFT spectrum analysis, the portion of the acquisition buffer to be is limited to several discrete sample length selections which are binary powers—256, 512, 1024, 2048, 4096, and if maximum DSP memory is present, 16,384 samples long.

## System One FFT Function

Figure 10-1 shows a functional block diagram of System One DSP FFT programs. Only DSP Channel 1 is shown in detail; Channel 2 is identical. Not all the processing modes and window choices shown are available on each program.

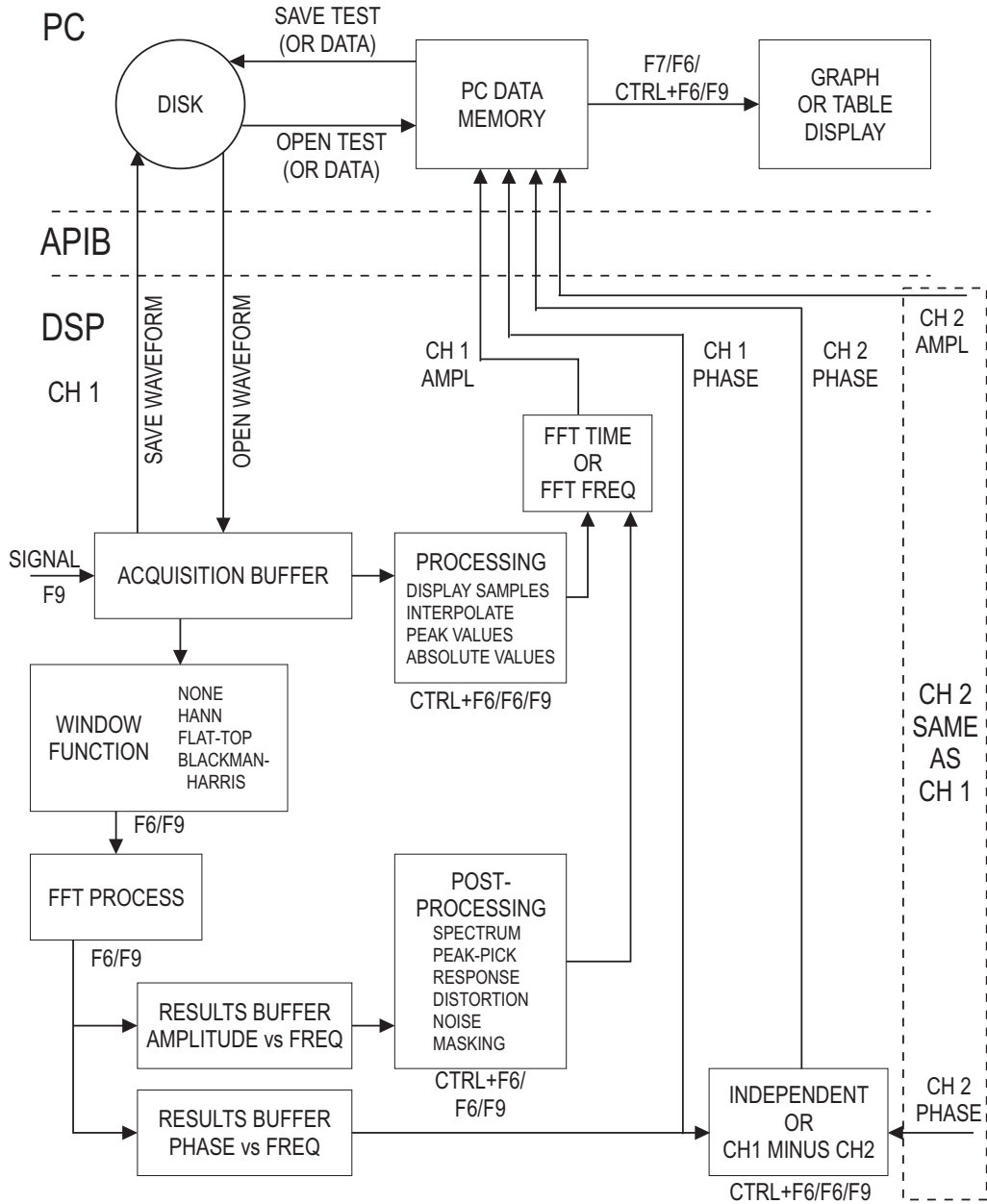



Figure 10-1 Function Block and Process Diagram, System One FFT Programs

Analog (via A/D converters) or digital signal samples are written into the Acquisition Buffers when the  function key is pressed (or equivalent action).




The acquired signal may be viewed in time or frequency domain mode, selected by the choices of FFT Time versus FFT Freq at Source 1 on the Sweep panel.

## Time Domain

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When time domain (oscilloscope) view is selected (FFT Time at Source 1), the amplitude values of the samples in the Acquisition Buffers are sent via the Processing block to the computer for display. The Start to Stop time span at Source 1 on the Sweep panel determines which section of the Acquisition Buffer is processed and sent to the PC. The processing choices include a “no processing” mode (Display Samples) and three processing modes, described in detail later in this chapter. The waveform is displayed by selecting Ch 1 Ampl and/or Ch 2 Ampl at any of the Data 1 through Data 6 browsers on the Sweep panel. Waveforms are typically displayed using a non-decibel unit (Volts or FFS or %FS) rather than a decibel unit, a linear vertical scale, and with the graph top and bottom values symmetrically located above and below zero (ground).

## Display Changes

Changes may be made in the selected units or graph top and bottom values and displayed most rapidly by the  key. This simply re-graphs the present contents of the computer Data Memory without involvement of the DSP unit at all. Changes in the time span viewed (often by zooming with the mouse), displayed resolution (by changing the number of Steps), and processing mode are made most rapidly without acquiring signal again by use of the   keystroke. This causes a re-processing and transmission of the processed data to the computer for display, but no new signal acquisition or FFT transform.

## Frequency Domain

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When frequency domain (spectrum analysis) view is selected (FFT Freq at Source 1), all or a selected portion of the Acquisition Buffer will be Fourier transformed, depending on panel settings. The record to be





transformed may be multiplied by a window function or (typically with synchronous signals) no window may be used. The FFT process results in magnitude (amplitude versus frequency) results. Some of the FFT-based programs also produce phase (phase versus frequency).

The Amplitude vs. Frequency information may be processed by one of several techniques, depending upon the FFT program and the user's needs. These post-processing techniques are described in the individual chapters. The results are displayed by selecting Ch 1 Ampl or Ch 2 Ampl at any of the Data 1 through Data 6 browsers on the Sweep panel. Spectrum analysis is typically displayed either in a non-decibel units (Volts, FFS, or %FS) with a logarithmic vertical scale, or with a decibel unit.

## Display Changes


Changes may be made in the displayed units or graph top and bottom values and re-graphed most rapidly by the  key. This simply re-graphs the present contents of the PC Data Memory without involving the DSP. Changes may be made in the window function, transform length, and (for some of the programs) transform starting point and displayed most rapidly by the  keystroke rather than acquiring signal again. The  key causes a new FFT transform and sends data to the computer but no new acquisition. Changes may be made in the post-processing selections most rapidly without either acquiring signal or transforming signal again via the  keys. Examples of post-processing changes include going from response to distortion to noise with a multitone signal or changing frequency span displayed by Source 1 Start and Stop. Other examples are changing graph top and bottom values and changing displayed resolution via number of Steps. The  keys re-process the existing FFT results according to the present settings on the Digital Analyzer and Sweep panels without either acquiring signal or performing a new FFT. Zooming in or out on the graph automatically sends a  command to the DSP unit unless this feature has been disabled by the Utilities Configuration command (Reprocess FFT Data on Zoom checkbox).

## Saving and Loading Waveforms

The complete contents of the Acquisition Buffers, or smaller sections of one or both channel buffers, may be saved to computer disk for a later download to the DSP and continued re-analysis. “Before and after” results may be viewed by saving a waveform to disk from one set of conditions, changing the device or configuration and saving new results, and then downloading one saved waveform into the Ch 1 DSP Acquisition Buffer and the other into Ch 2. The two buffers may then be compared in time or frequency domain, using the  or  keystrokes described above as appropriate. See the Saving and Loading Waveforms section of each FFT-based program chapter for details.


## DSP Memory vs. PC Memory

The full set of original signal samples remain in the Acquisition Buffer until:

-  is pressed again, or
- a different waveform is downloaded from computer disk, or
- the DSP program is changed (selection in Analyzer field at top of Digital Analyzer panel is changed), or
- power to System One is turned off.

The spectrum analysis results remain in the Results Buffers until one of those same events, or

-  is pressed again.

Only a processed (and usually very abbreviated) version of the data is ever in APWIN Data Memory in the computer. For example, assume a 16384 sample Acquisition Buffer and a 48 kHz Rate. Following , the Acquisition Buffer therefore contains over 341 milliseconds of signal represented with 48 samples per millisecond. If Sweep Source 1 is set to a Start time of zero, a Stop time of 2 milliseconds (typical if viewing a 1 kHz signal), and a Steps value of

200, the computer Data Memory will contain 201 amplitude values at 10 microsecond intervals. Changing the Stop time to 10 milliseconds and pressing the **F7** key (re-graph contents of computer Data Memory) only shows signal in the first 2 milliseconds, since nothing beyond that was sent from DSP to computer since the Sweep Stop value was 2 milliseconds when **F9** was pressed. Pressing **CTRL F6** (or **F6**) will then send data from the first 10 milliseconds of DSP Acquisition Buffer to the PC. If the Steps value was not changed, there will still be 201 amplitude values but now spaced at 50 microsecond intervals. If the test is saved (or data is saved) and later re-loaded, only the representation of the data which was in computer Data Memory will be available for view. If complete flexibility is desired in order to change between time and frequency domain views, zoom to the limits of the FFT or time sample resolution, or perform FFTs of different portions of the record, the entire acquired waveform should be saved to disk by the Save Waveform command. It can then later be downloaded to the DSP via Open Waveform and analysis continued.

### Analog/Digital Input Selection

All the FFT-based programs can acquire digital signals (two channels) directly from any of the digital interfaces of System One Dual Domain, and analog signals (two channels) via built-in 16-bit A/D converters. The Input control near the top of the Digital Analyzer panel permits selection of **Digital** or **A/D**. Click on the down arrow at the end of the field, then click on the desired input domain.

## Time Domain Display (Oscilloscope)

In time domain (storage oscilloscope mode), the Start time, Stop time, and Steps (or Multiply or Stepsize) values on the Sweep panel may be set independently of the FFT Sample Rate. It is thus possible to view only a millisecond or two (48 to 96 samples at the 48 kHz sample rate) spread across the graph, in which case there are many plotted points per sample. It is equally possible to view several hundred milliseconds of signal (up to 30,720 samples with FFTSLIDE or 16k samples for the other programs with full DSP memory) plotted across the graph with typically a few hundred plotted points, which results in one plotted point representing many tens of samples. This

flexibility in graphing different plotted point-to-sample relationships can also produce problems in some situations, with several tools available to produce the best results.

## Graphic Aliasing

Graphic aliasing is a phenomenon that may occur in time domain presentations (oscilloscope view) of signals when there are a low number of points displayed per cycle of signal waveform. The result may appear to be a signal of a completely different frequency than the actual signal—even DC. For example, assume that a 1 kHz sinewave (1 millisecond period) signal has been acquired at a 48 kHz sample rate. Each cycle of signal is represented in the acquisition buffer by 48 samples. If the combination of Start and Stop times and the Steps values at Source 1 result in many plotted points per cycle of signal waveform, the representation on the graph will be accurate (particularly in the Interpolate waveform display option is chosen on the Digital Analyzer panel). If the time span from Start to Stop covers many cycles of signal and only a few points are plotted per cycle, the display can be very misleading. For example, if the Start to Stop span is 200 milliseconds and the Steps value is 200, one point will be plotted per cycle of waveform in the DSP buffer. The result would be an apparent DC signal rather than a sinewave, since the one point plotted per cycle would be taken at the same phase angle from each cycle.

To help insure freedom from graphic aliasing, start out either with small time spans and relatively large numbers of Steps, or use the Peak Values or Absolute Values of the “Wfm Display” field on the Digital Analyzer panel. These selections search through the entire range of samples in the DSP buffer between plotted points and return the highest value in that range to the computer for plotting, assuring that signal peaks cannot be missed. It may also be helpful to use a prime number such as 177 or 377 as the Steps value at Source 1 to reduce the likelihood of a synchronous relationship between this value and the signal period.

## Frequency Domain (Spectrum Analysis)

The process of spectrum analysis of signals with System One DSP units consists broadly of three steps: acquiring the signal into the acquisition buffer, computing the FFT and putting the results into an amplitude-versus-frequency buffer, and displaying the amplitude-versus-frequency results on the computer screen. Depending on the application, there may also be some pre-processing of the acquisition buffer contents before the FFT is computed and some post-processing of the spectrum analysis results before they are graphed. All the FFT programs can multiply the time domain signal by a window function before performing the FFT. Post processing can include “peak picking” when there are more FFT bins than plotted points on the graph, plus specialized multitone signal post-processing for fast measurements of response, distortion, noise, phase, and computation of psychoacoustical masking curves.

## FFT Windowing Concepts

The fundamental mathematics behind a Fast Fourier Transform (FFT) makes the assumption that the portion of an acquired signal being transformed is a perfect, synchronous section of a signal which continues indefinitely. This is the same as assuming that if the section of signal in the transform buffer were spliced from the last sample to the first sample to form a loop and then “played back” continuously from that loop, the resulting signal would be identical to the acquired signal. In the real world, this is virtually never the case. Unless a test signal is deliberately generated in such a fashion that it goes exactly through some integer number of cycles in the length of the analyzer acquisition or transform buffer, there will be an amplitude discontinuity at the “splice”. Such a discontinuity results in energy spread across the entire spectrum, raising the baseline of the spectrum analysis to such high values that all but the strongest signals are buried.

The solution to this problem for typical, non-synchronous signals is to multiply the amplitudes of all the samples in the transform buffer (amplitude versus time) by a “windowing” function before the FFT is computed. The windowing function has a value of unity (1.000) at the center of the buffer and drops to zero at the ends. The resulting

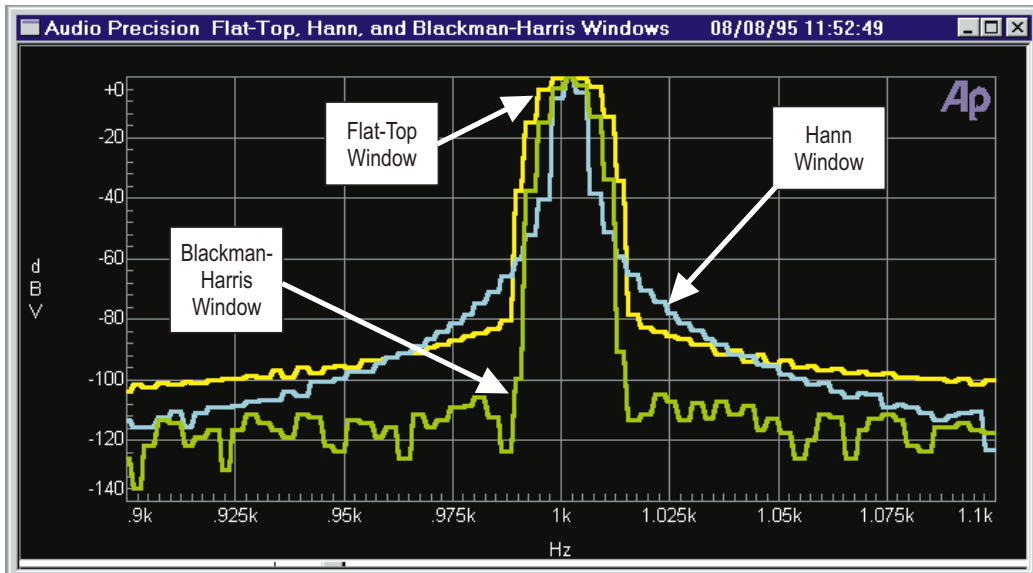



Figure 10-2 Selectivity Curves of Flat-Top, Hann, and Blackman-Harris Windows

“windowed” amplitude versus time signal will always splice smoothly without discontinuities, since the zero values at both ends match. The resulting FFT has greatly reduced signal “spillover” across the spectrum. However, the “windowed” function is clearly not exactly the same as the original signal. The FFT of the windowed function is different in two ways from a theoretical synchronous FFT (if that were possible) of the original signal before windowing. First, some signal spillover (spreading) into several adjacent and nearby frequency bins above and below the actual signal frequency will occur, obscuring any possible very low amplitude signals at very nearby frequencies. Second, the measured amplitude of any signal component will be somewhat lower than the true value unless that signal lies exactly in the center of an FFT bin. Bin centers are at frequencies exactly determined by sample rate and record length, and signal components being measured may fall at any point across a bin.

The available window selections for FFTSLIDE, FFTGEN, and FASTTEST are Blackman-Harris, Hann, Flat-top, and None. FASTTRIG and CODEC offer None and Hann. MLS has specialized windows designed for use with acoustical signals. Each available

window function has a different set of trade-offs in effective close-in selectivity (signal spillover into nearby FFT bins) and in amplitude measurement error if the signal does not fall at an exact bin center frequency. See Figure 10-2. The Hann window has the best selectivity near the top and its skirts fall off monotonically. The Blackman-Harris window has better selectivity than the Hann except for the few bins centered on the signal, and has side lobes at about -90 dB and below. The Flat-Top window has the poorest selectivity, but virtually no amplitude error across the center three bins.

To select a window, click on the down arrow at the end of the FFT Window box and click on the desired windowing function. Note that the window function is applied after signal acquisition and before the actual FFT transform. It is thus simple to change FFT Window selections and re-transform the data with the  key, without making another acquisition. With the Append box on the Sweep panel checked, FFTs of the same signal with several different window functions applied can be graphically compared.

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## Hann Window

The Hann window is a raised cosine window named after its inventor, Austrian meteorologist Julius von Hann. It provides good selectivity near the top, with no side lobes. Its skirts more than 3 bins off center are not as steep as the Blackman-Harris window. The Hann window causes approximately a -1.5 dB maximum amplitude error due to window attenuation if the signal is at the extreme edge of the bin.

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## Blackman Harris Window

The Blackman-Harris 4-term minimum sidelobe window was developed by R.B. Blackman and F.J. Harris. Compared to the Hann window, it is not quite as selective across the central several bins but has steeper skirts beyond that point. The Blackman-Harris window has sidelobes below -92 dB (response fall-off is not monotonic). It has a reasonably flat top with a maximum amplitude error of about -0.8 dB if the signal is at the extreme edge of the bin.

## Flat-Top Window

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The Flat-Top window is designed for the greatest amplitude measurement accuracy. It provides a maximum amplitude error due to window attenuation of less than 0.02 dB even if the signal is at the extreme end of the bin. However, its selectivity is poorer than either Hann or Blackman-Harris. The Flat-Top window is the appropriate window for accurate amplitude measurements (such as when measuring individual harmonics) except when signals are so closely spaced that its selectivity becomes a problem. For example, the 2.93 Hz bin width of a 16,384 sample FFT at the 48 kHz sample rate would permit accurate measurements of signals differing by nearly 90 dB in amplitude as long as they are at least 26.4 Hz (9 bins) apart.

## No Window (Rectangular)

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The “No Window” choice applies no window function before the FFT. It is sometimes called a rectangular window. The “No Window” choice is normally used only with synchronous signals in FASTTEST, FASTTRIG, and CODEC, and is not generally useful for non-synchronous signals since energy will appear to be spread across the entire spectrum unless the signal is exactly at bin center. The amplitude error due to window attenuation is about 4 dB when the signal is at the extreme edge of a bin.

## Post-FFT Processing

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At the conclusion of the Fast Fourier Transform, magnitude-vs.-frequency data arrays exist in DSP memory for both signal channels. In the multitone analysis programs and MLS, phase-vs.-frequency arrays also exist. Generally speaking, the entire contents of those arrays are not simply sent to the computer for display. The user-set Start and Stop frequency values on the Sweep panel may call for only a portion of the data to be sent to the computer. The number of Steps on the Sweep panel may be greater or smaller than the number of FFT bins in the Start-Stop frequency span, with either condition requiring some post-FFT processing. With FFTGEN, a spectrum averaging feature is available to improve the signal to noise ratio of the displayed spectrum. In multitone analysis, different forms of processing are used in order to extract frequency




response, phase response, total distortion and noise, noise in the presence of signal, and (in the case of CODEC) psychoacoustical masking curves from the FFT data.

## Displayed Points Do Not Equal FFT Bins

The “Maximum” FFT Length is 16384 samples with full memory, which results in 8192 FFT bins (spectral lines). Since 1024 pixels is a rather high graphic resolution and the portion of the screen dedicated to the graph is only a portion of that number, it is typical for the number of Steps at Source 1 on the Sweep panel to be set to 500 or less. Thus it is common to have many FFT bins represented by one displayed point. In order to avoid missing significant signals, the DSP employs a form of post-FFT processing known as “Peak Picking”. When the computer requests data at two points which span more than one FFT bin, the DSP scans across all FFT bins between the two requested frequency points, finds the greatest bin amplitude, and sends that value to the computer to be graphed as the amplitude of the new point.


When the user has set Start and Stop frequencies to a small span, or zoomed on the graph into a small frequency span, the opposite situation may occur—there may be many displayed points per FFT bin. The computer may be thus request data several times within the same FFT bin. In this case, the DSP simply returns the same bin amplitude value to each request that falls within the bin boundaries. With the typical 48 kHz sample rate and maximum 16384 sample FFT, for example, the width of each FFT bin is about 2.93 Hz. If the graph has been zoomed in to a 100 Hz linear span with 500 steps, approximately 15 displayed points are derived from each bin. The resulting graph has the appearance of a city skyline with rectangular flat-topped sections representing the width and amplitude of each bin.

## Transform Data without Acquire

FFT-based (batch mode) DSP programs have three distinct, sequential phases to their operation when the  key is pressed or Go button or green light icon is clicked. First, data is accumulated into the acquisition buffer until the buffer is filled to the specified acquisition length. Second, a Fast Fourier Transform (FFT) is performed to obtain


amplitude (and sometimes phase) versus frequency data which is stored in a different memory buffer from the acquired signal (amplitude versus time). Third, a post-processed version of the amplitude versus time or amplitude versus frequency data (depending upon Sweep panel settings at Source 1 and Data 1 or 2) is transmitted from the DSP module in the test system to the computer for graphing by APWIN software.

It is often desirable to re-transform the data already present in the acquisition buffer, rather than acquiring new data. This will be the case when a previously-acquired waveform is downloaded from computer disk to the DSP module for further analysis. It will also be the case if it is desired to re-transform the existing contents of the acquisition buffer while using a different window function. Re-transformation is frequently used with the MLS Quasi-Anechoic Acoustical Analyzer when going back and forth between time domain (impulse response) and frequency domain views of an acquired signal.

The Transform Data without Acquire command on the Sweep menu, or the equivalent  keystrokes, cause the second and third phases described above to take place (FFT transformation and transmission of results to computer).


## Reprocess Data

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FFT-based (batch mode) DSP programs have three distinct, sequential phases to their operation when the  key is pressed or Go button or green light icon is clicked. First, data is accumulated into the acquisition buffer until the buffer is filled to the specified acquisition length. Second, a Fast Fourier Transform (FFT) is performed to obtain amplitude (and sometimes phase) versus frequency data which is stored in a different memory buffer from the acquired signal (amplitude versus time). Third, a post-processed version of the amplitude versus time or amplitude versus frequency data (depending upon Sweep panel settings at Source 1 and Data 1 or 2) is transmitted from the DSP module in the test system to the computer for graphing by APWIN software.


It is sometimes desirable to perform a new post-processing of the FFT results and transmission of the post-processed data to the

computer for display, without either acquiring new data or re-transforming the data already present in the acquisition buffer. This is faster than re-transforming followed by new post-processing. It is automatically done (unless turned off in the Utilities Configuration menu) when zooming in on a section of an FFT display and wishing to obtain optimum resolution of the new sub-set of the original data, or when changing multitone post-processing methods between the normal (response and level), distortion, or noise functions.

The Reprocess command on the Sweep menu, or the equivalent  keystrokes, cause the third phase described above to take place (post-processing of FFT results and transmission of results to computer).

### Re-Process FFT Data on Zoom

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When this box is checked, each graph zoom in action (by dragging the mouse) or Zoomout or Zoomout to Original action (right mouse button menu on graph) will be accompanied by a “Sweep Re-Process Data” () command. If the acquired signal is still in DSP memory, this assures that the resolution is always the maximum supported by either the number of points graphed or the FFT transform length.



# Spectrum Analyzers

The Spectrum Analyzer DSP program (FFTSLIDE.AZ1) and the Spectrum Analyzer/Generator DSP Program (FFTGEN.AZ1) are general-purpose programs. They can acquire a sampled two-channel signal and display the results in time domain (storage oscilloscope) view. They can perform a Fast Fourier Transform to produce a frequency domain (spectrum analyzer) view of the signal. FFTSLIDE is generally more flexible and powerful in its analysis capabilities. FFTGEN is somewhat less flexible (although it does provide a spectrum averaging feature not present in FFTSLIDE), but incorporates a DSP signal generator for sinewaves or arbitrary waveforms.

## FFTSLIDE

The Spectrum Analyzer program (FFTSLIDE) is a general purpose signal acquisition, waveform display, and FFT spectrum analysis program. It features flexible hardware triggering for acquisition, pre-trigger, the ability to slide the start point of an FFT transform anywhere in the record, and the longest record length of any System One FFT program.

## FFTGEN

The Spectrum Analyzer/Generator program (FFTGEN) consists of a real-time digital domain sinewave generator which alternately can generate signals from a downloaded waveform file, and a general purpose signal acquisition, waveform display, and FFT spectrum analysis capability. FFTGEN is the only System One FFT program with spectrum averaging capability which can reduce the variance of noise to make coherent signals near the noise level stand out more clearly.


Program	Acq.buffer length, min mem	Acq. time at 48 kHz, min mem	Acq.buffer length, full mem	Acq. time at 48 kHz, full mem
<b>FFTSLIDE</b>	8192	171 ms	30720	640 ms
<b>FFTGEN</b>	8192	171 ms	16384	340 ms

In time domain (oscilloscope mode) analysis, any portion up to and including the full acquisition buffer length may be displayed. For FFT spectrum analysis, the portion of the acquisition buffer to be transformed by FFTSLIDE and FFTGEN is limited to several discrete sample length selections which are binary powers—256, 512, 1024, 2048, 4096, and if maximum DSP memory is present, 16,384 samples long. With FFTGEN, the selected portion of the record that will be transformed starts at the beginning of the record. FFTSLIDE has the additional flexibility of positioning the start point of the section to be transformed anywhere in the acquisition buffer.

## Time Domain

When time domain (oscilloscope) view is selected (FFT Time at Source 1), the amplitude values of the samples in the Acquisition Buffers are sent via the Processing block to the computer for display. The Start to Stop time span at Source 1 on the Sweep panel determines which section of the Acquisition Buffer is processed and sent to the PC. The processing choices include a “no processing” mode (Display Samples) and three processing modes, described in detail later in this chapter. The waveform is displayed by selecting Ch 1 Ampl and/or Ch 2 Ampl at any of the Data 1 through Data 6 browsers on the Sweep panel. Waveforms are typically displayed using a non-decibel unit (Volts or FFS or %FS) rather than a decibel unit, a linear vertical scale, and with the graph top and bottom values symmetrically located above and below zero (ground).

### Display Changes

Changes may be made in the selected units or graph top and bottom values and displayed most rapidly by the  key which simply re-graphs the present contents of the computer Data Memory without

involvement of the DSP unit at all. Changes in the time span viewed (often by zooming with the mouse), displayed resolution (by changing the number of Steps), and processing mode are made most rapidly without acquiring signal again by use of the **CTRL** **F6** keystroke. This causes a re-processing and transmission of the processed data to the computer for display, but no new signal acquisition or FFT transform.

## Frequency Domain

When frequency domain (spectrum analysis) view is selected (FFT Freq at Source 1), a portion of the Acquisition Buffer will be Fourier transformed. The portion transformed begins with the first sample of the Acquisition Buffer FFTGEN. With FFTSLIDE, the FFT Start Time field defines the location of the beginning of the transform. The length of the Acquisition Buffer to be transformed is set by the Transform Length field. The record to be transformed may be multiplied by a window function or (typically with synchronous signals) no window may be used. The FFT process results in magnitude (amplitude versus frequency) results.

The Amplitude vs. Frequency information may be processed by one of several techniques, depending upon the user's needs. These post-processing techniques are described in several sections later in this chapter. The results are displayed by selecting Ch 1 Ampl or Ch 2 Ampl at any of the Data 1 through Data 6 browsers on the Sweep panel. Spectrum analysis is typically displayed either in a non-decibel units (Volts, FFS, or %FS) with a logarithmic vertical scale, or with a decibel unit.

### Display Changes

Changes may be made in the displayed units or graph top and bottom values and re-graphed most rapidly by the **F7** key. This simply re-graphs the present contents of the PC Data Memory without involving the DSP. Changes may be made in the window function, transform length, and transform starting point (with FFTSLIDE) and displayed most rapidly by the **F6** keystroke rather than acquiring signal again. The **F6** key causes a new FFT transform and sends data to the computer but no new acquisition. Changes may be made in the

post-processing selections (for example, going from response to distortion to noise with a multitone signal), frequency span displayed by Source 1 Start and Stop, graph top and bottom values, displayed resolution via number of Steps, etc., most rapidly without either acquiring signal or transforming signal again via the **CTRL** **F6** keys. The **CTRL** **F6** keys re-process the existing FFT results according to the present settings on the Digital Analyzer and Sweep panels without either acquiring signal or performing a new FFT. Zooming in or out on the graph automatically sends a **CTRL** **F6** command to the DSP unit unless this feature has been disabled by the Utilities Configuration command (Reprocess FFT Data on Zoom checkbox).

## Analog/Digital Input Selection

All the FFT-based programs can acquire digital signals (two channels) directly from any of the digital interfaces of System One Dual Domain, and analog signals (two channels) via built-in 16-bit A/D converters. The Input control near the top of the Digital Analyzer panel permits selection of **Digital** or **A/D**. Click on the down arrow at the end of the field, then click on the desired input domain.

### Analog Input Selections

When “**A/D**” is selected in the Input field, the Ch 1 and Ch 2 Source selections are Anlr A, Anlr B, Anlr Reading, Gen Mon, DSP A, DSP B, and None.

“**Anlr A**” and “**Anlr B**”, respectively, connect the Ch 1 and Ch 2 A/D converter inputs across the System One Analog Analyzer input channels at points following the input ranging circuitry. These are essentially the same points monitored by the front panel Channel A and Channel B Monitor

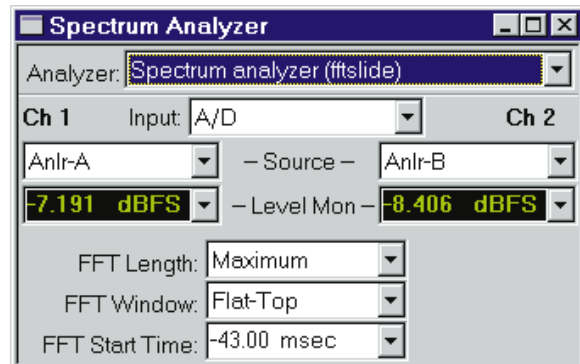


Figure 11-1 FFTSLIDE Panel, Small Version



Output BNC connectors. They are thus nearly-constant amplitude versions of the input signals following any required gain or attenuation and balanced-to-unbalanced conversion, but before any other signal processing such as filtering.

The “**Anlr Reading**” selection connects the Ch 1 and/or Ch 2 A/D converter input to the final analog signal in the System One Analog Analyzer, following all signal processing including the bandpass/bandreject filter, built-in low pass and high pass filters, optional filters or the external filter, and also following all signal processing of the IMD analyzer option or the Wow and Flutter option. This signal thus consists of harmonics and noise in THD+N mode, intermodulation distortion products in each IMD mode, the wow & flutter signal from the discriminator in W&F mode, and is band-limited by the filters selected in any mode.

The “**Gen Mon**” selection connects the Ch 1 and/or Ch 2 A/D converter input to a constant-amplitude monitor output from the Analog Generator, essentially the same signal as the Monitor Output BNC connector on the Generator Auxiliary Signals panel on the rear of System One.

The “**DSP A**” and “**DSP B**” selections, respectively, connect the Ch 1 and Ch 2 A/D converter inputs directly (DC coupled) to BNC input connectors. On System One Dual Domain versions which include optical digital input and output connectors on the front

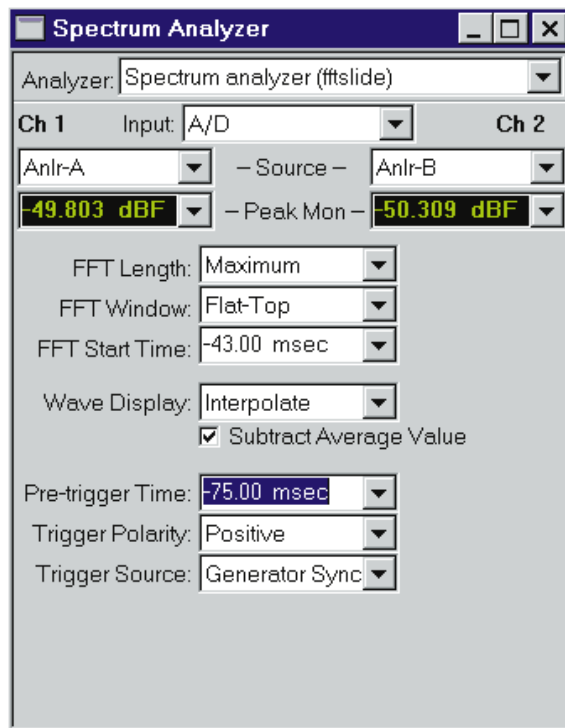


Figure 11-2 FFTSLIDE Panel, Large Version

panel, these connectors are located on the rear panel of System One and are labeled DSP A and B. On versions that do not have optical inputs and outputs, these BNC connectors are labeled Analog Inputs A and B and are at the left end of the lower-left front panel.

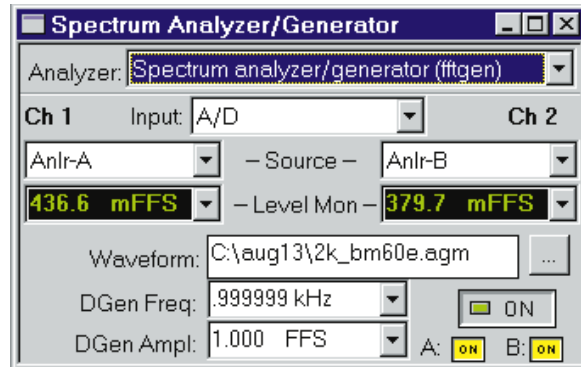


Figure 11-3 FFTGEN Panel, Small Version

The “None” selection disables acquisition into the acquisition buffer, exactly as described above for Digital Input, to enable the two acquisition buffers to be filled at different times.

## Digital Input Selections

When “Digital” is selected in the Input field, the Ch 1 and Ch 2 Source selections are A, B, or None.

“A” selects channel (subframe) A and “B” selects channel (subframe) B of two-channel digital input signals.

The “None” selection causes there to be no acquisition into the corresponding FFT acquisition buffer. This is useful when acquiring two signals that are not available at the same time. For example, signal may be acquired into the Ch 1 buffer from channel A of a device under one set of conditions by selecting A at Ch 1 and None at Ch 2. Device conditions may then be changed, and a second acquisition made with None selected at Ch 1 and A selected at Ch 2. A two-channel waveform display or FFT spectrum analysis will then show the “before” conditions on Ch 1 and the “after” conditions on Ch 2.

## Real-Time Digital Domain Level Monitors

The Level Mon readings fields (green digits on a black background) on the FFT panels continually display the digital domain amplitude at the output of the Ch 1 and Ch 2 A/D converters. Only digital domain units (FFS, dBFS, %FS, or bits) available for these meters. Clicking the down arrow at the end of the display field, then clicking the desired unit will change units.

The purpose of these Level Monitors is to avoid overload of the A/D converters. When signal is being acquired from the Anlr A, Anlr B, Anlr Reading, or Gen Mon sources and Automatic Ranging is in use in the System One Analog Analyzer, converter overrange should never be a problem. If any of the Analog Analyzer Range controls is fixed, or if the DSP A or DSP B input selection is used, it is the user's responsibility to see that the maximum signal amplitude never exceeds digital full scale. Note, however, that these meters are peak-reading and that digital full scale is defined as the RMS amplitude of a sinewave whose peaks just touch the maximum positive and negative digital codes. Therefore, is a signal whose crest factor is lower than a sinewave is adjusted so that its peaks touch maximum and minimum codes, these meters will read more than full scale. A full-scale digital domain squarewave, for example, will indicate 1.414FFS (+3.01 dBFS).

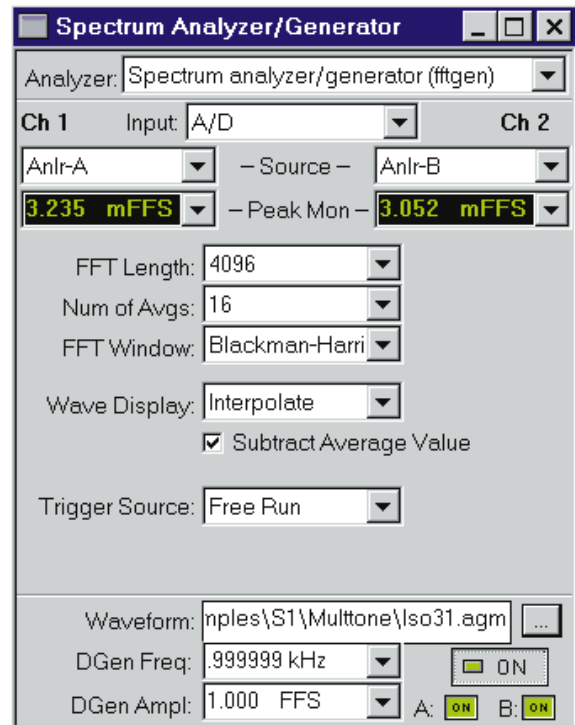



Figure 11-4 FFTGEN Panel, Large Version

## Triggering and Acquisition

FFTGEN and FFTSLIDE differ considerably in their capabilities for triggering the acquisition of signal into their acquisition buffers. FFTSLIDE is more flexible, with the ability to trigger on positive-going or negative-going signal transitions at a number of different points in the analog analyzer and generator plus the AC mains (power line) and several software triggers. FFTGEN offers only the software triggers.

### Trigger Source

Acquisition of signal into any FFT program acquisition buffer may start immediately when  is pressed or Go is clicked, or may wait for a trigger event, depending upon the setting of the Trigger Source field. The Trigger Source field is visible only on the large version of the Digital Analyzer panel. To change selections, click on the down arrow at the end of the Trigger Source box, then click on the desired selection. For FFTSLIDE, the available selections are Ch A Input, Ch B Input, DSP A, DSP B, Reading Signal, Generator Sync, AC Mains, Channel #1, Channel #2, Auto, and External. Seven of these selections are hardware trigger sources and three are software sources. FFTGEN has only the software choices of Channel #1, Channel #2, and Auto.

#### Hardware Sources (FFTSLIDE only):

**Ch A Input and Ch B Input:** these selections are the analog analyzer A and B input channels, following input ranging.

**DSP A and DSP B:** these selections are the two input BNC connectors, located on the front panel of SYS-222 and early SYS-322 units and on the rear of SYS-322 units which have optical connectors on the front panel.

**Reading Signal:** the signal at the output of the analog analyzer reading meter, following all filtering

**Generator Sync:** this is the same signal as at the Sync Output BNC on the Generator Auxiliary Signals panel, on the rear of all DSP units. This signal is a squarewave at the analog generator frequency in sinewave and squarewave waveforms, the envelope of the burst signal


in all Burst waveforms, a squarewave at the lower IMD frequency in SMPTE IMD waveform, a squarewave at 1/2 the frequency spacing in CCIF IMD waveform, the squarewave IMD signal in DIM IMD waveform, and a pulse at the pseudo-random repetition rate in Pseudo noise modes. There is no signal at this point in Random noise modes.

**AC Mains:** the power line frequency.

**External:** this source is operational only with SYS-322 (Dual Domain) units. It is the signal connected to pin 3 of the 15-pin D-sub connector on the rear of the DSP module. If pin 3 is high (or open circuit, in which case it is pulled high by an internal pull-up resistor), triggering occurs at the next digital sample. Pulling pin 3 low from an external device holds off triggering, with acquisition being triggered on the next sample after pin 3 is pulled high. The External selection is thus unaffected by the Trigger Polarity setting.

## Software Sources

The software triggering source selections are Off, Channel #1, Channel #2, Auto, and Digital Generator. If hardware triggering is required, use FFTSLIDE.

When “**Off**” is selected, signal acquisition begins immediately after  or Go is initiated, regardless of signal amplitude. This is the typical operating mode with steady-state test signals.

The “**Channel #1**” and “**Channel #2**” selections have a fixed trigger threshold of 0.1%FS (-60 dBFS) on the channel referred to.


The “**Auto**” selection will cause triggering if either Channel 1 or Channel 2 has a signal amplitude greater than digital “infinity zero”.

The “**Digital Generator**” selection functions only on Dual Domain units (SYS-300 series). If FFTGEN is generating a sinewave, a Digital Generator trigger occurs at the beginning (positive-going zero crossing) of each cycle of the sinewave. If FFTGEN is generating a signal from a waveform file, a Digital Generator trigger occurs as the first sample is read from the waveform file.

## Pre-Trigger Time (FFTSLIDE only)

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FFTSLIDE has the ability to acquire a signal with the first samples starting at a user-defined time before the trigger occurs and then continuing until the buffer is full. This permits analysis of signal conditions both before and after the triggering event. A negative time value entered in the Pre-Trigger Time field determines how much time (and thus how many samples) prior to the trigger event are retained. The Pre-Trigger Time field is visible only on the large form of the Digital Analyzer panel. The total length of signal acquired, including pre-trigger portion, will be as described in the FFT Transform Length section (see page 11-15). For example, with maximum memory the length of the FFTSLIDE acquisition buffer for each channel is 640 milliseconds. If the Pre-Trigger Time value is -50 milliseconds, for example, then 590 additional milliseconds of signal following the trigger will also be acquired to fill the entire 640 ms buffer.

Pre-trigger data is acquired in this fashion: when the  key is pressed or Go is clicked, FFTSLIDE and the DSP module immediately begin acquiring data samples, even though no trigger event may have yet occurred. If the acquisition buffer should completely fill before a trigger event occurs, data continues to be acquired in a FIFO (first in first out) basis with the oldest data being replaced as new data is added. When the trigger event occurs, FFTSLIDE creates a marker at that location (time zero) and another marker at the pre-trigger time before time zero, then continues acquiring until every location up to the pre-trigger marker is filled. Any portion from the pre-trigger time through time zero to the end of the record may then be displayed in oscilloscope fashion or transformed and viewed as a spectrum analysis.

## Trigger Polarity (FFTSLIDE only)

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The flexible hardware triggering capability of FFTSLIDE permits triggering upon either a positive-going or negative-going signal transition. The Trigger Polarity field (visible only on the large version of the Digital Analyzer panel) permits selection of the desired polarity. The selections are Off, Positive, and Negative.

When Off is selected, FFTSLIDE is “free running” and acquires immediately after the  key is pressed, without waiting for any trigger event.

When Positive is selected, FFTSLIDE’s “time zero” will be the first positive-going zero crossing of the trigger signal selected in the Trigger Source field.

When Negative is selected, FFTSLIDE’s “time zero” will be the first negative-going zero crossing of the selected trigger signal.

## Waveform Display Mode Selection

Four modes are available in FFTSLIDE and FFTGEN for processing the amplitude-versus-time relationship of a sampled signal before sending it to the computer for display of the waveform. These modes are applicable only to “digital storage oscilloscope” operation (amplitude versus time graphs) and have no effect on FFT spectrum analysis.

The four modes available in the “Wfm Display” (Waveform Display) field are Interpolate, Display Samples, Peak Values, and Absolute Values. To change Waveform Display modes, click on the down arrow at the end of the Wfm Display field and click on the desired selection. The Wfm Display field is visible only on the large version of the Digital Analyzer panel.

When “Interpolate” is selected, the DSP module will perform an interpolation calculation based on the assumption that the signal was band-limited by a low-pass filter before sampling. The Interpolate selection produces a much more accurate display of the signal waveform when the signal frequency is high (such as sample rate/100 or higher).

When “Display Samples” is selected, no processing takes place in the DSP module. At each time value plotted on the X-axis, the DSP simply sends the amplitude of the nearest-in-time acquired sample to the computer for plotting. When the signal frequency is low compared to the sample rate, this may produce an acceptable representation of the original signal waveform. At high signal frequencies, the waveform

may be entirely unrecognizable in the Display Samples mode. For example, a 16 kHz sinewave acquired at the 48 kHz sample rate will have each cycle of waveform represented by only three amplitude samples and the result will look very little like a sinewave. The Display Samples mode may be useful when examining the true, quantization-limited waveforms of very low amplitude digital domain signals.

When “Peak Values” is selected, the DSP searches all sample amplitudes in the acquisition buffer between each pair of X-axis time values plotted and sends to the computer for plotting the largest positive or negative value in that span, preserving the plus or minus sign. The intended use of the Peak Values mode is when graphing a relatively long time span on the X-axis, where the combination of Start-to-Stop time span and Steps value on the Sweep panel results in skipping across many actual acquired samples between plotted points. For example, assume a signal is acquired at the 48 kHz sample rate (20.8 microseconds between samples). If the waveform of that signal is being viewed from 0 to 200 milliseconds with 400 steps, the time span between plotted points on the graph X-axis is 0.5 milliseconds (500 microseconds). There are approximately 24 samples between plotted points. If Peak Values or Absolute Values modes are not used, an unfortunate combination of signal frequency, X-axis span, and Points value can make it appear that no waveform, a near-DC signal, or a waveform at a completely different frequency is present (see Graphic Aliasing). Since Peak Values searches through all sample values within each span between plotted points and sends the largest value to be plotted, signals cannot be missed.

When “Absolute Values” mode is selected, the DSP searches all sample amplitudes in each plotted-point-to-plotted-point span as it does in Peak Values mode, but takes the absolute value of the largest positive or negative value and thus always sends a positive number to the computer. The advantage of Absolute Values mode is that logarithms may be computed when all numbers involved are positive, so a dB units may be used on the Y axis to display the waveform. Waveform display with Absolute Values mode thus can create a wide dynamic range oscilloscope which displays the envelope of an audio signal, calibrated in familiar dB units such as dBV, dBm, dBu, etc.



Absolute Values mode is most effective when the X-axis span and Points values are selected to produce approximately two plotted points per cycle of the waveform being plotted. For example, if an envelope display of tone burst waveforms of a 1 kHz signal (1 millisecond period) are being plotted across a 50 millisecond span, the Points value on the Sweep panel should be set to approximately 100.

## Frequency Domain (Spectrum Analysis)

The process of spectrum analysis of signals with System One DSP units consists broadly of three steps: acquiring the signal into the acquisition buffer, computing the FFT and putting the results into an amplitude-versus-frequency buffer, and displaying the amplitude-versus-frequency results on the computer screen. Depending on the application, there may also be some pre-processing of the acquisition buffer contents before the FFT is computed and some post-processing of the spectrum analysis results before they are graphed. FFTSLIDE and FFTGEN pre-processing can include a quasi-AC-coupling computation (which also affects time domain displays). Both programs can multiply the time domain signal by a window function. Post processing can include “peak picking” when there are more FFT bins than plotted points on the graph, plus specialized multitone signal post-processing for fast measurements of response, distortion, noise, phase, and computation of psychoacoustical masking curves.

## Subtract Waveform Average


FFTSLIDE and FFTGEN include a computational feature which has many of the benefits of AC coupling. When the “Subtract Average Value” checkbox is checked, the DSP will compute the average value of all samples in the acquisition buffer and then subtracts that computed value from the value of each sample before an FFT transform or processing the values according to the “Wfm Display” field and sending the results to the computer for display. The effect of the “Subtract Average Value” function is thus very similar having used AC coupling before acquiring the signal, as long as no signal peaks exceeded digital full scale. Use of the “Subtract Average Value” function may be valuable when examining low-level signals which

contain a significant amount of DC offset, particularly in time domain (oscilloscope) presentations where the DC offset might otherwise cause the signal to be off-screen at the selected vertical span.

The Subtract Average Value checkbox is visible only on the large version of the Digital Analyzer panel.

## Window Functions


Windowing concepts are discussed in the preceding chapter. The available window selections for FFTSLIDE and FFTGEN are Blackman-Harris, Hann, Flat-top, and None. Each available window function has a different set of trade-offs in effective close-in selectivity (signal spillover into nearby FFT bins) and in amplitude measurement error if the signal does not fall at an exact bin center frequency. See Figure 10-2. The Hann window has the best selectivity near the top and its skirts fall off monotonically. The Blackman-Harris window has better selectivity than the Hann except for the few bins centered on the signal, and has side lobes at about -90 dB and below. The Flat-Top window has the poorest selectivity, but virtually no amplitude error across the center three bins.

To select a window, click on the down arrow at the end of the FFT Window box and click on the desired windowing function. Note that the window function is applied after signal acquisition and before the actual FFT transform. It is thus simple to change FFT Window selections and re-transform the data with the  key, without making another acquisition. With the Append box on the Sweep panel checked, FFTs of the same signal with several different window functions applied can be graphically compared.

## FFT Start Time (FFTSLIDE)



Unlike the other System One FFT programs which always compute the transform on the portion of the record beginning with the first sample, FFTSLIDE permits the user to select any point in the acquired signal record as the beginning of the portion to be transformed. The FFT transform is then computed for the contiguous section of samples starting at that sample and continuing for the number of samples

chosen in the FFT Length field. FFTSLIDE thus permits selective spectrum analysis of different sections of a complex signal such as program material or special test signals such as sinewave bursts.

If the original signal acquisition () was made with a negative value in the Pre-trigger Time field, negative values up to and including that same value may be used as FFT Start Time values to permit spectrum analysis of the pre-trigger section of the acquired record. A common example is when using the Analog Generator's BUR option sinewave burst signal as test stimulus to some dynamic processor such as a compressor or limiter.

## FFT Transform Length

The FFT transform length determines the frequency resolution of the FFT spectrum analysis. On FFTGEN, the FFT Length field also determines the amount of signal that will be acquired. FFTSLIDE operates differently; see the description below and the table on page 11-16. Transform length is an operator-selectable parameter on FFTSLIDE and FFTGEN.

The FFT Length field value controls the record length used as input to the FFT process when either /Go is initiated to acquire and transform, or the  function key or “Sweep Transform Data without Acquire” menu command is used to re-transform any portion of a record previously acquired. Longer transform lengths produce greater frequency resolution in the resulting FFT, but require longer times to acquire and to transform the signal.





To change the FFT Length, click the down arrow at the end of the field and select the desired value. Available selections are Maximum, 4096, 2048, 1024, 512, and 256. “Maximum” means 16384 on all System One Dual Domain units and on System One Plus DSP units with the MEM (memory) option, but is the same as the “4096” selection on System One Plus DSP units without the MEM option.

The record length (number of samples acquired into the acquisition buffer) is not the same as the FFT Length in FFTSLIDE. Available acquisition buffer size with FFTSLIDE loaded is 8192 samples per

channel in an SYS-222 series unit without the MEM option, and 30720 samples per channel in an SYS-222 series with MEM option or in any SYS-322 series. At any sample rate above 1 kHz (set by the Rate field on the Digital I/O Panel), FFTSLIDE fills the entire acquisition buffer upon trigger. At the 1 kHz sample rate, the number of samples acquired is 1.5 times the value in the FFT Length field. Acquisition times thus become quite long with FFTSLIDE at low sample rates, as shown:

FFTSLIDE, Acquisition time vs. sample rate and memory

Rate	Time (sec.), small memory	Time (sec.), full memory
192k	0.043	0.160
48k	0.171	0.640
32k	0.256	0.960
8k	1.024	3.840
1k	6.144	24.576

After the initial acquisition and transform (/Go), re-transformation may be made of another portion of the acquired signal by specifying (if desired) a new value for FFT Start Time, changing (if desired) the FFT Length, and pressing the  key or using the Sweep Transform Data without Acquire command. Transformation begins with the sample at the time value in the FFT Start Time field and extends through the number of samples selected in the FFT Length field. The full acquired signal remains in memory until the next acquisition () , a change in DSP program, or turning off power to System One. If a negative value was entered in the Pre-Trigger Time field before  was pressed, the portion selected by the FFT Start Time field can extend back into the pre-trigger portion of the record by entering a negative value in the FFT Start Time field.

## Post-FFT Processing

At the conclusion of the Fast Fourier Transform, magnitude-vs.-frequency data arrays exist in DSP memory for both signal channels. Generally speaking, the entire contents of those arrays are not simply sent to the computer for display. The user-set Start and Stop frequency values on the Sweep panel may call for only a portion of the data to be sent to the computer. The number of Steps

on the Sweep panel may be greater or smaller than the number of FFT bins in the Start-Stop frequency span, with either condition requiring some post-FFT processing. With FFTGEN, a spectrum averaging feature is available to improve the signal to noise ratio of the displayed spectrum.

## FFT Averaging (FFTGEN)

FFTGEN has the ability to average a number of successive acquisitions and spectrum analyses of a signal and display the averaged result. Since noise is random in amplitude and phase, averaging a succession of noise measurements results in a degree of cancellation and the averaged result will have less variance than the range of initial acquisitions. Coherent signals, however, are the same at each acquisition and thus are not affected by averaging. Thus, spectral averaging will reduce the maximum peak excursions of the noise “baseline” in a typical signal spectrum while not affecting continuous signals, making it easier to detect and measure low amplitude signals and distortion products. Averaging over many seconds or minutes of program material such as music or voice may also be useful in order to determine the long-term average amplitude versus frequency distribution.

To enable averaging, click the down arrow at the end of the “Num of Avgs” field and click on the desired value of the list that appears. The “Num of Avgs” field is visible only on the large version of the Digital Analyzer panel. The selections are 1, 4, 16, 64, 256, and 1024. “1” produces no averaging. Any selections other than “1” will produce the sequence shown in flow chart form in Figure 11-5.

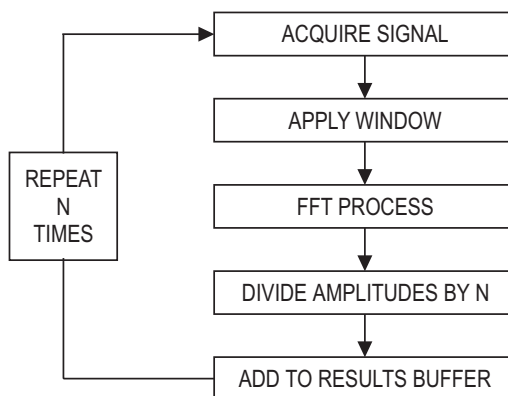


Figure 11-5 Functional Flow Chart of FFTGEN Spectrum Averaging Process

Note that use of the **F6** key (Re-transform without new acquisition) will destroy the result of the averaging process by producing a new FFT transform of the last acquisition. The **CTRL F6** key (transmit data from DSP module to computer according to present settings) and zooms made by dragging on the graph will not spoil the results of averaging.

## Transform Data without Acquire

FFT-based (batch mode) DSP programs have three distinct, sequential phases to their operation when the **F9** key is pressed or Go button or green light icon is clicked. First, data is accumulated into the acquisition buffer until the buffer is filled to the specified acquisition length. Second, a Fast Fourier Transform (FFT) is performed to obtain amplitude (and sometimes phase) versus frequency data which is stored in a different memory buffer from the acquired signal (amplitude versus time). Third, a post-processed version of the amplitude versus time or amplitude versus frequency data (depending upon Sweep panel settings at Source 1 and Data 1 or 2) is transmitted from the DSP module in the test system to the computer for graphing by APWIN software.

It is often desirable to re-transform the data already present in the acquisition buffer, rather than acquiring new data. This will be the case when a previously-acquired waveform is downloaded from computer disk to the DSP module for further analysis. It will also be the case if it is desired to re-transform the existing contents of the acquisition buffer while using a different window function.


The Transform Data without Acquire command on the Sweep menu, or the equivalent **F6** keystrokes, cause the second and third phases described above to take place (FFT transformation and transmission of results to computer).

## Reprocess Data

FFT-based (batch mode) DSP programs have three distinct, sequential phases to their operation when the **F9** key is pressed or Go button or green light icon is clicked. First, data is accumulated into the


acquisition buffer until the buffer is filled to the specified acquisition length. Second, a Fast Fourier Transform (FFT) is performed to obtain amplitude (and sometimes phase) versus frequency data which is stored in a different memory buffer from the acquired signal (amplitude versus time). Third, a post-processed version of the amplitude versus time or amplitude versus frequency data (depending upon Sweep panel settings at Source 1 and Data 1 or 2) is transmitted from the DSP module in the test system to the computer for graphing by APWIN software.

It is sometimes desirable to perform a new post-processing of the FFT results and transmission of the post-processed data to the computer for display, without either acquiring new data or re-transforming the data already present in the acquisition buffer. This is faster than re-transforming followed by new post-processing. It is automatically done (unless turned off in the Utilities Configuration menu) when zooming in on a section of an FFT display and wishing to obtain optimum resolution of the new sub-set of the original data, or when changing multitone post-processing methods between the normal (response and level), distortion, or noise functions.

The Reprocess command on the Sweep menu, or the equivalent  keystrokes, cause the third phase described above to take place (post-processing of FFT results and transmission of results to computer).

### Re-Process FFT Data on Zoom


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When this box is checked, each graph zoom in action (by dragging the mouse) or Zoomout or Zoomout to Original action (right mouse button menu on graph) will be accompanied by a “Sweep Re-Process Data” () command. If the acquired signal is still in DSP memory, this assures that the resolution is always the maximum supported by either the number of points graphed or the FFT transform length.

## Saving and Loading Waveforms

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Both channels of acquired waveform remain in the DSP Acquisition buffer until one of the following events:

-  (Go, Sweep Start, etc) is pressed to acquire new waveforms
- a different DSP program (or no DSP program) is loaded, by changing the “Analyzer” field contents on the Digital Analyzer panel
- power is turned off to System One
- different waveforms are downloaded from disk as described in this section.

While the waveform is still present in the Acquisition buffer, full flexibility exists to change back and forth between time and frequency domain analysis (FFT Time vs. FFT Frequency at Source 1), to view different time portions of the record, to perform FFTs of different transform length (and with FFTSLIDE to position the transform starting point at different places in the record), to change window functions, to change FFT post processing modes, to change time domain processing modes, etc. The results of any and all of these different modes of analysis may be saved, one at a time, as .AT1 test files to disk. However, once the waveform is gone from DSP memory, no further ability exists to analyze the signal other than re-loading .AT1 test files to look at the specific analyses which were saved. If full flexibility for continued analysis is desired, the acquired waveform itself may be saved to disk, rather than merely saving .AT1 files each of which represents one particular processed result of one type of analysis. The saved waveform can then later be downloaded to the DSP Acquisition buffers for further analysis. Single-channel waveforms saved at different times can later be individually loaded into the two channels of Acquisition buffer for comparison in the time or frequency domains.

## **Saving Waveforms**

---

Selected sections or the entire contents of the Acquisition buffers may be saved to a disk file by the File Save As menu command, selecting either the “Stereo Waveforms (\*.AAS)” or “Mono Waveform (\*.AAM)” as the Save As file type.

A Stereo Waveform file (.AAS) consists of two waveforms joined together into a single file. A Mono Waveform file (.AAM) contains a



single waveform. To simultaneously save the waveforms from both Acquisition buffer channels, select the Stereo choice. To save only one of the channels, select Mono.

When the waveform file type is selected, a file name is typed in by the user, and the Save button is clicked, a dialog box as illustrated in Figure 11-6 will appear. The left half of the dialog box defines what will be saved into the first section of a waveform file and the right half defines what will go into the second section.

### Full Acquisition Buffer vs. Transform Sub-Section

The choices in each section are None, Acquired-Ch 1, Acquired-Ch 2, Transform-Ch 1, and Transform-Ch 2. When saving a Mono file, the “None” selection will be volunteered for the right half. The “Acquired” choices will result in saving the entire contents of an Acquisition buffer to disk. The “Transform” choices result in saving only the sub-section of the Acquisition buffer specified by the FFT Length selection. If the Transform Length is set to Maximum at the time the waveform is saved, there will be no difference between the “Acquired” and the “Transform” selections except with FFTSLIDE in a full memory unit, where the Acquisition buffer length is 30,720 samples and the Maximum Transform Length is 16,384 samples. If the Transform Length is set to a shorter value, only the portion of the Acquisition buffer corresponding to that value will be saved. When the “Transform” selection is chosen and the currently-selected Transform Length is less than the full buffer (for example, a 1024 sample transform out of a 16,384 sample acquisition buffer), the starting point of the section saved to disk is the first sample in the buffer except with FFTSLIDE. With FFTSLIDE, the starting point of the section saved when the

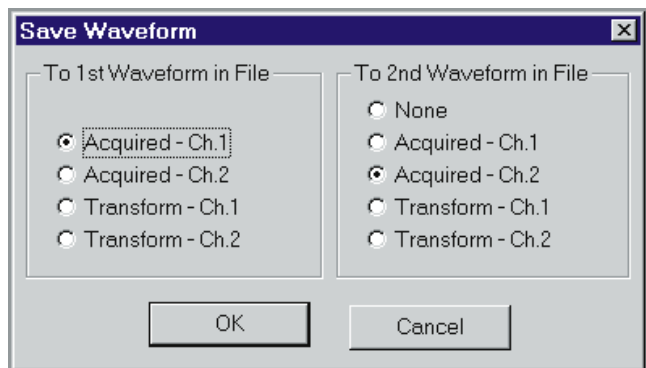


Figure 11-6 Save Waveform Dialog Box

“Transform” choice is used is the sample corresponding to the FFT Start Time setting. If a smaller section of the waveform is sufficient for the intended future analysis purposes, use of the “Transform” selections can result in much smaller disk files than the “Acquired” selections. DSP memory width is 24 bits and computer disk files are organized around 8-bit bytes, so the length of a disk file is approximately three times the length of buffer it was saved from. One channel from a 16,384 sample acquisition buffer will thus result in approximately a 48k file size, while use of the “Transform” selection with a selected Transform length of 1024 samples would result in about a 3k file size.

### One vs. Two Waveforms

The selections in the left half of the dialog box determine which buffer contents go into a Mono waveform file or into the first section of a Stereo waveform file. The contents of the second section of Stereo waveform files are defined by the settings of the right half of the dialog box. If a Ch 1 selection is made in the left side and a Ch 2 selection in the right side, a Stereo waveform file results. It is normally preferable for the two sections of a Stereo waveform file to contain records of the same length, so normally one would not use the “Acquired” selection for one section and the “Transform” selection for the other.

### Combining Mono to Stereo

It is possible to save single channel waveforms acquired at two different times and then combine them into a Stereo waveform so that simultaneous time or frequency domain analysis may be made for some “before and after” conditions. Save a Mono waveform from the first set of conditions, specifying the channel number where the desired data is presently located. Under different test conditions or with a different device under test, save another Mono waveform with a different name. Use the Open Waveform feature described below to load one of these waveform files into the Ch 1 Acquisition buffer and the other into the Ch 2 Acquisition buffer. Then, use the Save As command, select Stereo waveform and supply a new file name, and select the Ch 1 waveform into the first section and Ch 2 into the second.

It is also possible to use the DOS Copy command to concatenate two Mono waveform files into a new Stereo file. The command (from the DOS prompt) is:

```
COPY /B FILENAME1.AAM + FILENAME2.AAM NEWFILE.AAS
```

## Loading Waveforms

Existing waveform files may be downloaded from computer disk to the DSP via the File Open command. In the File Open dialog box, select the Stereo Waveforms (.AAS) or Mono Waveforms (.AAM) file type and select the desired directory and file name. Clicking the Open button then displays the Open Waveform dialog box as illustrated in Figure 11-7. The left half selections determine which DSP buffer the first section of the waveform file will be loaded into. The right half of the dialog defines which buffer the second section of the waveform file loads into. The “Acquired” and “Transform” selections work similarly to the Save Waveform operations described above. Clicking OK then actually downloads the waveform file contents. After the waveform is loaded, the **F6** and **CTRL F6** keys and settings on the Digital Analyzer panel and Sweep panel may be used as described earlier for continued analysis of the waveform.

If the downloaded file length is greater than the length of the specified buffer, an error warning will be displayed. If the downloaded file length is greater than the buffer size, the portion which fits in the buffer can be analyzed normally and the remainder cannot be viewed. Since FFTSLIDE has the longest buffer, it can completely view files saved from any of the other FFT programs but they will not be able to accept a full waveform file saved from FFTSLIDE. If the downloaded file

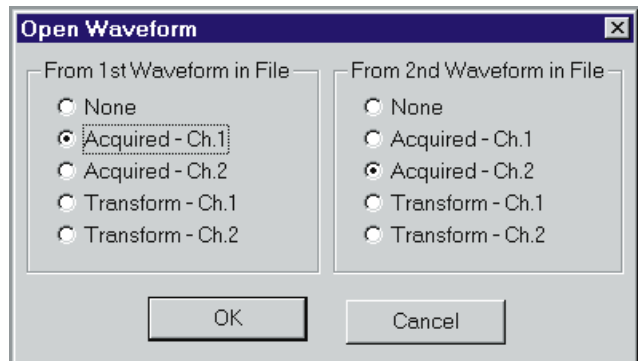


Figure 11-7 Open Waveform Dialog Box

length is shorter than the buffer size, erroneous analysis will result if the selected Transform Length extends beyond the last data sample in the buffer. The contents may first be displayed in Time Domain mode to determine exactly where the signal ends.

## FFTGEN Signal Generation

FFTGEN can generate digital domain waveforms in addition to acquiring and displaying waveforms and performing FFT spectrum analysis. FFTSLIDE has no signal generation capability. FFTGEN has two modes of signal generation. Sinewaves of any frequency can be generated on a real-time basis via internal algorithm (identically to GENANLR) when “Sine” is selected in the field near the Output button of the large form of the FFTGEN panel. If “Arb” (arbitrary) is selected, a waveform defined by a downloaded file can be re-created point-by-point in arbitrary waveform generation fashion.

### Digital Generator Waveform (FFTGEN)

To load a waveform file into the FFTGEN generator buffer, select “Arb” rather than “Sine”. Click on the Ellipsis button at the right end of the Waveform box which appears. A dialog will appear which permits navigating through directories (folders) and selecting the file name of the desired waveform file. Clicking on Open will open that file and download it to the signal generation buffer. The signal defined in the waveform file will immediately start being generated as the program steps through the samples in the waveform file at the sample rate selected in the Rate field on the Digital I/O panel. The signal will be generated continuously with the DSP stepping from the last sample to the first sample. This in effect creates a continuous loop, “splicing” the waveforms together.

Many waveform files, including distortion and wow and flutter calibration signals and multi-tone signals, are furnished with APWIN software. Users may also easily create custom waveform files from an ASCII table of frequency and amplitude (and phase, if desired) values. See Creating Waveform Files with MAKEWAV2 section of the Multitone Analysis chapter for more information.

## Mono Waveforms Only (FFTGEN)

FFTGEN supports only single waveform (monaural) generator waveform files. Monaural waveform generator files have the extension .AGM and stereo generator waveform files have the extension .AGS. FFTGEN has only a single generator buffer and the waveform in that buffer drives both digital output channels (subframes). If a stereo waveform file is opened, the second waveform in that file is loaded into FFTGEN's generator buffer.

## Digital Generator Frequency (FFTGEN)

The frequency of the sinewave is controlled by the value in the DGen Freq field.

## Digital vs. Analog Output

The generated signal may be routed to any of System One Dual Domain's digital interface output connectors, as controlled from the Output Format and Serial Type fields on the Digital I/O panel. By choosing the Output Format "D/A" selection on the Digital I/O panel, the generator channel B output signal may instead be converted to the analog domain by a built-in D/A converter. The converter output feeds the D/A Output BNC connector and may be routed through the Analog Generator output stage by the DGen waveform selection in the Analog Generator Waveform field.

## Digital Generator Amplitude

The amplitude of the digital domain signal is controlled by the DGen Ampl field. This is true whether the signal is an algorithmically-generated sinewave or from a downloaded generator waveform file. The available units are the standard digital domain units of FFS, dBFS, %FS, and bits.

When an analog output signal is converted by the D/A selection in the Output Format field of the Digital I/O panel, the input amplitude to the D/A converter and thus the analog output amplitude is controlled by the DGen Ampl field. If the analog signal is routed through the Analog Generator output stage by the DGen selection in the Waveform field of the Analog Generator panel, the DGen Ampl field value must

be set to full scale if the Analog Generator amplitude calibration is to be correct.

### **On/Off and Channel Select**

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The OFF/ON button near the bottom right corner of the Digital Analyzer panel with FFTGEN in use controls the digital output signal flow from the digital generator. No signal output is present when the button reads OFF and the “LED” is not “lighted”. Clicking on the button turns on the generator output and the button label changes to ON with the “LED” green. Note that when analog signals are supplied from this program via the “D/A” Output Format selection on the Digital I/O panel, this control must be ON in order to obtain analog signals.

The two ON/OFF buttons labeled A and B, under the main generator ON/OFF button, provide individual control of each channel (subframe) when digital domain audio is generated. Each button permits output signal flow to its respective channel when the button is yellow and indicates ON. When the button is gray and indicates OFF, no signal is supplied to that channel. When analog signals are supplied via the “D/A” Output Format selection on the Digital I/O panel, the status of these A and B buttons is not important. Signal feeds the D/A converter even if both A and B buttons are set to OFF.

# Multitone Analyzers

System One DSP programs (Digital Analyzer panel) include three programs specifically designed for generation and analysis of multitone signals. They are the Multitone Generator/Analyzer program (FASTTEST.AZ1), the Triggered Multitone Tester (FASTTRIG.AZ1), and the Codec Tester (CODEC.AZ1). All have many features in common and are described in this chapter.

FASTTEST, FASTTRIG, and CODEC operate only at the 48 kHz, 44.1 kHz, and 32 kHz sample rates and thus offer maximum bandwidth to 24 kHz. Each program includes a DSP-implemented signal generator which may be used to provide digital domain stimulus of devices while simultaneously measuring their output in the digital or analog domain (D/A converters, for example). When analog domain signals are required, the DSP-generated signal is converted to the analog domain through System One's built-in D/A converter and may be fed through the power amplifier, transformer, and amplitude control circuitry of the analog generator.

## FASTTEST

The Multitone Generator/Analyzer program (FASTTEST) generates signals, usually multitone, from a downloaded file. It acquires signals and performs FFT spectrum analysis optimized by post-processing features for use with synchronous (window-less) FFTs. FASTTEST can measure frequency response, phase response, distortion, and noise in the presence of a test signal in a few seconds.

## FASTTRIG

The Triggered Multitone Tester program (FASTTRIG) duplicates nearly all the multitone generation and analysis features of FASTTEST. To this, FASTTRIG adds two key features for measurements of multitone signals originating at distant locations, or previously

recorded. A signal recognition and triggering feature acquires signal into the analyzer only when it is a sufficiently close match to the waveform presently in the generator buffer. A frequency error correction feature uses the generator buffer waveform as a reference to correct for frequency errors introduced when the multitone signal is recorded and reproduced or is furnished by a different generator operating from a crystal clock at a slightly different frequency, a signal acquisition.

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**CODEC**

The Codec tester (CODEC) is designed for testing low bit rate coders and decoders which employ perceptual coding concepts to mask the higher levels of quantization noise and distortion caused by encoding the signal with much less information bandwidth than required for linear digital audio recording or transmission. CODEC is based on the FASTTRIG program with the addition of internal masking curve generation in the DSP program. CODEC is intended to be used with multitone signals that simulate complex program material. Following acquisition of the output of the companion decoder, CODEC generates a composite masking curve for the multitone signal received (based on some of the best current psychoacoustical research) which is used as a reference limit of audibility for artifacts. CODEC then integrates quantization noise and distortion across critical bandwidths and compares those integrated measurements with the masking curve. Noise and distortion below the masking curve should be inaudible. Noise and distortion above the masking curve may be audible, especially to critical listeners and in listening environments that do not further mask the signal with ambient noise.

FASTTEST, FASTTRIG, and CODEC with maximum DSP memory have acquisition buffers 16,384 samples long for each channel. In a System One + DSP without the MEM option, the acquisition buffer length is 8,192 samples for FASTTEST. FASTTRIG and CODEC require the maximum DSP memory versions to operate.



Program	Acq.buffer length, min mem	Acq. time at 48 kHz, min mem	Acq.buffer length, full mem	Acq. time at 48 kHz, full mem
<b>FASTTEST</b>	8192	171 ms	16384	340 ms
<b>FASTTRIG</b>	N/A	N/A	16384	340 ms
<b>CODEC</b>	N/A	N/A	16384	340 ms

## Multitone and Synchronous FFT Concepts

Multitone testing with synchronous generator and analyzer waveform buffers provides many advantages:

- Testing speeds with multitone techniques are 10 to 100 times faster than conventional swept sinewave techniques.
- Multitone testing can provide total distortion and noise measurements, including both harmonic and intermodulation distortion products in one test
- Synchronous multitone testing is the only known technique for measuring noise in the presence of a test signal, providing for the first time meaningful noise measurements on dynamic processors such as compressors, modulation processors, and noise gates
- A multitone signal is much more realistic than a single sinewave, with both spectral and dynamic characteristics (crest factor) similar to music and voice
- A brief (fractional second) inserted burst of multitone signal in a broadcast program or recording can be recognized and captured by an analyzer for regular, unattended testing of broadcast transmission facilities

- Multitone signals provide realistic stimulus to low bit rate encoders and the output can be analyzed according to psychoacoustic models including frequency masking effects

The major advantages of multitone testing depend on synchronous signals and window-less FFT analysis. As discussed later in this chapter (see page 10-9), windowing functions are required when analyzing non-synchronous signals by FFT. Application of a window function, however, causes energy to appear to spread over 5 to 10 FFT bins above and below the actual signal frequency. A synchronous signal is one with an exact integer number of signal cycles in the FFT analysis transform buffer. With a synchronous signal, each FFT bin represents a perfect rectangular selectivity curve one bin wide with no spillover to adjacent bins. Synchronous signals are created by:

- having a generator waveform buffer size equal to or an integer sub-multiple of the analyzer FFT transform length
- using the MAKEWAV2 utility to create waveform files in which every multitone signal component is forced to have an exact integer number of cycles in the generator buffer length. See the MAKEWAV2 program description on page 12-33.

## Multitone Signals

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A typical multitone test signal consists of a number of sinewaves at frequencies distributed across the audio spectrum. It is usually convenient for all the sinewaves to have the same amplitude, but amplitudes may be set individually to more closely match typical spectral energy distribution of program material, for example. Also convenient is for the sinewave frequencies to be logarithmically spaced across the spectrum, such as at 1/3-octave intervals, but any arbitrary spacing scheme may be used that does not violate the synchronous condition. The practical result is that all signal frequencies must be an integer multiple of a basic frequency which is the sample rate divided by the generator waveform length. For example, with a 48 kHz sample rate and the maximum System One DSP generator buffer length of 8192, the corresponding basic frequency is 5.859375 Hz. It is normally desirable not to locate signals at the exact 2nd or 3rd

harmonic of lower frequency signals in order to leave those bins free for measurement of harmonic distortion products.

## Multitone Analysis

The multitone signal, after passing through the device under test, is captured into the analyzer and an FFT performed. Rather than sending a complete FFT analysis to the computer for display, additional post-processing is done following the FFT to extract the most relevant audio performance information:

- Frequency response is plotted by the DSP unit sending to the computer only the amplitudes of the FFT bins containing the fundamental sinewave products of the test signal. For the example of a 31-tone 1/3 octave-spaced multitone signal from 20 Hz to 20 kHz, this means that only 31 amplitude values (per channel) are sent to the computer rather than a representation of all 8,192 FFT bins (assuming maximum transform length). The computer plots those 31 points for a conventional frequency response graph, identical to that obtained by sweeping a single sinewave in sequence through those 31 frequencies. See the discussion of the Response function of FASTTEST, FASTTRIG, and CODEC on page 12-22.
- Phase is plotted by the DSP sending to the computer the phase values at the FFT bins containing fundamental sinewave products. The computer plots those points for a phase-vs.-frequency graph identical to that obtained by sweeping a single sinewave through the same frequencies. Additional post-processing in the DSP subtracts the difference in the absolute phase values of the two channels and sends the difference to the computer to be plotted as inter-channel phase difference.
- Total distortion and noise is plotted by the DSP unit integrating the amplitudes of all FFT bins except those which contain fundamental sinewave signals. Thus, all harmonic distortion products, all intermodulation distortion products, and all noise and interfering signals are included in the integration except for

those which happen to fall into the same FFT bin as a fundamental signal. Typically, the distortion and noise integration is done across the spans between fundamental test signals. Thus, for the 31-tone 1/3 octave spaced example, the DSP would send 31 values to the computer, each of which represents total distortion and noise summed across the 1/3 octave interval between tones. When graphed by the computer, the result is a 1/3-octave resolution analysis of distortion and noise. See the description of the Distortion function of FASTTEST, FASTTRIG, and CODEC on page 12-22.

- Noise in the presence of signal depends upon the generator buffer length being exactly half the analyzer transform buffer length. The result is that the analyzer has twice the frequency resolution of the generator. For the 8,192 sample generator buffer length example above, all sinewave frequencies in the multitone signal will be exact integer multiples of the basic 5.859... Hz value. All harmonic or intermodulation distortion products generated in the device under test will also be exact integer multiples of that same basic frequency. The analyzer transform length for noise measurements would be 16,384 samples with a resulting bin width (resolution) of 2.9296875 Hz. Therefore, the analyzer has FFT bins in between each possible frequency that can contain a generated sinewave fundamental or a distortion product. These alternate bins will, however, have noise from the device under test. Therefore, a summation of the amplitudes of these alternate, “empty bins” produces a measurement of noise in the presence of test signal. Typically, the DSP unit integrates the empty bin amplitudes across the span between fundamental signals and the computer plots the resulting curve as a spectrum analysis of noise. If the DSP integrates the empty bin amplitudes across the entire audio spectrum, the result is a single number for integrated noise. Since only half the bins are included, the resulting number needs to be corrected by 3 dB to compare to conventional noise measurements across the same spectrum. See the Noise function description for FASTTEST, FASTTRIG, and CODEC on page 12-24.

- For psychoacoustically-based analysis of coders and decoders, DSP post-processing after the FFT can generate a composite frequency masking curve for any multitone signal. Comparison of a total distortion curve, integrated across frequency spans corresponding to the critical bands of human hearing, to the masking curve will show whether the increased noise and distortion due to low bit rate coding is likely to be audible. See the Masking function description of the CODEC program on page 12-24.

The phase versus frequency results may be displayed with all the multitone programs. FASTTEST permits graphing Channel 1 and Channel 2 phase independently or, instead of Channel 2 phase, can subtract the difference between Channel 1 and 2 phase and display this inter-channel phase via the Ch 2 Phase parameter.

## Analog/Digital Input Selection

All the FFT-based programs can acquire digital signals (two channels) directly from any of the digital interfaces of System One Dual Domain, and analog signals (two channels) via built-in 16-bit A/D converters. The Input control near the top of the Digital Analyzer panel permits selection of **Digital** or **A/D**. Click on the down arrow at the end of the field, then click on the desired input domain.

### Analog Input Selections

When “**A/D**” is selected in the Input field, the Ch 1 and Ch 2 Source selections are Anlr A, Anlr B, Anlr Reading, Gen Mon, DSP A, DSP B, and None.

“**Anlr A**” and “**Anlr B**”, respectively, connect the Ch 1 and Ch 2 A/D converter inputs across the System One Analog

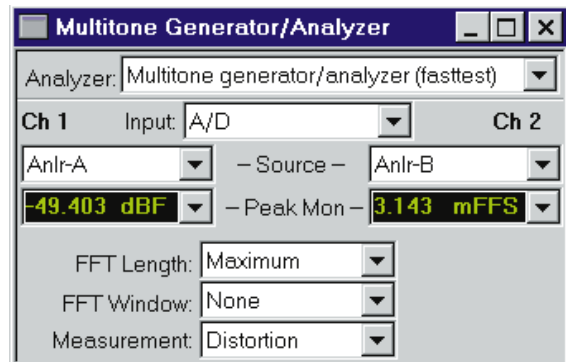


Figure 12-1 FASTTEST Panel, Small Version

Analyzer input channels at points following the input ranging circuitry. These are essentially the same points monitored by the front panel Channel A and Channel B Monitor Output BNC connectors. They are thus nearly-constant amplitude versions of the input signals following any required gain or attenuation and balanced-to-unbalanced conversion, but before any other signal processing such as filtering.

The “**Anlr Reading**” selection connects the Ch 1 and/or Ch 2 A/D converter input to the final analog signal in the System One Analog Analyzer. This mode would rarely be used in multitone signal analysis.

The “**Gen Mon**” selection connects the Ch 1 and/or Ch 2 A/D converter input to a constant-amplitude monitor output from the Analog Generator, essentially the same signal as the Monitor Output BNC connector on the Generator Auxiliary Signals panel on the rear of System One.

The “**DSP A**” and “**DSP B**” selections, respectively, connect the Ch 1 and Ch 2 A/D converter inputs directly (DC coupled) to BNC input connectors. On System One Dual Domain versions which include optical digital input and output connectors on the front panel, these connectors are located on the rear panel of System One and are labeled DSP A and B. On versions that do not have

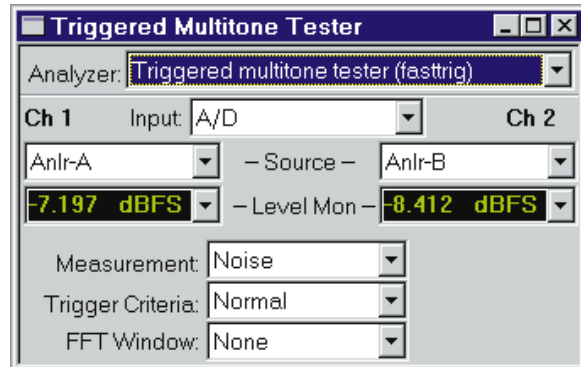


Figure 12-2 FASTTRIG Panel, Small Version

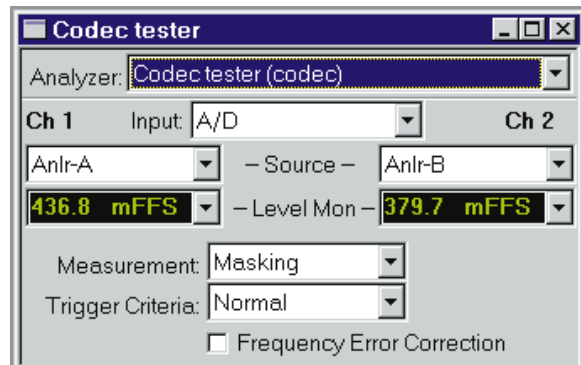


Figure 12-3 CODEC Panel, Small Version

optical inputs and outputs, these BNC connectors are labeled Analog Inputs A and B and are at the left end of the lower-left front panel.

The “**None**” selection disables acquisition into the acquisition buffer, exactly as described above for Digital Input, to enable the two acquisition buffers to be filled at different times.

## Digital Input Selections

When “Digital” is selected in the Input field, the Ch 1 and Ch 2 Source selections are A, B, or None.

“A” selects channel (subframe) A and “B” selects channel (subframe) B of two-channel digital input signals.

The “None” selection causes there to be no acquisition into the corresponding FFT acquisition buffer. This is useful when acquiring two signals that are not available at the same time. For example, signal may be acquired into the Ch 1 buffer from channel A of a device under one set of conditions by selecting A at Ch 1 and None at Ch 2. Device conditions may then be changed, and a second acquisition made with None selected at Ch 1 and A selected at Ch 2. A two-channel waveform display or FFT spectrum analysis will then show the “before” conditions on Ch 1 and the “after” conditions on Ch 2.

## Real-Time Digital Domain Level Monitors

The Level Mon readings fields (green digits on a black background) on the FFT panels continually display the digital domain amplitude at the output of the Ch 1 and Ch 2 A/D converters. Only digital domain units (FFS, dBFS, %FS, or bits) available for these meters. Clicking the down arrow at the end of the display field, then clicking the desired unit will change units.

The purpose of these Level Monitors is to avoid overload of the A/D converters. When signal is being acquired from the Anlr A, Anlr B, Anlr Reading, or Gen Mon sources and Automatic Ranging is in use in the System One Analog Analyzer, converter overrange should never be a problem. If any of the Analog Analyzer Range controls is fixed, or

if the DSP A or DSP B input selection is used, it is the user's responsibility to see that the maximum signal amplitude never exceeds digital full scale. Note, however, that these meters are peak reading and that digital full scale is defined as the RMS amplitude of a sinewave whose peaks just touch the maximum positive and negative digital codes. Therefore, is a signal whose crest factor is lower than a sinewave is adjusted so that its peaks touch maximum and minimum codes, these meters will read more than full scale. A full-scale digital domain squarewave, for example, will indicate 1.414FFS (+3.01 dBFS).

FASTTEST offers only the software trigger choices (Channel #1, Channel #2, and Auto) are available. . FASTTRIG and CODEC include sophisticated signal recognition capabilities that compare incoming signals to internally stored replicas of the expected multitone signal and trigger only when there is a good match between these signals. This feature enables FASTTRIG and CODEC to continually monitor (and reject) normal music and voice program material but to quickly capture the expected multitone signal when it occurs.

## Triggering and Acquisition

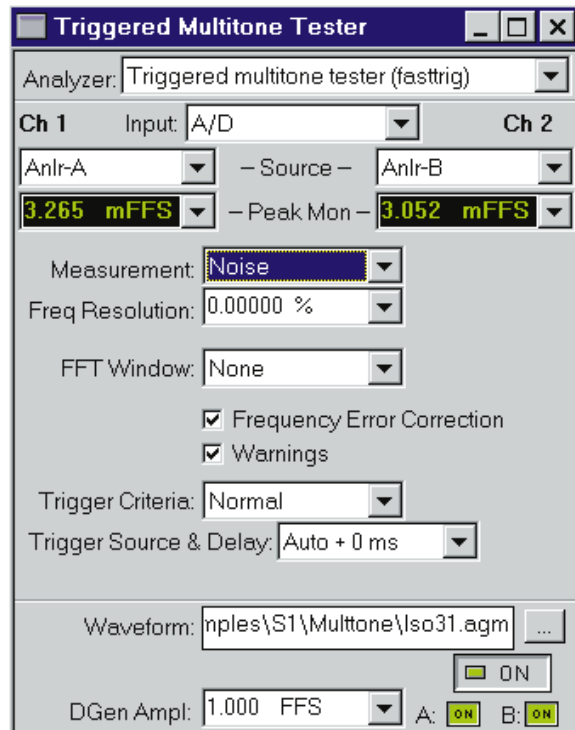



Figure 12-4 FASTTRIG Panel, Large Version




## Trigger Source

Acquisition of signal into any FFT program acquisition buffer may start immediately when  is pressed or Go is clicked, or may wait for a trigger event, depending upon the setting of the Trigger Source field. The Trigger Source field is visible only on the large version of the Digital Analyzer panel.

### Software Sources:

The software triggering source selections are Free Run, Channel #1, Channel #2, Auto, and Digital Generator.

When “Free Run” is selected, signal acquisition begins immediately after  or Go is initiated, regardless of signal amplitude. This is the typical operating mode with steady-state test signals.

The “Channel #1” and “Channel #2” selections have a fixed trigger threshold of 0.1%FS (-60 dBFS) on the channel referred to.

The “Auto” selection will cause triggering if either Channel 1 or Channel 2 has a signal amplitude greater than digital “infinity zero”.

The “Digital Generator” selection functions only on Dual Domain units (SYS-300 series). A Digital Generator trigger occurs as the first sample is read from the waveform file.

## Trigger Criteria (FASTTRIG, CODEC)

FASTTRIG’s and CODEC’s multitone signal recognition and triggering capability is based on comparison of a short sample of received signal with a digital reference copy of the expected multitone signal, as stored in the generator buffers. To determine whether or not to trigger, these programs perform two types of comparisons:

- Using all the tone frequencies represented in the generator buffer as a reference, the DSP looks at the received signal to see if the amplitude at each of a majority of those frequencies is within an acceptable relative amplitude range (plus or minus) of the corresponding component of the reference signal. This criterion allows FASTTRIG and CODEC to ignore simple single-tone test

signals, relatively simple program material such as may be produced by a solo musical instrument, and conditions of silence.

- Across all sections of the spectrum between tones in the reference signal, FASTTRIG and CODEC look at the received signal to assure that its amplitude does not exceed a threshold of acceptability. This criterion allows the DSP to ignore complex voice and music program material, which tends to have energy spread across much of the spectrum.

To permit user control of the triggering criteria, the allowable deviation from reference signal amplitude at generator tone frequencies (1 above) and the amount that energy at all other frequencies must be attenuated (2 above) are settable at three values. The “Triggering Criteria” field allows the user to select Tight, Normal, and Loose conditions. Each selection represents a different trade-off between the chance of false response on non-multitone signals versus the possibility of not triggering on legitimate multitone signals from a device with large amounts of noise and distortion and/or large deviations from flat frequency response. Select “Tight” for the minimum chance of false response. This may be necessary when using very short generator waveform files (less than 2048 samples) since the consequent poorer frequency resolution makes it more difficult to discriminate between multitone

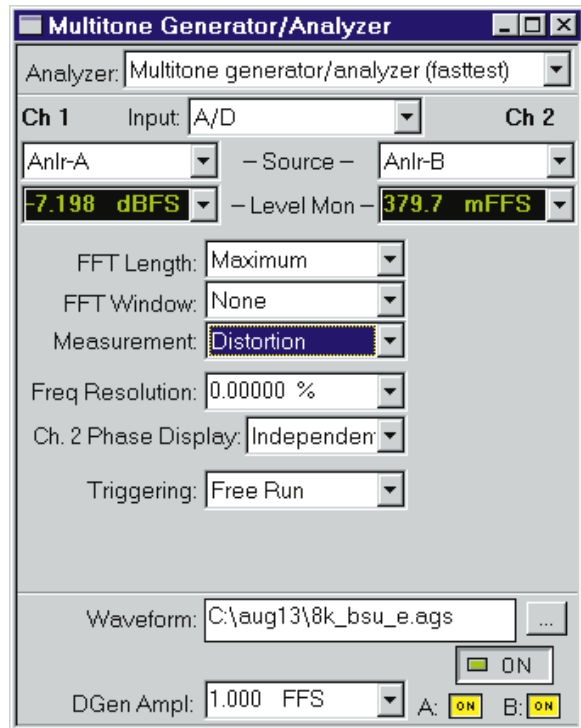


Figure 12-5 FASTTEST Panel, Large Version

signals and program material. Use “Loose” if FASTTRIG will not otherwise trigger on highly distorted or noisy signals or signals passed through narrow-band or otherwise non-flat devices.

### Trigger Source and Delay (FASTTRIG, CODEC)

In-service multitone testing of broadcast transmission circuits and other similar applications involves inserting a brief burst of multitone signal into a short pause in program material. Broadcast transmission circuits typically include modulation processors and other forms of compressors and limiters. It may be desirable to allow a certain stabilization time following the start of the multitone signal before the measurement is made, in order that the measured conditions represent something approaching steady state rather than initial transient conditions. The Trigger Source and Delay field permits control of this stabilization time. The duration of multitone signal transmitted must also be lengthened when non-zero values of delay are used.

To change the triggering source or, with Auto source, the value of delay, click on the down arrow at the right end of the “Trigger Source and Delay” field and select the desired value from the list which appears. The “Trigger Source and Delay” field is visible only on the large version of the Digital Analyzer panel.

The available selections for Trigger Source and Delay are: Auto + 0 ms, Auto + 100 ms, Auto + 200 ms, Auto + 400 ms, Auto + 1 second, Auto

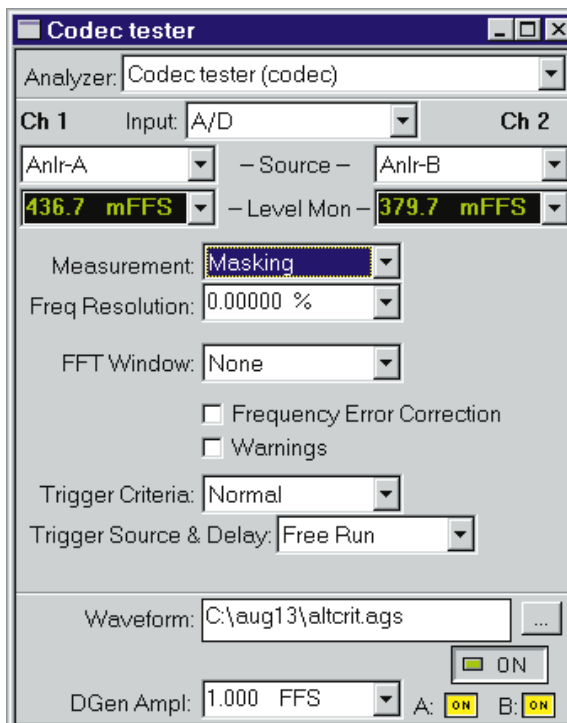



Figure 12-6 CODEC Panel, Large Version

+ 2 seconds, DGen, Extern, and Free Run.

The “Auto” selections all will cause triggering (after the stated time delay) if either Channel 1 or Channel 2 has a signal amplitude greater than digital “infinity zero”. These are the normal operating modes of FASTTRIG and CODEC, with the zero delay selection requiring the shortest signal duration.

When “Free Run” is selected, signal acquisition begins immediately after  or Go is initiated, regardless of signal amplitude.

External: this source is operational only with SYS-322 (Dual Domain) units. It is the signal connected to pin 3 of the 15-pin D-sub connector on the rear of the DSP module. If pin 3 is high (or open circuit, in which case it is pulled high by an internal pull-up resistor), triggering occurs at the next digital sample. Pulling pin 3 low from an external device holds off triggering, with acquisition being triggered on the next sample after pin 3 is pulled high.

The “Digital Generator” selection functions only on Dual Domain units (SYS-300 series). A Digital Generator trigger is issued each time the first sample from the waveform file is generated.

### Signal Duration (FASTTRIG, CODEC)

The minimum duration of multitone signal which must be present to guarantee analyzer triggering (assuming zero delay in the “Trigger Delay” field) depends primarily upon the generator record length used to create the multitone signal. The required signal duration also varies inversely with the sample rate at which the signal was generated (and, in the case of digital inputs, will be received). Small variations in required duration are also a function of whether the signal is acquired from an analog or the AES/EBU serial digital source, and whether or not Frequency Error Correction is in use.

The table below shows the minimum multitone signal duration as a function of generator record length in samples, assuming a 48 kHz sample rate, analog input, and use of the Frequency Error Correction feature. Multiply the values shown by 1.088 if the signal is generated with the 44.1 kHz sample rate and by 1.50 if the 32 kHz sample rate

was used. Multiply the values shown by 1.06 if the AES/EBU input is used.

<b>Generator Record Length</b>	<b>Minimum Signal Duration</b>
8192	1.05 second
4096	520 milliseconds
2048	260 milliseconds
1024	135 milliseconds
512	70 milliseconds
256	40 milliseconds

If a delay value greater than zero is selected in the “Trigger Source and Delay” field, that value must be directly added to the values computed from the multipliers and table values above. For example, with a 2048-sample generator waveform at 44.1 kHz,  $260 \text{ ms} * 1.088 = 283 \text{ ms}$ . With a 200 ms Trigger Delay, the minimum signal duration for reliable triggering would be 483 ms.

There are performance tradeoffs as the generator record length is reduced. The poorer frequency resolution inherent in shorter record lengths results in reduced ability to measure distortion and noise at the lower audio frequencies while using enough tones in the multitone signal to provide good resolution for frequency response measurements. FASTTRIG and CODEC also become more prone to false triggering on program material as short record lengths reduce resolution. The 2048 sample generator record length should probably be considered the shortest for good rejection of falsing and excellent distortion and noise measurements across the entire spectrum down to 20 Hz. The shorter record lengths are useful when testing quiet channels and when testing need not extend down to the lowest audio frequencies.

## Frequency Domain (Spectrum Analysis)

The process of analysis of multitone signals with System One DSP units consists broadly of four steps: acquiring the signal into the acquisition buffer, computing the FFT and putting the results into an amplitude-versus-frequency buffer, post-processing the high-resolution FFT into the specific parameters desired (such as response or

distortion), and displaying the results on the computer screen. FASTTRIG and CODEC can compare the acquired signal to a digitally stored reference copy of the multitone waveform that created the signal. If the frequencies have been shifted due to effects such as tape speed error or different clock rates between two digital systems, FASTTRIG and CODEC can correct the frequency shifts to bring the acquired signal back into a synchronous relationship before the FFT so that no window function is needed. All three Multitone analysis programs can multiply the time domain signal by a Hann window function, but this is not used for synchronous multitone signals. Post processing normally consists of specialized multitone signal processing for fast measurements of response, distortion, noise, phase, and computation of psychoacoustical masking curves.

## Frequency Error Correction (FASTTRIG, CODEC)

A key feature of FASTTRIG and CODEC is their ability to compare the tone frequencies in an acquired multitone waveform with the digital reference copy of the transmitted or pre-recorded waveform presently in the generator buffers. If this comparison shows that the tone frequencies have been shifted up or down due to the signal originating from a device with a different clock frequency from the analyzer or due to analog tape player speed errors, the DSP can correct all the tone frequencies to the reference signal values. This re-creates the original synchronous relationship so that no window function is required before the FFT, and maximum theoretical FFT selectivity is obtained. The maximum frequency difference that can be corrected is  $\pm 3\%$ .

The normal operating mode of FASTTRIG and CODEC is with Frequency Error Correction operating. If it is desired to operate without this feature, the “Frequency Error Correction” checkbox can be unchecked. The Frequency Error Correction checkbox is visible only on the large version of the Digital Analyzer panel.

### FFT Window

The “None” choice in the FFT Window field applies no window function before the FFT. It is sometimes called a rectangular window.

The “None” choice is normally used with synchronous signals in FASTTEST, FASTTRIG, and CODEC.

Normal operation of FASTTEST (and often CODEC) is in testing a local device with both inputs and outputs available at the same location. The same System One unit generates the signal and performs the analysis, with both DSP generator and analyzer controlled from the same internal crystal frequency reference. The generator and analyzer buffer lengths are chosen so as to be synchronous. Every sinewave in the generated signal goes through an exact integer number of cycles in the generator buffer and in the analyzer transform buffer. No windowing function is then required and maximum theoretical FFT selectivity is achieved with full dynamic range available in bins adjacent to a bin containing a full-scale signal.

Normal operation of FASTTRIG (and sometimes CODEC) involves acquisition of a multitone signal generated at a distant location, or previously recorded, from a multitone waveform file by System One with FASTTRIG, FASTTEST, or CODEC loaded. The synchronous relationship exists except that the multitone frequencies may be shifted in frequency either due to small differences in clock rates of the transmitting and receiving System One, or clock rate variance of a digital reproducer, or to tape speed errors in analog tape machines. The frequency error correction feature of FASTTRIG and CODEC should be used to bring the signal back into the synchronous relationship, with no windowing function then required.

## Warnings



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When a generator waveform file is first loaded, FASTTRIG and CODEC analyze the multitone signal described in the waveform file and determine whether it contains enough sinewaves across the spectrum above 200 Hz, at sufficient frequency spacing, to provide reliable frequency error correction on an acquired signal. If the “Warnings” checkbox is checked, a warning message will be displayed whenever a waveform file is loaded which does not meet the built-in criteria. In many cases, the built-in criteria are somewhat conservative and signals with fewer tones or closer spacing may still work properly. Since display of a warning message will interrupt the flow of an automatic test procedure, unchecking the “Warnings” checkbox

provides a means of ignoring the message and thus avoiding hang-up of a procedure when it has been determined that frequency error correction works properly with a particular signal.

## FFT Transform Length

The FFT transform length determines the frequency resolution of the FFT spectrum analysis and the amount of signal which will be acquired. Transform length is an operator-selectable parameter on FASTTEST and is determined automatically from the downloaded generator waveform in FASTTRIG and CODEC.

The FFT length, whether manually or automatically set, controls the record length used as input to the FFT process when either /Go is initiated to acquire and transform, or the  function key or “Sweep Transform Data without Acquire” menu command is used to re-transform any portion of a record previously acquired. Longer transform lengths produce greater frequency resolution in the resulting FFT, but require longer times to acquire and to transform the signal.

To change the FFT Length on FASTTEST, click the down arrow at the end of the field and select the desired value. Available selections are Maximum, 4096, 2048, 1024, 512, and 256. “Maximum” means 16384 on all System One Dual Domain units and on System One Plus DSP units with the MEM (memory) option, but is the same as the “4096” selection on System One Plus DSP units without the MEM option.

For synchronous operation with multitone signals so that window functions are not required, the FFT Length must be an integer multiple of the generator buffer length which produced the multitone signal being measured. For the Noise processing function of FASTTEST to work with multitone signals, the FFT Length must be manually set to twice the generator buffer length which produced the signal. For example, if the signal came from a 1024 sample generator buffer, then the 2048 sample FFT Length must be used. With FASTTRIG and CODEC, the length of the analyzer acquisition buffers for both channels are automatically set at twice the length of the generator waveform file loaded in the Waveform field. This assures the



necessary synchronous relationship and assures that Noise function will operate properly. There is no manual control to set FFT transform length differently from this automatic value.

Note that there are performance tradeoffs as the generator record length and analyzer transform length are reduced. The poorer frequency resolution inherent in shorter record lengths results in reduced ability to measure distortion and noise at the lower audio frequencies while using enough tones in the multitone signal to provide good resolution for frequency response measurements. FASTTRIG and CODEC also become more prone to false triggering on program material as resolution is reduced due to short record lengths. The 2048 sample generator record length should probably be considered the shortest for good rejection of falsing and excellent distortion and noise measurements across the entire spectrum down to 20 Hz. The shorter record lengths are useful when testing quiet channels and when testing need not extend down to the lowest audio frequencies.

## Phase results

An FFT can produce a phase-versus-frequency output in addition to the magnitude (amplitude) versus frequency results. FASTTEST, FASTTRIG, and CODEC permit graphing phase results. FASTTEST can either graph the phase versus frequency of both channels independently, or can graph the difference between phase of the two channels instead of Channel 2 phase. FASTTRIG and CODEC can graph interchannel phase difference, but only Channel 1 phase independently.

### Channel 2 and Interchannel Phase Display

The FFT computes both magnitude and phase arrays as a function of frequency. The phase of coherent signals, such as multitone signals, may be plotted for either or both channels by selecting FASTTEST as the instrument and Ch 1 Phase or Ch 2 Phase as the parameter in the Data browser of the Sweep panel. The DGen trigger source and a sweep table (.ADS file) listing the fundamental signals would be used in this mode. When the Channel 2 Phase Display is selected as “Independent”, the Ch 1 and Ch 2 Phase parameters each show the absolute phase of the fundamental tones.

FASTTRIG and CODEC can plot independent phase only on Channel 1. The DSP computes the phase difference between the Ch 1 and Ch 2 Phase signals at each sweep table value and reports that computed value to the computer as the “Ch.1-2 Phase” parameter in the Data browser. FASTTEST can also plot the interchannel phase difference of stereo signals. Selecting “Interchannel” on the FASTTEST panel causes the “Ch.2 Phase” parameter to contain the interchannel phase data. The Ch 1 Phase parameter is unaffected by the Independent vs. Interchannel setting and thus still plots independent phase of the channel 1 signal.

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## Post-FFT Processing

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At the conclusion of the Fast Fourier Transform, magnitude-vs.-frequency and phase-vs.-frequency data arrays exist in DSP memory for both signal channels. Generally speaking, the entire contents of those arrays are not simply sent to the computer for display. The user-set Start and Stop frequency values on the Sweep panel may call for only a portion of the data to be sent to the computer. The number of Steps on the Sweep panel may be greater or smaller than the number of FFT bins in the Start-Stop frequency span, with either condition requiring some post-FFT processing. In multitone analysis, different forms of processing are used in order to extract frequency response, phase response, total distortion and noise, noise in the presence of signal, and (in the case of CODEC) psychoacoustical masking curves from the FFT data.

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### Displayed Points Do Not Equal FFT Bins

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The “Maximum” FFT Length is 16384 samples with full memory, which results in 8192 FFT bins (spectral lines). Since 1024 pixels is a rather high graphic resolution (in 1995, at any rate) and the portion of the screen dedicated to the graph is only a portion of that number, it is typical for the number of Steps at Source 1 on the Sweep panel to be set to 500 or less. Thus it is common to have many FFT bins represented by one displayed point. In order to avoid missing significant signals, the DSP employs a form of post-FFT processing known as “Peak Picking”. When the computer requests data at two points which span more than one FFT bin, the DSP scans across all

FFT bins between the two requested frequency points, finds the greatest bin amplitude, and sends that value to the computer to be graphed as the amplitude of the new point.

When the user has set Start and Stop frequencies to a small span, or zoomed on the graph into a small frequency span, the opposite situation may occur—there may be many displayed points per FFT bin. The computer may be thus request data several times within the same FFT bin. In this case, the DSP simply returns the same bin amplitude value to each request that falls within the bin boundaries. With the typical 48 kHz sample rate and maximum 16384 sample FFT, for example, the width of each FFT bin is about 2.93 Hz. If the graph has been zoomed in to a 100 Hz linear span with 500 steps, approximately 15 displayed points fall into each bin. The resulting graph has the appearance of a city skyline with rectangular flat-topped sections representing the width and amplitude of each bin.

## Multitone Post-FFT Processing

The key features that distinguish the multitone analyzers from the general-purpose spectrum analyzers are processing done in the DSP after the FFT is performed. This post-processing extracts commonly desired audio test parameters such as frequency response, distortion, and noise from the large volume of data produced by the FFT.

### Measurement (FASTTEST, FASTTRIG, CODEC)

The Measurement field controls the type of post-processing done to FFT results before they are sent to the computer for display and possible limits comparison. The four selections are Spectrum, Response, Distortion, and Noise. To change the selection, click on the down arrow at the right of the Measurement field, then click on the desired selection.

**Spectrum:** this selection provides a normal FFT spectrum display with no processing except for “peak picking”. The Spectrum selection is typically used without a sweep table (.ADS file), and with a relative large number of Steps at Source 1 of the Sweep panel to provide good frequency resolution. Typical Steps values are from 250 to 500. If the transform length in use results in more FFT bins in the Start-Stop

frequency span being plotted than the number of Steps, peak-picking takes place. With peak-picking, the DSP searches all FFT bins between the previous plotted point and the point presently being plotted and sends the highest bin amplitude in that range as the amplitude of the new point to be sure that no signals are missed.

**Response:** this selection is always used with a sweep table (.ADS file) which lists the exact frequencies of the sinewaves in the multitone signal which are to be used for frequency response measurements. The DSP sends to the computer to be plotted only the amplitudes of the FFT bins containing those exact frequencies, resulting in a frequency response graph. There are typically from 3 to 30 sinewaves in most multitone signals.

If the value in the Frequency Resolution field is greater than zero, the DSP performs an RSS (root-sum-square) integration of all the bin amplitudes within plus or minus the Frequency Resolution value around each sweep table frequency and sends the integrated sum value to the computer to be plotted. This mode is intended for frequency response measurements on devices such as analog tape recorders that introduce frequency modulation (flutter) to signals. Flutter spreads each tone's energy across a small region of the spectrum. This reduces the amplitude of the fundamental tone, since the total energy in the fundamental and all sidebands remains constant during frequency modulation. The RSS summation of FASTTEST combines this spread energy back into a single value, much as the human hearing system responds to signals with small amounts of FM.

**Distortion:** this selection is always used with a sweep table (.ADS file) which lists the exact fundamental frequencies of the sinewaves in the multitone signal being used. The DSP sends to the computer to be plotted the RSS (root-sum-square) summation of all the FFT bin amplitudes between the previous sweep table frequency and the present sweep table frequency, so that the plotted point represents the integration of all the noise and distortion products across each frequency span between fundamentals but not including the amplitude of the fundamentals.

If the value in the Frequency Resolution field is greater than zero, the DSP excludes all the bin amplitudes within plus or minus the

Frequency Resolution value around each sweep table frequency before sending the integrated sum value to the computer to be plotted. This mode is intended for distortion measurements on devices such as analog tape recorders that introduce frequency modulation (flutter) to signals. Flutter spreads each tone's energy across a small region of the spectrum. If these close-in sidebands which fall outside the bin containing the fundamental are not to be measured as distortion, they must be excluded, much as the human hearing system masks low amplitude signals nearby in frequency to a stronger signal.

Distortion function differs between FASTTEST on one hand and FASTTRIG/CODEC on the other and in that no sweep table is required for FASTTRIG and CODEC. This is because FASTTRIG and CODEC “know” from the reference signal in their generator buffer exactly what the multitone fundamental signal frequencies are. They automatically exclude those bins from their distortion summation. It is possible to integrate distortion with FASTTRIG and CODEC across arbitrary portions of the spectrum. (FASTTEST does not have this feature and relies on a sweep table, excluding from distortion measurements the bin corresponding to each frequency entry in the sweep table.) All other bin amplitudes are summed (RSS) between each adjacent pair of frequencies requested from the DSP by the computer. Distortion and noise can thus be summed across spans determined by the Sweep panel Start, Stop, Log/Lin, and number of Steps, or the spans can be determined by a sweep table. If it is desired to sum the noise and distortion into critical bands, a sweep table can be used which defines the edges of the human hearing system critical bands.

If the value in the Frequency Resolution field is greater than zero, the DSP excludes all the bin amplitudes within plus or minus the Frequency Resolution value around each sweep table frequency before sending the integrated sum value to the computer to be plotted. This mode is intended for distortion measurements on devices such as analog tape recorders that introduce frequency modulation (flutter) to signals. Flutter spreads each tone's energy across a small region of the spectrum. If these close-in sidebands which fall outside the bin containing the fundamental are not to be measured as distortion, they must be excluded, much as the human hearing system masks low amplitude signals nearby in frequency to a stronger signal.

With CODEC, the distortion and noise curve is normally compared to the composite masking curve generated in Masking function.

Noise: this selection may be used with a sweep table (.ADS file) listing the fundamental frequencies of the multitone signal in use, but need not be. Noise mode depends on the Transform length being set to the value twice the length of the waveform file that generates the multitone signal. The analyzer frequency resolution is thus twice the resolution of the generated signal. The result is that every alternate analyzer FFT bin falls between bins at which the generated signal could contain fundamentals or bins into which harmonic or intermodulation distortion products due to the generated signal fundamental signals could fall (assuming that the device under test does not shift fundamental frequencies or produce frequency modulation). The amplitudes of these alternate “empty” bins consists of noise generated in the device under test, largely unaffected by fundamental signals or distortion. If the same sweep table is used in Noise mode that is used for response and distortion measurements, the resulting graph will be a spectrum analysis of noise in the presence of test signal. If a two-point sweep is made with Start at 20 Hz and Stop at 20 kHz, for example, the plotted value at 20 kHz represents the RSS integration of all “empty bins” across the audio band

Masking: this selection generates a composite masking curve for the particular multitone signal in use. The shape of the curves is based on a model published by psychoacoustician Brian Moore in the Proceedings of the AES 12th International Conference, June 1993, pp 22-23. The shape of the curves varies with frequency. The center frequency of each section of the composite masking curve is located at the fundamental frequencies present in the waveform file downloaded to the CODEC generator buffer. The measured amplitude at each fundamental frequency determines the reference amplitude at each frequency. The masking curve is normally used by saving it as a limit (.ADL) file, then comparing a Distortion function curve (usually with critical band spacing) to that limit curve.

## Frequency Resolution

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The Frequency Resolution field is a numeric entry field with % units. The user may enter values up to 5% in FASTTEST and

FASTTRIG and 13% in CODEC which are used in Response and Distortion Measurement functions.

In Response function, the amplitudes of all FFT bins within plus and minus the Frequency Resolution value of each sweep table value are combined in RSS (root-sum-square) fashion and furnished to the computer as the integrated amplitude of the bins within that range. The purpose of this function is to provide accurate frequency response measurements of devices with wow and flutter. Wow and flutter spreads the energy from a single tone across a narrow spectral band.

In Distortion function, the amplitudes of all FFT bins within plus and minus the Frequency Resolution value of each sweep table value are excluded from the RSS computation of energy falling between tones. Distortion function defines all signals other than the fundamental tones as distortion and noise. Entering a non-zero value of Frequency Resolution causes flutter sidebands to not be included in the distortion measurement.


### **Transform Data without Acquire**

FFT-based (batch mode) DSP programs have three distinct, sequential phases to their operation when the  key is pressed or Go button or green light icon is clicked. First, data is accumulated into the acquisition buffer until the buffer is filled to the specified acquisition length. Second, a Fast Fourier Transform (FFT) is performed to obtain amplitude (and sometimes phase) versus frequency data which is stored in a different memory buffer from the acquired signal (amplitude versus time). Third, a post-processed version of the amplitude versus time or amplitude versus frequency data (depending upon Sweep panel settings at Source 1 and Data 1 or 2) is transmitted from the DSP module in the test system to the computer for graphing by APWIN software.

It is often desirable to re-transform the data already present in the acquisition buffer, rather than acquiring new data. This will be the case when a previously acquired waveform is downloaded from computer disk to the DSP module for further analysis. It will also be the case if it is desired to re-transform the existing contents of the acquisition buffer while using a different window function. Re-transformation is




frequently used with the MLS Quasi-Anechoic Acoustical Analyzer when going back and forth between time domain (impulse response) and frequency domain views of an acquired signal.


The Transform Data without Acquire command on the Sweep menu, or the equivalent  keystrokes, cause the second and third phases described above to take place (FFT transformation and transmission of results to computer).

## Reprocess Data

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FFT-based (batch mode) DSP programs have three distinct, sequential phases to their operation when the  key is pressed or Go button or green light icon is clicked. First, data is accumulated into the acquisition buffer until the buffer is filled to the specified acquisition length. Second, a Fast Fourier Transform (FFT) is performed to obtain amplitude (and sometimes phase) versus frequency data which is stored in a different memory buffer from the acquired signal (amplitude versus time). Third, a post-processed version of the amplitude versus time or amplitude versus frequency data (depending upon Sweep panel settings at Source 1 and Data 1 or 2) is transmitted from the DSP module in the test system to the computer for graphing by APWIN software.

It is sometimes desirable to perform a new post-processing of the FFT results and transmission of the post-processed data to the computer for display, without either acquiring new data or re-transforming the data already present in the acquisition buffer. This is faster than re-transforming followed by new post-processing. It is automatically done (unless turned off in the Utilities Configuration menu) when zooming in on a section of an FFT display and wishing to obtain optimum resolution of the new sub-set of the original data, or when changing multitone post-processing methods between the normal (response and level), distortion, or noise functions.

The Reprocess command on the Sweep menu, or the equivalent  keystrokes, cause the third phase described above to take place (post-processing of FFT results and transmission of results to computer).




## Re-Process FFT Data on Zoom

When this box is checked, each graph zoom in action (by dragging the mouse) or Zoomout or Zoomout to Original action (right mouse button menu on graph) will be accompanied by a “Sweep Re-Process Data” () command. If the acquired signal is still in DSP memory, this assures that the resolution is always the maximum supported by either the number of points graphed or the FFT transform length.

## Saving and Loading Waveforms

Both channels of acquired waveform remain in the DSP Acquisition buffer until one of the following events:

-  (Go, Sweep Start, etc) is pressed to acquire new waveforms
- a different DSP program (or no DSP program) is loaded, by changing the “Analyzer” field contents on the Digital Analyzer panel
- power is turned off to System One
- different waveforms are downloaded from disk as described in this section.

While the waveform is still present in the Acquisition buffer, full flexibility exists to change back and forth between time and frequency domain analysis (FFT Time vs. FFT Frequency at Source 1), to view different time portions of the record, to perform FFTs of different transform length (FASTTEST only), to change from no window to Hann window, to change FFT post processing modes, to change time domain processing modes, etc. The results of any and all of these different modes of analysis may be saved, one at a time, as .AT1 test files to disk. However, once the waveform is gone from DSP memory, no further ability exists to analyze the signal other than re-loading .AT1 test files to look at the specific analyses which were saved. If full flexibility for continued analysis is desired, the acquired waveform itself may be saved to disk, rather than merely saving .AT1 files each of

which represents one particular processed result of one type of analysis. The saved waveform can then later be downloaded to the DSP Acquisition buffers for further analysis. Single-channel waveforms saved at different times can later be individually loaded into the two channels of Acquisition buffer for comparison in the time or frequency domains.

## Saving Waveforms

Selected sections or the entire contents of the Acquisition buffers may be saved to a disk file by the File Save As menu command, selecting either the “Stereo Waveforms (\*.AAS)” or “Mono Waveform (\*.AAM)” as the Save As file type.

A Stereo Waveform file (.AAS) consists of two waveforms joined together into a single file. A Mono Waveform file (.AAM) contains a single waveform. To simultaneously save the waveforms from both Acquisition buffer channels, select the Stereo choice. To save only one of the channels, select Mono.

When the waveform file type is selected, a file name is typed in by the user, and the Save button is clicked, a dialog box as illustrated in Figure 12-7 will appear. The left half of the dialog box defines what will be saved into the first section of a waveform file and the right half defines what will go into the second section.

### Full Acquisition Buffer vs. Transform Sub-Section

The choices in each section are None, Acquired-Ch 1, Acquired-Ch 2, Transform-Ch 1, and Transform-Ch 2. When saving a Mono file, the “None” selection will be volunteered for

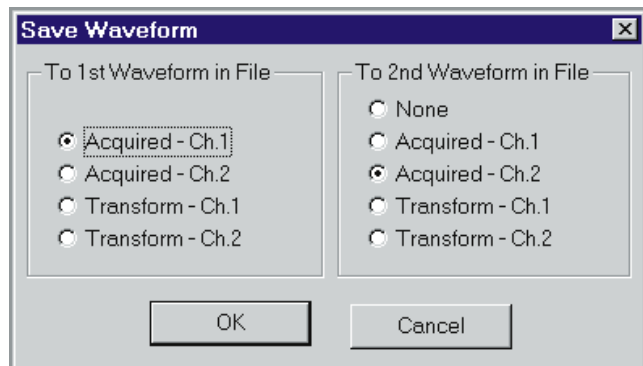


Figure 12-7 Save Waveform Dialog Box

the right half. The “Acquired” choices will result in saving the entire contents of an Acquisition buffer to disk. The “Transform” choices result in saving only the sub-section of the Acquisition buffer specified by the present FFT Length selection on FASTTEST. If the FFT Length of FASTTEST is set to Maximum at the time the waveform is saved, and with CODEC and FASTTRIG, there will be no difference between the “Acquired” and the “Transform” selections. If a smaller section of the waveform is sufficient for the intended future analysis purposes, use of the “Transform” selections can result in much smaller disk files than the “Acquired” selections. DSP memory width is 24 bits and computer disk files are organized around 8-bit bytes, so the length of monaural and stereo disk files are respectively three times and six times the length of buffer they were saved from. One channel from a 16,384-sample acquisition buffer will thus result in approximately a 48k file size, while use of the “Transform” selection with a selected Transform length of 1024 samples would result in about a 3k file size.

### One vs. Two Waveforms

The selections in the left half of the dialog box determine which buffer contents go into a Mono waveform file or into the first section of a Stereo waveform file. The contents of the second section of Stereo waveform files are defined by the settings of the right half of the dialog box. If a Ch 1 selection is made in the left side and a Ch 2 selection in the right side, a Stereo waveform file results. It is normally preferable for the two sections of a Stereo waveform file to contain records of the same length, so normally one would not use the “Acquired” selection for one section and the “Transform” selection for the other.

### Combining Mono to Stereo

It is possible to save single channel waveforms acquired at two different times and then combine them into a Stereo waveform so that simultaneous time or frequency domain analysis may be made for some “before and after” conditions. Save a Mono waveform from the first set of conditions, specifying the channel number where the desired data is presently located. Under different test conditions or with a different device under test, save another Mono waveform with a different name. Use the Open Waveform feature described below to load one of these waveform files into the Ch 1 Acquisition buffer and

the other into the Ch 2 Acquisition buffer. Then, use the Save As command, select Stereo waveform and supply a new file name, and select the Ch 1 waveform into the first section and Ch 2 into the second.

It is also possible to use the DOS Copy command to concatenate two Mono waveform files into a new Stereo file. The command (from the DOS prompt) is:

```
COPY /B FILENAME1.AAM + FILENAME2.AAM NEWFILE.AAS
```

## Loading Waveforms

Existing waveform files may be downloaded from computer disk to the DSP via the File Open command. In the File Open dialog box, select the Stereo Waveforms (.AAS) or Mono Waveforms (.AAM) file type and select the desired directory and file name. Clicking the Open button then displays the Open Waveform dialog box as illustrated in Figure 12-8. The left half selections determine which DSP buffer the first section of the waveform file will be loaded into. The right half of the dialog defines which buffer the second section of the waveform file loads into. The “Acquired” and “Transform” selections work similarly to the Save Waveform operations described above. Clicking OK then actually downloads the waveform file contents. After the waveform is loaded, the **F6** and **CTRL F6** keys and settings on the Digital Analyzer panel and Sweep panel may be used as described earlier for continued analysis of the waveform.

If the downloaded file length is greater than the length of the specified buffer, an error warning will be displayed. The portion that fits in the buffer can be analyzed normally and the remainder

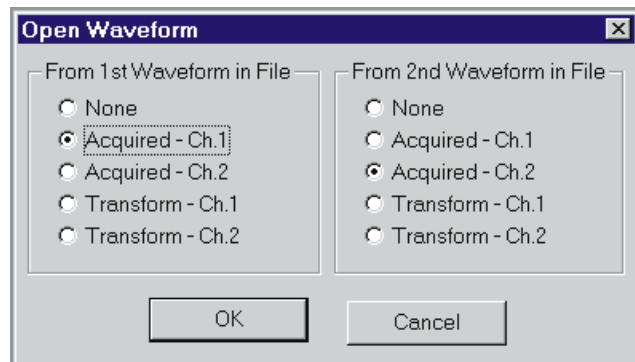


Figure 12-8 Open Waveform Dialog Box

cannot be viewed. If the downloaded file length is shorter than the buffer size, erroneous analysis will result if the selected FFT Length extends beyond the last data sample in the buffer. The contents may first be displayed in Time Domain mode to determine exactly where the signal ends.

## Signal Generation

FASTTEST, FASTTRIG, and CODEC can generate digital domain waveforms in addition to acquiring and displaying waveforms and performing FFT spectrum analysis. The waveforms generated are defined by downloading files selected as the waveform (“Wfm”) in arbitrary waveform generation mode.

To load a waveform file into any of these programs, click on the Ellipsis button at the right end of the “Wfm” (waveform) box. A dialog will appear which permits navigating through directories (folders) and selecting the file name of the desired waveform file. Clicking on Open will open that file and download it to the signal generation buffer. The signal defined in the waveform file will immediately start being generated as the program steps through the samples in the waveform file at the sample rate selected in the Rate field on the Digital I/O panel. The signal will be generated continuously as the DSP steps from the last sample to the first sample. This in effect creates a continuous loop, “splicing” the waveforms together.

Many multitone waveform files are furnished with APWIN software. Users may also easily create custom waveform files from an ASCII table of frequency and amplitude (and phase, if desired) values. See Creating Waveform Files with MAKEWAV2 later in this chapter for more information.

### Stereo vs. Mono Waveforms

FASTTEST, FASTTRIG, and CODEC generator waveform files may consist of one waveform (monaural) or of two waveforms (stereo). Monaural waveform generator files have the extension .AGM and stereo generator waveform files have the extension .AGS. When a mono waveform file is opened in the Waveform browser, the

waveform is loaded into both the generator buffers. When a stereo waveform file is opened, the first waveform in the file is loaded into the generator channel A buffer and the second waveform is loaded into the channel B buffer.

## Digital vs. Analog Output

---

The generated signal may be routed to any of System One Dual Domain's digital interface output connectors, as controlled from the Output Format and Serial Type fields on the Digital I/O panel. By choosing the Output Format "D/A" selection on the Digital I/O panel, the generator channel B output signal may instead be converted to the analog domain by a built-in monaural D/A converter. The converter output feeds the D/A Output BNC connector and may be routed through the Analog Generator output stage by the DGen waveform selection in the Analog Generator Waveform field.

## Digital Generator Amplitude

---

The amplitude of the digital domain signal is controlled by the DGen Ampl field. This is true whether the signal is an algorithmically-generated sinewave or from a downloaded generator waveform file. The available units are the standard digital domain units of FFS, dBFS, %FS, and bits.

When an analog output signal is converted by the D/A selection in the Output Format field of the Digital I/O panel, the input amplitude to the D/A converter and thus the analog output amplitude is controlled by the DGen Ampl field. If the analog signal is routed through the Analog Generator output stage by the DGen selection in the Waveform field of the Analog Generator panel, the DGen Ampl field value must be set to full scale if the Analog Generator amplitude calibrations are to be correct.

## On/Off and Channel Select

---

The OFF/ON button near the bottom right corner of the Digital Analyzer panel controls the digital output signal flow from the digital generator. No signal output is present when the button reads OFF and the "LED" is not "lighted". Clicking on the button turns on the generator output and the button label changes to ON with the "LED"

green. Note that when analog signals are supplied from this program via the “D/A” Output Format selection on the Digital I/O panel, this control must be ON in order to obtain analog signals.

The two ON/OFF buttons labeled A and B, under the main generator ON/OFF button, provide individual control of each channel (subframe) when digital domain audio is generated. Each button permits output signal flow to its respective channel when the button is yellow and indicates ON. When the button is gray and indicates OFF, no signal is supplied to that channel. When analog signals are supplied via the “D/A” Output Format selection on the Digital I/O panel, the status of these A and B buttons is not important. Signal feeds the D/A converter even if both A and B buttons are set to OFF.

## Multitone Generation From Files

Many waveform files are furnished with APWIN and System One, for applications such as audio instrument calibration, multitone signal generation for production test and broadcast systems testing, and low bit rate coder testing. These files must be 8192 samples long for FFTGEN, and can be 8192 samples or shorter binary powers (4096, 2048, 1024, 512, 256) for FASTTEST, FASTTRIG, or CODEC. If the user has special needs not met by any of the furnished waveform files, custom files may be created by use of the MAKEWAV2 utility.

### Creating Waveform Files with MAKEWAV2

A utility program named MAKEWAV2 creates files to generate the multitone signals and .ADS tables used by FFTGEN, FASTTEST, FASTTRIG, and CODEC. Even though many waveform files are furnished with these two programs, the user may desire to create his own.

MAKEWAV2 operates with a standard APWIN data export file (.ADX) or the equivalent created with a word processor or text editor as its input. MAKEWAV2 creates two output files; the downloadable monaural generator waveform file (.AGM) and an ASCII file list (.ADF file) of exact fundamental frequencies suitable for importing into APWIN in order to create a matching sweep table.

## Creating a .ADX File

The starting point to create a custom multitone waveform is entering the desired frequency, amplitude, and (optionally) phase information into the APWIN Data Table. When the information has been created, use the File Export menu command to create an ASCII export (.ADX) file. Alternately, any ASCII text editor could be used if the .ADX file format including the header lines is duplicated.

To generate the waveform file specifications Source 1 (the first column in the Data Table) must be Frequency, in Hz units. Data 1 (the second column in the Data Table) must be an amplitude value, obtained by selecting either the Level meter or an amplitude function of the Reading meter as the Data 1 instrument and parameter. The second column units may be one of the decibel units such as dBV, dBu, dBr, etc. if you wish to specify amplitudes in dB, or Volts if you wish to work in linear units. Which decibel unit you use is not important since it is used only to specify the relative amplitude of the sinewaves to be generated. The absolute amplitude will be determined when MAKEWAV2 is run. If you wish to also specify the starting phase of each sinewave to be generated, Data 2 must be selected as the Analyzer with Phase as the parameter. If you are willing to have all signal components start at zero degrees phase, it is not necessary to select a Data 2 instrument and parameter. Below the column headers, type one row in the Data Table for each sinewave desired in the multitone signal. Imbedded commas must not be used. Use a minus sign for negative decibel or phase values. Values without a minus sign will be interpreted as positive, or you may enter an explicit plus sign. When complete, use File Export to save the file to disk with a descriptive name and the .ADX file extension. The five-frequency file illustrated below was saved as FIVE.ADX.



```

C:\APWIN\five.at1, 08/31/95 14:10:48
Gen.Freq,      Anlr.Level A,      ,,,,,,
Source 1,      Data 1,              ,,,,,,
Hz,            dBV,                ,,,,,,
70,            0,                  ,,,,,,
270,           0,                  ,,,,,,
1020,          0,                  ,,,,,,
3920,          0,                  ,,,,,,
15000,         0,                  ,,,,,,

```

At this point, use MAKEWAV2 as described below from the Windows icon or directly from the DOS prompt.

### Using MAKEWAV2

MAKEWAV2 operates with a data (.ADX) file as input and creates two output files. **The input .ADX file must be located in the C:\PROGRAM FILES\APWINBIN directory and the output files will be placed in that same directory.** One output file is a binary waveform file that will be selected in the waveform (“Wfm”) field of the digital analyzer program in use for generation of the multitone signal. The second is an ASCII .ADF file (meaning data fundamentals) containing a list of the exact fundamental frequencies that make up the multitone signal. This is an intermediate step toward production of the sweep table (.ADS) which will be used during data analysis.

MAKEWAV2 is run by clicking on the icon in the Audio Precision APWIN Program Group (Windows 3.11) or clicking on the Start button, then Programs, then the Audio Precision APWIN group, and finally the Audio Precision Makewav2 program name. A dialog box will appear;

the figure shows the Windows 95 version. Supply the name of the .ADX file to

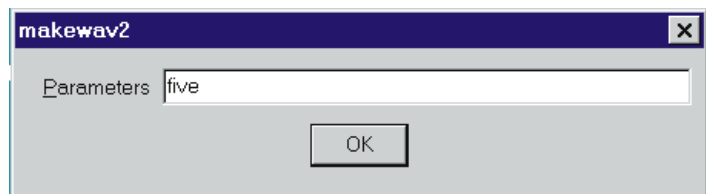


Figure 12-9 Makewav2 Dialog for Entry of File Name and Command Line Options

use as input, plus any desired command line options as discussed below. The input file must be in the C:\PROGRAM FILES\APWINBIN directory.

MAKEWAV2 may also be run from the DOS prompt by typing MAKEWAV2 followed by the .ADX file name to be converted and any command line options. For example:

```
MAKEWAV2 FIVE 
```

for the case of FIVE.ADX created above. It is not necessary to supply the .ADX file extension for the input file nor file names for the output files, if you are willing for the output files to carry the same names as the input .ADX file except for the file extension. In the example above, MAKEWAV2 will automatically name the two output files FIVE.AGM and FIVE.ADF. The .ADF file for the above example is reproduced below:

```
C:\APWIN\five.at1, 08/31/95 14:10:48
Gen.Freq,      Anlr.Level A,      ,,,,,,
Source 1,      Data 1,          ,,,,,,
Hz,            dBV,              ,,,,,,
70.3125,       0,                 ,,,,,,
269.531,       0,                 ,,,,,,
1019.53,       0,                 ,,,,,,
3919.92,       0,                 ,,,,,,
15000,         0,                 ,,,,,,
```

Note that the requested frequencies have been rounded off by MAKEWAV2 to the nearest “legal” frequency (integer multiple of 5.8593772 Hz for the default 48 kHz sample rate and 8,192 sample record). Every sinewave must go through an exact integral number of cycles in the generator buffer so that no discontinuity is created when the generator “splices” from the end to the beginning of the record. Control can then be shifted to APWIN and FIVE.ADF selected as the generator waveform file with the Waveform field near the bottom of the FFT program to be used (FFTGEN, FASTTEST, FASTTRIG, or CODEC).

To create a matching sweep table, use the File Import command to load FIVE.ADF into the data space. The same units used when creating the original .ADX file should be chosen at Data 1, Data-2, and Source 1 to avoid an error message when the .ADF file is loaded. The File Save As command can now be used, the Sweep Tables (.ADS) file type selected, a file name furnished, and the Save button clicked to make a sweep table.

If you wish the output files to have different names, you may supply those in the Parameters field when running MAKEWAV2 under Windows or on the command line when running MAKEWAV2 from DOS. However, file management is normally simplified by using the same file name for the original .ADX file, the waveform (.AGM), fundamental frequency list (.ADF), and the resulting sweep (.ADS) files containing fundamental frequencies.


MAKEWAV2 makes a large number of complex computations in generating the .AGM file. The speed with which it runs depends strongly upon the type of computer used and presence or absence of a math co-processor in the computer. It also depends upon the number of signal frequencies specified, plus several of the options described below. Execution time of MAKEWAV2 with a 386-based computer with co-processor can range from a few seconds for .ADX files containing only a few frequencies to a few minutes for complex files. Operation speed can be improved by use of some of the command line options.

### **MAKEWAV2 Command Line Options**

There are a number of options for operation of MAKEWAV2, summarized below. These options may be viewed on screen by double-clicking the MAKEWAV2 icon and then clicking the OK button without supplying anything in the Parameters box, or from DOS by typing MAKEWAV2 /H (for help) or MAKEWAV2 /?.

/L #	Waveform repetition length (lowest period) in samples. (default and max = 8192).
/T	Truncate frequencies (default = round).
/R #	Sampling rate in Hz. (default = 48kHz).
/O #	Oversample ratio. (default = 8).
/F #	Number of coefficients (Taps) in filter. MUST BE ODD. (default = 401).
/M #	Margin (Headroom) to allow, in dB (default = 1.0).
/W	Cancel waveform output (Do just .ADF file).
/C #	Conversion type. (default = 1) 1 = .ADX to .AGM & .ADF, 2 = .ADX to .WAA & .ADF 3 = .WAA to .AGM).
/S	Record size in samples (file length)
/A	Absolute amplitude (dBFS) in output file of a unity (one Volt or zero dB) value in input .ADX file

To use one or more of these options, simply type them (separated by one space) on the command line, following the file name to be processed. For example, using FIVE.ADX as the input file and specifying the waveform length, record size, and sample rate values would be accomplished as follows:

```
MAKEWAV2 FIVE /L2048 /S8192 /R32000 
```

### Waveform Length/Record Size

To avoid using a window during FFT processing, the generated waveform length and the analysis record length must be synchronous. This is achieved when the generated waveform length is exactly equal to or an exact integer sub-multiple (1/2, 1/3, etc.) of the analysis record length.

FASTTEST, FASTTRIG, and CODEC have only a few specific analysis record length choices available, all binary powers (from 8192 down to 256 samples). The required synchronization between generated and acquired record lengths must be established by controlling the generated record length appropriately for the intended

acquisition record length. The /L option of MAKEWAV2 determines the period length, in samples, of the lowest frequency sinewave which can be generated. All other frequencies which MAKEWAV2 generates will be exact integer multiples of this basic frequency. The /L option thus determines the potential generator frequency resolution and the minimum analyzer acquisition buffer length.

The record size option (/S) determines the size in samples of the waveform file generated. If the waveform length and record size are equal, the lowest frequency possible would go through exactly one cycle in the waveform file. If the waveform length is an integral sub-multiple of the record size, the waveform file would contain two or more exact cycles of the lowest possible frequency. MAKEWAV2 will not permit the waveform length to be greater than the record size. The /S option must be used to create the 2048-sample generator record size required to operate FASTTEST in an SYS-222 without MEM option. FASTTEST with full memory requires an 8192 sample record size (/S8192). The /S option is also used to create shorter generator records for FASTTRIG in order to permit testing with shorter burst lengths of signal.

### Truncate vs. Round

With the default rounding operation, MAKEWAV2 will round each user-requested frequency to the nearest “legal” frequency, with a consequent maximum discrepancy of one-half the basic frequency (2.93 Hz with 8192 samples and 48 kHz). If truncation is selected by the /T option, MAKEWAV2 will select the largest “legal” frequency equal to or smaller than the user-requested frequency. The resulting discrepancy can approach the basic frequency (5.859 Hz in this case).

### Sampling Rate

The sampling rate selected when generating files for the waveform and sweep table must be the exact sampling rate that will be used during acquisition and FFT analysis of the signal. Failure to use the same sampling rate during MAKEWAV2 operation and actual waveform generation from FASTTEST, FASTTRIG, or CODEC will scale all frequencies up or down by the ratio of the two sampling rates.

## Oversample Ratio

In producing the .AGM file, MAKEWAV2 computes the peak amplitude of the complex signal and then scales the amplitudes of all components so that the peak approaches but stays below digital full scale. The oversample ratio used affects the accuracy with which MAKEWAV2 predicts the amplitude of the complex waveform. This oversample ratio effectively determines the interpolation between waveform samples. A higher oversample ratio reduces the likelihood that peaks will fall between interpolated points, but increases the execution time of MAKEWAV2. If peaks are missed, the actual complex waveform will have peaks higher than predicted by MAKEWAV2 and clipping could occur. Operation of MAKEWAV2 can be significantly speeded by using smaller oversample ratios, but with increased risk that actual complex signal peaks may exceed full scale. For applications where use of the full dynamic range is not important, the headroom option (see below) can be used in conjunction with a low oversample ratio to save time during waveform file generation. Oversample ratio must be an even number; two is the minimum acceptable.

## Digital Filter

This option determines the number of coefficients (taps) in the digital filter used in the computation of complex output waveform amplitude. A higher number of coefficients produces a more accurate estimate of the complex signal amplitude, but increases the execution time of MAKEWAV2. Operation of MAKEWAV2 can be significantly speeded by using smaller numbers of taps, but with increased risk that actual complex signal peaks may exceed full scale. For applications where use of the full dynamic range is not important, the headroom option (see below) can be used in conjunction with a lower number of taps to save time during waveform file generation. The number of taps must be an odd number.

## Headroom

After computing the peak amplitude of the complex signal, MAKEWAV2 sets that computed peak below digital full scale by an additional margin for headroom. The default value is one dB. If the

oversampling and filter taps were sufficient to accurately estimate the peak amplitude, this default margin may be decreased in order to obtain a small improvement in signal-to-noise ratio in the signal. If lower values of oversampling and a smaller number of filter taps are used to speed execution of MAKEWAV2, a headroom value larger than one dB should be used to avoid the chance of clipping the generated signal.

### No Waveform

If the waveform files (.AGM) are not desired, but merely the .ADF file which ultimately leads to .ADS tables, MAKEWAV2 will accomplish this very rapidly by using the /W option. This is useful to permit trial-and-error experimentation in finding the best frequency list without taking the time to generate the waveform file.

### Absolute Amplitude

When the /A option is not used, MAKEWAV2 automatically sets the time domain peak amplitude of the output .AGM file to -1.00 dBFS. The one-dB headroom allowance is to cover possible small variations between the composite peak as computed by MAKEWAV2 versus the actual amplitude at the output of the reconstruction filter following the D/A converter in the DSP module.

When creating waveforms for stereo separation tests, it is strongly desirable that the sinewaves in both left and right channel waveforms have the same amplitude. If they do not, a device will appear to have stereo imbalance. The /A option is used to force this equal amplitude condition. It is normally used by first running each of the .ADX files (left and right channel) through MAKEWAV2 without using the /A option, and noting the last message on the screen when MAKEWAV2 runs. This message reads:

“Absolute output level = -nn.nn dBFS (corresponding to 1 Volt or 0 dB input file specification)”

For the typical case of an input .ADX file with all amplitudes specified at zero dBV (or zero in any other dB unit) and no positive dB values, this value shows the highest that a zero dB input sinewave

amplitude can be placed without the composite signal exceeding -1.00 dBFS. Note the numbers from the left and right channel waveforms. Then, run both .ADX files again, using the /A option with the same numeric value for both channels. The numeric value must be chosen so that neither waveform will exceed full scale. For example, if the left waveform produces an absolute level of -20.43 dBFS and the right waveform produces an absolute level of -21.29 dBFS, run them both again with the /A-21.29 argument. The right channel will now still have about 1.0 dB of headroom; the left channel will have about 1.86 dB of headroom, and the sinewave amplitudes will be equal on both channels.

### **Making Sweep Files from .ADF Files**

---

To create a .ADS file from an .ADF file, use the File Import command with Source 1, Data 1, and (if phase was specified) Data 2 set to the units used when the .ADX file was created. Then, use the Save As command, select the Sweep Table (.ADS) file type, and supply a file name which is typically the same as the .AGM, .ADX, and .ADF files.

### **Waveform Files, Stereo Separation**

---

To make your own stereo waveform files that support stereo separation testing, each channel waveform must be made individually. Create two separate .ADX files, each listing the frequencies desired in the specific channel. Save them with names indicating which channel each is intended for; for example, ABCLEFT.ADX and ABCRIGHT.ADX. Run MAKEWAV2 individually for each .ADX file, using the /A capability as described above to force the same sinewave amplitudes in these two files. The waveform files in this example would be automatically named ABCLEFT.AGM and ABCRIGHT.AGM. Then, use the DOS Copy command to concatenate the left and right channel monaural (.AGM) files into a stereo (.AGS) waveform file. The command is COPY /B ABCLEFT.AGM + ABCRIGHT.AGM ABC.AGS. This stereo waveform file will then load automatically into both buffers when selected in the Waveform dialog of FASTTEST, FASTTRIG, or CODEC. Stereo waveform files cannot be used with FFTGEN, which has only one generator buffer.



## Waveforms for FASTTEST

In full-memory units (SYS-222 with MEM or SYS-322), FASTTEST requires an 8192-sample generator record size (/S8192). If the waveform length in this record is also 8192 samples, this is exactly 1/2 the 16384 sample analysis record length and must be used in conjunction with the Maximum choice on the FASTTEST panel. Users can manually choose analysis record lengths of 16384 (selected on the FASTTEST panel as Maximum when full memory is installed in the DSP) or the shorter analysis records of 4096, 2048, 1024, 512, or 256 samples. For Noise mode of FASTTEST to work, the analysis record must be twice the waveform length. If Noise mode is not used, the analysis record may be equal to the waveform length. Thus, for example, an analysis record length selection of 2048 requires a waveform length of 2048 (/L2048 /S8192) if Noise mode is not used and 1024 (/L1024 /S8192) if Noise is to be measured.

An SYS-222 without MEM option requires a 2048-sample generator record size. An SYS-222 without MEM option has a maximum analysis record length of 4096 samples, selected either by Maximum or 4096. The 2048, 1024, 512, and 256 sample analysis record lengths can also be selected. MAKEWAV2 will by default generate an 8192 sample waveform length and record size when neither the /L nor /S option is used.

To generate waveforms for the small-memory SYS-222 units, MAKEWAV2 must be operated with the /S2048 option to generate waveforms with a 2048-sample length. When only the /Snnnn or the /Lnnnn option is used, MAKEWAV2 generates a waveform with both the length and record size set to the specified amount. For example, either MAKEWAV2 /L1024 or MAKEWAV2 /S1024 will have the same effect, producing a waveform length of 1024 samples in a record size of 1024 samples. If the waveform length and record size are to be different, both options must be specified with different numeric arguments for each.

## Waveforms for FASTTRIG, CODEC

FASTTRIG and CODEC only operate with full memory DSP units (SYS-222 with MEM option or SYS-322). Both programs

automatically select an FFT analysis record length twice the downloaded generator signal length. The analysis record length choices automatically selected by FASTTRIG and CODEC are 16384, 8192, 4096, 2048, 1024, 512, or 256. Thus, the only allowable generator record sizes for FASTTRIG and CODEC are 8192, 4096, 2048, 1024, 512, or 256 samples. Either the /S or the /L option with the desired record size will cause MAKEWAV2 to generate the appropriate waveform.

Longer generator waveforms provide better generator frequency resolution. For example, with 8192 samples and the 48 kHz sample rate all signals must be multiples of the 5.859 Hz basic frequency. With 2048 samples and the 48 kHz sample rate, the basic frequency becomes 23.436 Hz. At 512 samples, the basic frequency is 93.75 Hz. Lower resolution means signals cannot be spaced as close together. This limits the ability to gain detailed information about low-frequency response of the system. Lower resolution becomes even more restrictive when additional criteria are established. For example, it may be desired that signals are not generated in the analyzer bins where AC mains hum signals fall (fundamental, second and third harmonics of AC mains frequency) so that FASTTEST-FASTTRIG measurements will be sensitive to hum. If harmonic distortion produced by low frequency sinewaves is to be measured, the bins at lower order harmonics of all generated sinewaves must be kept free from signals. The reduced resolution of shorter generator records is less likely to be a limitation at high frequencies where it represents a smaller and smaller percentage deviation from the desired frequency.

The advantage of shorter record lengths is the shorter required acquisition time and FFT computation time during measurements. A maximum 16384-sample record takes 341.3 milliseconds to acquire at 48 kHz sample rate. A 2048-point analysis record length requires only 42.67 milliseconds to acquire. Computation time of the FFT after acquisition reduces even faster with shorter analysis record lengths. For applications such as a quick check, using only a few tones across the audio spectrum, a shorter sample length and consequently faster acquisition and FFT computation may be appropriate. Required signal burst length with FASTTRIG and CODEC depends on the generator record length as shown in the table below.

## Generator Record Length vs. Attributes (48 kHz sample rate)

<b>Generator Record Length</b>	<b>Minimum Burst Length</b>	<b>Lowest Freq. &amp; Minimum Freq. Spacing</b>
8192	1.050 sec	5.86 Hz
4096	520 ms	11.7 Hz
2048	260 ms	23.4 Hz
1024	135 ms	46.9 Hz
512	70 ms	93.7 Hz
256	40 ms	187.5 Hz



## Quasi-Anechoic Acoustical Tester (MLS)

To load the Quasi-Anechoic Acoustical Tester (MLS) program, display the Digital Analyzer panel. Click on the down arrow at the right end of the “Analyzer” field, at the top of the panel. In the list that appears, select “Quasi-anechoic acoustical tester (mls)”. The program will load from disk, the title bar of the panel will change from “No DSP Program” to “MLS Sequence Analyzer”, and control and display fields will appear.

### Introduction

The Quasi-Anechoic Acoustical Tester (MLS) program for the Digital Analyzer uses Maximum Length Sequence (MLS) testing to characterize the linear response of acoustical and electronic devices. It permits time-selective measurements in which one signal, such as the direct sound from a loudspeaker, may be separated from another similar signal, such as a room reflection. The time window may be adjusted to allow measurement of any arrival in a complex reverberation pattern. These signals may be examined in the time domain (showing “energy” as a function of time) or in the frequency domain (amplitude and phase vs. frequency). Impulse responses may be saved to disk for later down-load to the DSP and further analysis.

Except in repetitive testing with unchanged dimensions between loudspeaker under test, measurement microphone, and reflecting surfaces, use of MLS typically involves both time domain and frequency domain displays. It is normally necessary to examine the time domain impulse response from MLS to determine the exact arrival time of the signal and the first reflection, designation of that time section for FFT spectrum analysis, and finally graphing of the anechoic frequency (and possibly phase) response for examination or comparison to limits.

## Acquisition and Analysis

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Digital signals may be acquired directly from the digital domain via any of System One Dual Domain's digital inputs, or analog signals may be acquired via System One's built-in A/D converters. Signal bandwidth extends to 24 kHz at the 48 kHz sample rate and proportionately less at the 44.1 kHz and 32 kHz sample rates. Only these three sample rates may be used with MLS. Sample rate is selected on the Digital I/O Panel, not on the MLS panel.

## MLS Overview

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This program generates a special digital noise signal called a Maximum Length Sequence (MLS) in the DSP module. For typical loudspeaker measurement applications this digital signal is converted to the analog domain in the D/A converter and fed through the analog generator output stage. The generator typically connects to a power amplifier that drives the loudspeaker under test. A measurement microphone is used to pick up the acoustical signal and return it to the System One analog input. System One performs balanced-to-unbalanced conversion, automatic gain ranging, and drives the DSP's A/D converters. The DSP module and MLS software then perform a cross-correlation between the received and transmitted signals to obtain the impulse response, which is stored into DSP memory. The impulse response may be displayed on the computer, permitting the user to select the portion of impulse response of interest. This portion may be transformed into the frequency domain to study both magnitude and phase response versus frequency.

These properties are of use when measuring loudspeakers or other electroacoustic devices. The time-selective capability permits separating the device-under-test response from that of the room in which the measurements are made. Alternately the room itself may be measured, studying the reflection characteristics of each surface in the room or of the room taken as a whole. Electronic effects devices such as digital delays and reverberators may be measured for both frequency and time domain characteristics. The noise rejection properties of MLS analysis may be valuable when dealing with noisy electronic devices such as communications lines or two-way radio equipment. The analysis technique depends on repeatable time

domain behavior in the device under test and is therefore not suited to measuring tape recorders, pitch shifters, time compressors or other devices with wow and flutter or speed error.


The MLS Sequence Analyzer performs typical loudspeaker production tests very rapidly. Stimulus generation, acquisition, cross-correlation, and fast Fourier transform operations are all performed within the DSP module of System One + DSP or System One Dual Domain and are thus independent of computer speed. On the order of two seconds total is typically required for these functions. Transmission time of the data to the computer for graphing and/or limits comparison is computer-speed-dependent. The operating speed of MLS is identical whether evaluating a short section of the record for anechoic response or longer portions for integrated room response. The signal need not be re-acquired or re-correlated in order to evaluate response under both anechoic and wide-bandwidth conditions. The impulse response remains stored in DSP memory until a new acquisition is made.

A common manufacturing defect in multi-way loudspeaker systems is for one of the drivers to be connected with the opposite polarity to that which the designer intended, creating response dips due to cancellation near the crossover frequencies. It is also common for the external system terminals to have their polarity reversed, producing severe stereo imaging problems. These defects may be detected by applying upper and lower limits to the phase-vs.-frequency data.

MLS offers a selection of four different Maximum Length Sequences, each 32k long. This feature permits up to four different System One-equipped test stations to be located near one another on the production test floor with no interference effects between them.

### Setting Time Spans

The section of the impulse response (time domain) which is to be transformed into the frequency domain is selected by the Start and Stop time values at Source 1 on the Sweep panel last used for a time domain display before Source 1 is changed to MLS Frequency. Before selecting a frequency domain display it is essential that a time domain display be selected and plotted. To limit FFT analysis to only the direct

sound, for example, the Source 1 Start and Stop values must set to the beginning and end of the direct sound arrival. The  key must then be pressed to send this time span information to the DSP, followed by frequency-domain analysis of an already-acquired signal. For a series of frequency response measurements under fixed physical relationships between loudspeaker under test and measurement microphone (typical of production test), data for each new speaker can be acquired and transformed in one operation with Source 1 set to MLS Frequency. The time span used in each acquisition and transform is the last time span that had been set at Source 1.

If the characteristics of the impulse response are not known in advance, the Source 1 Start and Stop values can initially be set to a wide span which includes the whole impulse response. For typical loudspeaker testing applications this might be 0 and 50 ms respectively. The direct sound and the first reflections can be identified, with the beginning of build-up of the first peak in the response determining Start time. By studying the decay of the impulse, the onset of the first reflection can be determined. The Stop value should be set to just before the beginning of the reflected signal. The graph can be zoomed to those points and values copied to the Sweep panel (right mouse button menu) to display only the portion of interest.

### **Time Span vs. Lowest Usable Frequency**


The setting of Source 1 Start and Stop values limits all subsequent frequency domain analysis to the portion of the time record bounded by these values. An impulse response will oscillate or be active for a length of time roughly proportional to the period of its lowest frequency component. The time record must be long enough to include this oscillation to obtain meaningful information about the lowest frequency component. For accurate measurements this oscillation must be allowed to die down for several cycles, requiring a time record approximately three times the period of the lowest frequency of interest. For example, the period of a 200 Hz signal is 5 ms, implying a 15 ms required time span. For loudspeaker measurements this also implies an approximate 5 meter (16 foot) distance between arrival paths of the direct sound and the interfering reflections, which requires an unusually large space for testing.



## Resolution vs. Time Span

The frequency resolution of any FFT spectrum analysis depends upon the record length (number of samples) upon which the FFT was performed; there will be half as many FFT bins as there are amplitude samples in the record transformed. With MLS, actual resolution is further limited by the fact that only a short time span of the acquired signal (before the first reflection arrives) is normally selected for analysis by the Start and Stop times on the Sweep panel while the impulse response is viewed in time domain mode. The remainder of the transform buffer is padded out with zeros. The number of non-zero samples determines actual resolution. That actual resolution is then effectively interpolated, with the interpolation resolution (apparent resolution) being 2.93 Hz with full memory or 11.7 Hz with minimum memory. Typical selected time spans for anechoic measurements in real rooms are on the order of three to six milliseconds. At the 48 kHz sample rate, these time spans may thus include on the order of 150 to 300 data samples and produce actual resolutions on the order of 320 Hz  $\left(\frac{48000}{150}\right)$  to 160 Hz  $\left(\frac{48000}{300}\right)$ .

## Waveform Display vs. Frequency Response

Waveform display and frequency response tests may be loaded without destroying the waveform data in DSP memory if both tests have MLS attached. You may acquire a signal with any one of these tests and then easily switch analysis domains by simply loading another of these tests and pressing . This is generally easier than making the necessary Sweep panel changes to switch between time and frequency domain displays.

## Dual Channel Operation

The DSP hardware and MLS software are both capable of two-channel operation. Two independent A/D converters permit simultaneous acquisition of two analog signals. The digital interfaces are also of two-channel architecture. The input signal selection capability of the analog interface permits assigning a signal to one DSP channel and another related or completely independent signal to the other channel. The digital interface permits selecting the “A” digital

signal channel as one signal and the “B” signal as the other. Both channels will be acquired, processed, and displayed when the **F9** key is pressed or “Go” is clicked. Both will be re-processed and displayed when the **F6** key is pressed.

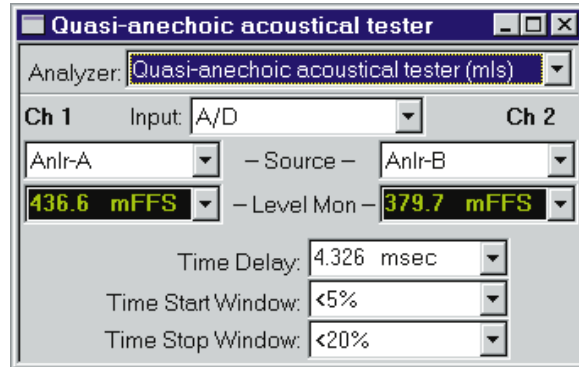


Figure 13-1 MLS Panel, Small Version

It is also possible to acquire a signal into only one channel and then later acquire another signal into the other channel. If one of the two channel input selections is set to NONE, no signal will be acquired into that channel when **F9** is pressed. Conditions or devices may be changed to another desired test condition. Then, the originally-used input channel can be set to NONE and the desired signal source selected at the second channel, where no signal has yet been acquired. A second operation of the **F9** key acquires data into that channel but does not over-write data in the first channel. The data in both channels will then be processed, transmitted to the computer, and displayed.

Signal in both channels will be processed every time **F6** or **F9** is pressed, according to the present settings on the Digital Analyzer (MLS) panel. It is thus not possible to use different windows or different display options for the two channels. Both sets of data must be acquired at the same sample rate. Processing will be done according to the rate selected at the time of pressing **F6**. If the data in one channel had been acquired at a different rate, it will be erroneously transformed or displayed with a resulting frequency error directly proportional to the difference in the two rates.

## Input Selection and Level Monitors

MLS analog vs. digital input source selection, input channel selection, and the function of the Level Monitor Displays are identical to the other FFT programs; see page 10-7.

### Time Delay

The Time Delay field is used to tell the DSP the distance from the speaker under test to the measurement microphone as a reference for the phase measurements. This information allows the DSP to subtract out the transit time delay (propagation time) from the phase readings. As the Time Delay value is adjusted the phase response will slope up or down reflecting the constant time delay component of the data. The initial value of Time Delay

may be estimated from a measurement of the distance between loudspeaker and microphone. The proper final Time Delay value may be determined experimentally as the peak amplitude on a time domain graph or to obtain the smallest slope on phase. The Time Delay field is a numeric entry field that accepts input from the keyboard or by pasting from the Windows clipboard.

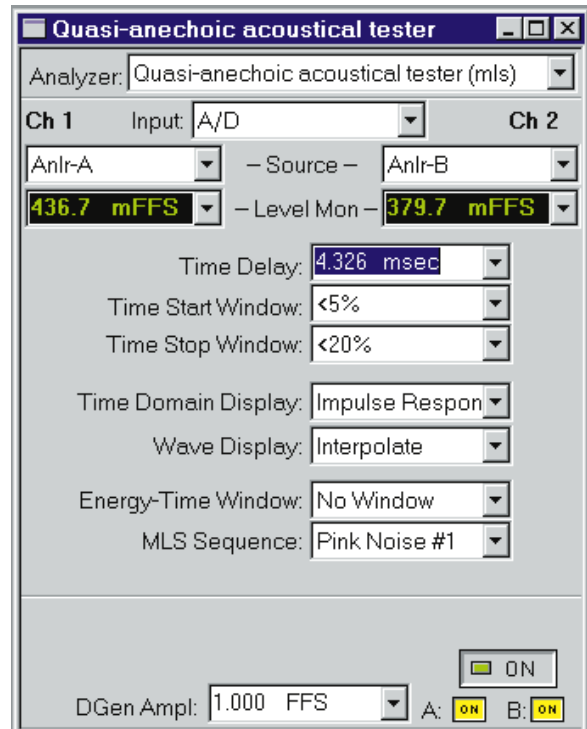


Figure 13-2 MLS Panel, Large Version

## Setting Time Delay Value

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The Time Delay field is used to compensate for the effect of time delay on the measurements. This is of concern only when making phase measurements since a time delay becomes an increasing phase shift with frequency. This field is normally set to a value of the exact peak of the impulse. In practice, the Time Delay is “fine tuned” by examining the phase response and adjusting the value for the flattest curve.

## Time Windows

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When a section of the impulse response (direct arrival signal before reflections, for example) is isolated and transformed into the frequency domain, the impulse amplitude at the beginning and ending of that section will generally not be exactly the same and thus will not “splice” smoothly. The sharp edges introduced into the impulse response by splicing unequal amplitudes will produce ripples in the resulting frequency response plot. Windowing the time domain data by attenuating the amplitude at the beginning and end of the section to be transformed will reduce this rippling, but also reduces the steepness of transitions in the frequency response plots. The “Time Start Window” and “Time Stop Window” fields select the window applied to the impulse response (time domain) when transforming it to the frequency domain.

The time window is made up of two “half-windows”. The first half is selected in the “Time Start Window” field and is used to process the first portion of data, beginning at the Source 1 Start time on the Sweep panel. The second “half-window” is selected in the “Time Stop Window” field and processes the later portion of data, ending at the selected Stop time on the Sweep panel. Separate selection of the Source 1 Start and Stop “half-windows” permits creation of asymmetrical windows, which provide the optimum match to the asymmetrical shape of the typical impulse response. To change selections, click on the down arrow at the right of the field and click on the desired selection in the list which is displayed. The available selections at both the Time Start Window and Time Stop Window fields are a family of half-cycle raised cosine functions labeled NONE,

<5%, <10%, <20% and <30%. The numeric value refers to the amount of the data record (time span multiplied by sample period) taken up by the window's transition from zero to full amplitude. The Time Start Window "half-window" starts with an amplitude of zero at the Sweep panel Start time and climbs to an amplitude of 1.00 (no attenuation) at or before the selected percentage of the record. The Time Stop Window "half-window" starts with an amplitude of 1.00 at or following a point during the record which is within the selected percentage of the record end, and falls to zero at the Sweep panel Stop time. The windows with a steeper transition will alter the data less but will also have less impact on the frequency response ripples. The more gradual transitions have greater ripple reduction but alter the data more. See the Advanced Theory topic for more details and illustrations of the available choices.

## Impulse Response vs. Energy-Time

The Time Domain Display field selects between Impulse Response and Energy-Time curve displays. To change display modes, click on the down arrow at the right of the field and click on either Impulse Response or Energy-Time.

The Impulse Response selection will show the results of the MLS correlation, which is the actual impulse response of the device under test.

The Energy-Time selection will display what is commonly called an "energy-time curve". The energy-time curve computation process involves transforming the impulse response to the frequency domain, doing further processing in the frequency domain, and transforming the result back to the time domain.

A frequency window may be used for the conversion from frequency domain back to time domain. The frequency window is selected in the "Energy-Time Window" field. Click on the down arrow at the right of the field to display the list of selections. The choices are No window, half Hann, Hann, <240 >8kHz, and <120 >16kHz.

The No Window selection will perform the required transformations with all frequency components of the signal included in the computations.

The Hann selection reduces both high and low frequency energy, concentrating on arrivals at the center of the frequency range. Since the processing occurs on a linear frequency scale, this will focus analysis on signals around one quarter of the sample rate. At 48 kHz this will result in the 12 kHz energy dominating the energy-time display. This selection is not fundamentally useful for most applications, but is included for correlation to measurements by other manufacturer's equipment where this window is used.

The Half Hann selection reduces the contribution of high frequencies. The low frequency information remains unchanged. When operating at the 48 kHz sample rate this window filters out energy above 12 kHz.

The <240 >8kHz window filters out energy below 240 Hz and above 8 kHz, producing equal sensitivity to signals over a 5 octave range.

The <120 >16kHz window spreads the analysis over a 7 octave range.

## Energy-Time Curve Concepts

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The Energy-Time selection causes the DSP to transform the impulse response into the frequency domain, apply a frequency-domain window and Hilbert Transform to the complex frequency response, and then inverse transform the data back to the time domain to produce a plot of the estimated arrival of energy versus time. This "energy-time curve" is an approximation to the actual energy arriving at the microphone, since this energy can never be known without the simultaneous measurement of the velocity and pressure of the sound. The term "energy-time response" is actually a misnomer since a true computation of energy requires knowledge of both kinetic and potential energy and a microphone signal can only supply one of these. A more accurate term sometimes used in technical papers is the analytic signal magnitude. However, the term energy-time has become common usage and, to avoid confusion, will

be used here. Energy-Time curves may also be considered as similar to the envelope of the impulse response. The DSP computes an estimate of the energy by applying a Hilbert transform to the frequency domain data before transforming back to the time domain. The resulting trace will not show the negative excursions of the impulse response. This display is useful for determining arrival times and relative energy distribution in time.

Since the energy-time graph is computed with transforms, a frequency window must be applied to the data to prevent alias behavior. The frequency window to be used is selected in the “Window for Energy-Time curve” field. The choices are No window, half Hann, Hann, <240 >8kHz, and <120 >16kHz. These windows operate in the frequency domain. The No window selection will perform the required transformations with all frequency components of the signal included in the computations. The deviations from a flat frequency response create ripples in the time domain energy response. The Hann selection is the one window found on software from other manufacturers. This reduces both high and low frequency energy, concentrating on arrivals at the center of the frequency range. Since the processing occurs on a linear frequency scale, this will focus analysis on signals around one quarter of the sample rate. At 48 kHz this will result in the 12 kHz energy dominating the energy-time display. This produces very attractive displays, which are very wrong.

The half Hann selection is a window suggested by Lipshitz and Vanderkooy, which only reduces the contribution of high frequencies. The low frequency information remains unchanged. When operating at the 48 kHz sample rate this window filters out energy above 12 kHz. Audio Precision developed the remaining two windows for even more accurate measurements of typical audio signals. The <240 >8kHz window filters energy below 240 Hz and above 8 kHz, producing equal sensitivity to signals over a 5 octave range. The <120 >16kHz window spreads the analysis over a 7 octave range. Both windows produce much more accurate results than the Hann window with only minor increases in alias behavior.

For an excellent technical discussion of the effect of windows on energy-time-curve computation see Lipshitz and Vanderkooy, “Uses

and Abuses of the Energy-Time Curve”, Journal of the AES, Vol. 38, No. 11, November 1990, pp. 819-836.

## Display Processing

The “Display Processing” field permits selection of the display mode for all graphing of results in time or frequency domain. To change modes, click on the down arrow at the right of the field and click on the desired selection in the list which is displayed. The selections are Interpolate, Normal, and Peak.

If “Interpolate” is selected the DSP will compute the data value, interpolated from the nearby measured values. This smoothes out the “stair-step” appearance of frequency response curves at low frequencies with a Log horizontal axis, where the bin width (usually 2.93 Hz at the 48 kHz sample rate) occupies a significant portion of the screen.

If “Normal” is selected the DSP will return the closest actual measured value without altering the data. Normal is the recommended display mode for frequency response data with a Linear horizontal axis or with a Log axis above 100 to 300 Hz. In these cases, the jagged lines caused by the FFT bin width are not usually noticeable.

The “Peak” mode will return the largest value between the last requested sweep point and the current one. Peak is recommended for time domain MLS displays (Impulse Response and Energy-Time). Peak mode would not normally be used for frequency response displays with MLS.AZ1, since high values are of no more interest than low values when plotting frequency response.

## Save and Load MLS Waveforms

Saving and loading waveforms from and to the MLS program is possible, but with some detailed differences from the same feature with the other FFT-based programs. See the Save and Load Waveform section in the Spectrum Analyzer chapter for the general discussion, with only the variances treated here.



- MLS effectively places the impulse response resulting from cross-correlation of the transmitted and received signals into the Acquisition Buffer of the DSP. Therefore, the waveform that is saved to disk for later use is the impulse response of the device under test, not the original “raw” unprocessed samples.

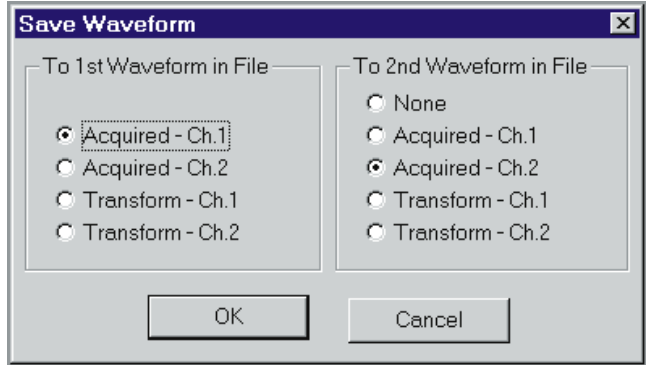


Figure 13-3 Save Waveform Dialog Box

- the option of saving only a shorter section of the impulse response does not exist. Only the entire length of the impulse response may be saved to disk.

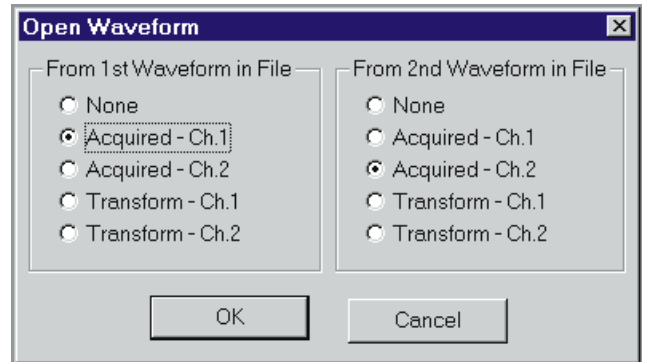


Figure 13-4 Open Waveform Dialog Box

## MLS Sequence Control

The MLS Sequence field selects from among four different MLS sequences to avoid interference when several acoustic test stations are operating near one another. The four pink-noise-filtered 32k point maximum length sequences are numbered Pink Noise #1 through Pink Noise #4. Each will cross-correlate to approximately -45 dB

against any of the other three. These four sequences are all weighted with a pink noise filter to increase their low frequency energy and provide a constant power per octave across the audio band. This greatly improves the signal-to-noise ratio at low frequencies, increasing measurement accuracy in typical room ambient noise conditions. A single white noise sequence is also provided, labeled White Noise #1, for unusual applications where the large high frequency energy level may be desired and signal-to-noise ratio is not a concern. If the sequence is recorded on RDAT or other digital tape for later measurement, it is important that the same sequence number be selected on playback. Otherwise no impulse response will be obtained.

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## Generator Amplitude

The DGen Ampl field is used to set the MLS generator output amplitude when it is not routed through the analog generator outputs, but instead to the digital outputs or the BNC connector D/A output. DGen Ampl is a numeric entry field that accepts data from the keyboard or may be pasted into from the Windows clipboard. All level adjustment is then performed with the Analog Generator Amplitude setting.

The generator operates continuously and normally during panel mode and signal acquisition. It is, however, shut down during correlation computation, FFT computation, and waveform saving or loading operations. Off/On and Channel Select with MLS is identical to the other FFT-based programs containing generators; see page 11-26.

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## Advanced MLS Theory

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### Impulse Response of Linear Systems

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Any linear device or system may be completely characterized by knowing its impulse response. The impulse response is the output that results when the device under test is stimulated with an infinitely narrow pulse of infinite amplitude. In practice, if the stimulus pulse width is short compared to the length of the impulse response the measurements will still be accurate. This duration is typically 10

microseconds for audio frequency work. However, in spite of the short impulse duration, data must be acquired for the full length of time occupied by the impulse response of the device under test. This time is dictated by the lowest frequency desired in the measurement. An additional limitation which is usually more stringent is that the reverberation time of the space in which the measurements are performed must allow the signal to adequately die out before another impulse may be generated.

The amplitude of the pulse will directly affect the amplitude of the output response. Since the peak handling capacity of the device under test limits the allowable impulse amplitude, the signal-to-noise ratio of impulse measurements is quite poor. For measurements on electronic systems this is typically not a problem. In loudspeaker or acoustic measurements the poor signal-to-noise of impulse measurements can be a severe limitation.

The impulse amplitude is limited by the peak signal-handling capability, but the energy in the test is represented by the RMS signal value. The energy directly sets the signal-to-noise ratio for any given background noise level. A single impulse has infinitesimal energy content, and so has very poor signal-to-noise ratio. Because of this, most engineers working with impulse test signals average the results of many impulses to improve signal-to-noise ratio. However, the impulse repetition rate is limited by the required acquisition time and reverberation time. If the impulses are coherently averaged and the noise is not synchronous with the impulse repetition rate, the impulses will reinforce each other and the interfering noise will tend to zero.

### **Impulse Response from Pseudo-Random Noise**

Pseudo-random noise can be viewed as a random sequence of impulses, some positive and some negative. This sequence repeats at a specific rate called the repetition rate of the noise. If the device under test is linear, the response to the pseudo-random noise will be the sum of the responses to the individual impulses. When these impulses arrive to be measured, the DSP effectively shifts each one in time to align them at the same point and averages them together. As long as the interfering noise is asynchronous to the signal, the noise will average out toward zero. The averaging operation creates a single

impulse response that has lower noise than any of the individual impulse responses. The signal-to-noise ratio improvement is proportional to the square root of the number of impulse responses averaged. For the 32767-point pseudo-random sequence used in the DSP, the noise improvement relative to a single impulse is 181 times or 45 dB.

In the frequency domain, pseudo-random noise has a flat spectrum, with components spaced at the repetition rate of the noise. For a 32,767 point sequence operating at a 48 kHz sample rate the pseudo-random sequence will repeat every 0.68 seconds, or a rate of 1.46 Hz. This is called white noise and has equal energy per unit of bandwidth when analyzed on a linear frequency scale. The ear hears on a logarithmic scale (in fractions of an octave) and the spectrum of most interfering noise is also flat on a logarithmic frequency scale. Consequently, white noise produces more energy than is necessary at high frequencies and, conversely, less energy at low frequencies than is desirable. For example, in the octave band from 40 Hz to 80 Hz there will be 27 frequency components of the noise. In the octave band from 10 kHz to 20 kHz there will be 6849 components, 254 times as many or a power level about 24 dB higher. To compensate for this effect, the DSP software filters the test signal with a pinking filter that attenuates the higher frequency components in direct proportion to their number. The result is an equal power in each octave band, providing a more constant signal-to-noise ratio across the measurement frequency range.

## **Hadamard Transforms**

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The cross-correlation operation required to shift the individual impulses in time and average them together is accomplished with a Fast Hadamard Transform.

For a description of the Fast Hadamard Transform and its application to MLS testing see Borish and Angell, "An Efficient Algorithm for Measuring the Impulse Response Using Pseudo-Random Noise", Journal of the AES, Vol. 31, No. 7, July/August 1983, pp. 478-488.

For a description of a technique to simplify computation of the Hadamard Transform see Borish, “Self-Contained Crosscorrelation Program for Maximum Length Sequences”, Journal of the AES, Vol. 33, No. 11, November 1985, pp. 888-891.

## Frequency Response From Impulse Response

The impulse response is a time domain expression of the device behavior but most people find a frequency domain expression more useful. To convert the time domain display to a frequency domain display it is merely necessary to transform the impulse response into the frequency domain using a Fourier transform. The result of a Fourier Transform on a perfect impulse is a flat spectrum in the frequency domain. If the device under test modifies the impulse, its Fourier transform will be a display of the frequency response.

## Time Windows for Time-to-Frequency Transforms

The FFT algorithm is used to transform a segment of the impulse response into the frequency domain in order to see the frequency response. This segment is selected from the original impulse response by setting to zero all data outside the region of interest. If signal in the data record being transformed does not naturally decay to zero at the beginning and end of the segment, there will be sharp discontinuities introduced by the selection of this segment for transforming. These discontinuities in the waveform produce large amounts of high frequency energy in the transformed result. This high frequency energy results in ripples on the displayed frequency response curve.

To alleviate this problem, a “window” may be applied to the data. The idea behind a window is to gradually taper the data at both ends of the record toward zero so that it will always make a smooth transition with the following and preceding repetitions of the record. This is accomplished by multiplying each point in the data record by a mathematical function that is near unity (1.000) in the center of the data record and small at the ends of the record. The simplest such function is an inverted cosine wave raised above zero with an added DC offset so that its negative peaks just reach zero. After multiplication by the window function the data record goes to zero at the ends and so smoothly meets each data record on either side of the one being transformed. Multiplying the data by the window function does alter

the spectrum of the original signal. As might be expected by visualizing the envelope of the repeating windowed data record, the frequency response will be smoothed. However, the spurious high frequency components produced by the sharp discontinuities will have been eliminated. The raised cosine window described is called the HANN window after its inventor, Austrian meteorologist Julius von Hann. (It is often incorrectly called a Hanning window due to confusion with the “Hamming” window, named after its inventor Richard Hamming.)

The generic term window comes about because it restricts the view of the FFT to the central portion of the data record in much the same way that a window restricts the view of a person looking through it. There have been an endless variety of windows developed which trade off the spread in the spectral peak versus the ultimate attenuation of the spurious energy created by the ends of the data record. The sharper the roll-off in the skirts, the wider the peak must be in the passband. The bandwidth of the peak will be a specific number of bins for any given window. Increasing the length of the data record will reduce the bin width and therefore reduce the bandwidth in Hertz.

Transformation of impulse responses of loudspeakers is a special case, since the typical impulse has a fast rise and slow decay. Thus, it is desirable to use an asymmetrical window function which also has a fast rise and slow decay in order to taper values at the two ends of the selected portion to zero with minimal effect on the important information in the impulse. The window selection of MLS.AZ1 is therefore made up of two sections, each with its own panel selection field. The left-hand field on the TIME WNDW line selects among four raised half-cosine choices, each corresponding to the first 180 degrees of a cosine and thus rising from zero at the beginning of the selected portion of record to a value of 1.00 later in the record. The right-hand field on the same line selects among four raised half-cosine choices, each corresponding to the last 180 degrees of a cosine and thus falling from a value of 1.00 late in the selected portion to zero at the end of the record. The nominal time selections (<5%, <10%, <20%, and <30%) refer to the percentage of the selected portion of the impulse across which the half-window makes its full transition. The “less than” sign ( indicates that the actual percentage may be less than specified, since the actual number of samples for the transition from zero to unity

(or vice-versa) will always be an exact binary power such as 4, 8, 16, etc. The DSP therefore rounds down from the selected value to the largest exact binary power within that value.

For example, assume a START time of 4 ms and a STOP time of 8 ms, resulting in a 4 ms time span. If the 48 kHz sample rate (20.8 microsecond sample period) is in use, there will be approximately 192 samples in the selected span. A START selection of <5% would nominally make the transition in the first 5% of the 192 samples, or 9.6 samples; the actual transition time will be rounded down to 8 samples as an exact binary power. Similarly, if <30% is chosen for the STOP selection, the nominal transition would take place during the final 30% of the 192 samples, or 57.6 samples. The actual transition time will be rounded down to 32 samples, the next lower exact binary power. All the data between the 8th sample from the beginning and the 32nd sample from the end will be unattenuated.

For an excellent technical discussion of windows and their characteristics see F. J. Harris, "On the use of Windows for Harmonic Analysis with the Discrete Fourier Transform", Proc. of the IEEE, Vol. 66, No. 1, Jan 1978, pp. 51-83.

## **Frequency Windows for Energy-Time Displays**

The Energy-Time selection causes the DSP to transform the impulse response into the frequency domain, apply a frequency-domain window and Hilbert Transform to the complex frequency response, and then inverse transform the data back to the time domain to produce a plot of the estimated arrival of energy versus time. This "energy-time curve" is an approximation to the actual energy arriving at the microphone, since this energy can never be known without the simultaneous measurement of the velocity and pressure of the sound. Since the Energy-Time curve is computed from the pressure response alone it can never be complete.

The Time Start Window field selects the window function applied to the frequency-domain data when computing energy-time curves. The available window choices are No Window, Half Hann, Hann, <240 >8kHz, and <120 >16kHz. The Hann window, although used on other measurement equipment, yields inaccurate results because it

excludes behavior of the majority of most device's frequency range. The Half Hann, <240 >8kHz, and <120 >16kHz are preferable for most applications.

The first selection turns off windowing, allowing use of all frequency components of the arriving sound in the energy-time computation. The Half Hann selection uses a raised half-cycle cosine window which does not attenuate low frequencies but attenuates high frequencies above one quarter of the sample rate (approximately 12 kHz). The Hann selection uses a standard raised cosine window as provided on measurement equipment from other manufacturers. It centers its analysis on the energy around one quarter of the sample rate (approximately 12 kHz) attenuating both low and high frequencies. This yields inaccurate results on most devices because it excludes the behavior of the vast majority of the device's frequency range.

The last two window selections have been developed by Audio Precision to reduce the inaccuracies of the Hann window at both low and high frequencies while still reducing the window ripples which would occur if no window was used. These windows gradually roll off the spectral data above and below the frequency values specified in their labels. The <240 >8kHz window focuses the analysis between 240 Hz and 8 kHz by applying asymmetrical half-cycle raised cosine windows which attenuate below 240 Hz and above 8 kHz. The <120 >16kHz window shifts these frequencies by an additional octave to improve measurements of wideband devices at the expense of slightly more window rippling.

For an excellent technical discussion of the effect of windows on energy-time-curve computation see Lipshitz and Vanderkooy, "Uses and Abuses of the Energy-Time Curve", Journal of the AES, Vol. 38, No. 11, November 1990, pp. 819-836.



## Measurement Units

A number of different units of measurement are common in audio applications. For example, audio signal amplitude is frequently expressed in Volts, dBu, dBm, dBV, or dB relative to an arbitrary reference such as amplitude at mid-band or amplitude with normal signal level. Other units, although not common, can be very convenient and can reduce the data analysis and reporting tasks. Examples include amplitude in dBg (dB relative to the present value of generator output amplitude) or frequency expressed in octaves or decades with respect to a user-settable reference frequency.

APWIN software provides a wide range of selectable units for most measurements. Clicking on the down arrow at the right of a numeric display causes a list of the available units for that meter to appear. The user can then click on the desired unit. While the list is visible, the equivalent numeric values for each of the available units is displayed along with the unit. These values all represent the last measurement that that meter had displayed before the list was displayed, and the meter will not update while the list is displayed. The units displayed at the real-time numeric display of a particular meter on the instrument panel are independent from the units used to display that same meter in bargraph form or when the meter measurements are plotted on a graph. For example, a meter could be set for Volts units on the analyzer panel, dBm units in a bargraph, and dBu or dBr (relative dB) on an X-Y graph when sweeping, or several different bargraphs could all display the same meter measurement but each using a different unit.

### AC Amplitude units

The available analog domain ac amplitude units (although not all are available at all settings and meters) are:

**V** (Volts, Analyzer only). The analyzer meters that measure level or amplitude have the Volts unit available. The Level meters always use true RMS detectors. The main (Reading) meter offers a selection

of detectors including RMS, average, quasi-peak, and peak, so the value actually displayed depends upon the signal waveform and which detector is in use.

**Vrms** (Volts RMS, Generator only). The generator output may be set in Volts RMS. The calibration will be accurate only when a sine waveform is being transmitted. Other waveforms will be set to the amplitude that produces the same peak-to-peak value as a sinewave of the value set. Vrms is an open-circuit value and the actual terminal voltage will be less if a finite load resistance is connected.

**Vp** (Volts peak, Generator only). The generator output may be set in Volts peak. This calibration assumes a sine waveform is being transmitted, and is simply scaled to the Vrms value by the square root of two (1.414 . . .). Since non-sinusoidal waveforms are calibrated to have the same peak-to-peak value as a sinewave, the Vp unit should be correct for most non-sinusoidal waveforms. Vp is an open-circuit value and the actual terminal voltage will be less if a finite load resistance is connected.

**Vpp** (Volts peak-to-peak, Generator only). The generator output may be set in Volts peak-to-peak. This calibration assumes a sine waveform is being transmitted, and is simply scaled to the Vrms value by two times the square root of two (2.828 . . .). Since non-sinusoidal waveforms are calibrated to have the same peak-to-peak value as a sinewave, the Vpp unit should be correct for most non-sinusoidal waveforms. Vpp is an open-circuit value and the actual terminal voltage will be less if a finite load resistance is connected.

**W** (Watts, Generator or Analyzer). The Watts (power) unit is calculated based on the “V squared over R” relationship. In the Generator, the voltage division across the resistive divider made up of the generator source resistance and the resistance of the load must also be involved in the computation. Therefore, the resistance of the load must be accurately known and entered into the Generator Reference Watts field (on the large form of the generator panel). APWIN software will use that value and the selected value of generator output resistance to compute the voltage division ratio, and will then compute and set the required generator open-circuit voltage (emf) behind the output impedance to cause the set value of power to be dissipated in the load.



In the Analyzer, APWIN uses the Analyzer Reference Watts field value for  $R$ , measures  $V$ , and computes and displays power in Watts. The resistance value entered into the Analyzer Reference Watts field must be the net parallel resistance of all external and internal terminations. Since Watts is most commonly used in connection with loudspeaker impedance values that are typically under 16 Ohms, the analyzer input resistance of 100,000 Ohms may be neglected.

**dBm** (decibels relative to one milliwatt, Generator or Analyzer). The dBm is a power unit since its reference is stated in power terms (one milliwatt). It is calculated based on the  $V$  squared over  $R$  relationship in the load impedance. For Generator dBm units, the voltage division across the resistive divider made up of the Generator source resistance and the external load must also be known. Therefore, the resistance of the load must be accurately known and entered into the Generator Reference dBm field. APWIN software will use the Generator panel dBm Reference value and the selected value of generator output resistance to compute the voltage division ratio, and will then compute and set the required generator open-circuit voltage (emf) behind the output impedance to cause the set value of power to be dissipated in the load. As an Analyzer unit, APWIN measures and squares the voltage, divides it by the Analyzer panel dBm Reference value, and converts to decibels for display. The dBm Reference value entered must be the net parallel equivalent of all resistances involved including external circuitry and internal terminations selected in the Analyzer.

**dBu** (decibels relative to 0.7746 Volts, Generator or Analyzer). The reference value 0.7746 is the voltage across a 600 Ohm resistor when one milliwatt is being dissipated in that resistor. Thus, a dBu value will be numerically equal to a dBm value when the circuit or load resistance is 600 Ohms. When used for generator output, dBu is an open-circuit value and the actual terminal voltage will be less if a finite load resistance is connected.

**dBg** (decibels relative to generator, Analyzer only). The presently-programmed value of the generator output is the reference for dBg. This unit is particularly convenient during amplitude sweeps for detecting the threshold of compression of a processor or the clipping point of any device under test. dBg is computed with

reference to the amplitude command value to the generator and measurements in dBg therefore are subject to generator load resistance changes by the operator in addition to gain or input impedance changes in the device under test.

**dBr** (relative decibels, Generator or Analyzer). The reference value for dBr may be typed into the Reference dBr field (generators and analyzers have separate and independent Reference dBr fields), or the present value of measurement or generator output may be copied into the Reference dBr field by pressing the  function key (or selecting the Edit Set Analyzer dBr Reference menu command) for the analyzer or the  function key (or selecting the Edit Set Generator dBr Reference menu command) for the generator. As a generator unit, dBr is an open-circuit value and the actual terminal voltage will be less if a finite load resistance is connected.

**dBV** (decibels relative to 1.000 Volt, Generator or Analyzer). As a generator unit, dBV is an open-circuit value and the actual terminal voltage will be less if a finite load resistance is connected.

## **dBm vs. dBu Concepts**

The dBm unit (decibels relative to one milliwatt) is one of the oldest amplitude measurement units in professional audio and broadcast fields. Since a milliwatt is a unit of power, dBm is also a power unit. When the dBm unit came into common usage, audio equipment was normally transformer-coupled at both input and output and circuits were normally operated in a matched condition (output impedance of the driving device equals the input impedance of the driven device) for maximum power transfer.

However, audio measurement instruments today and through all the history of audio measurements are basically voltmeters, not power meters. Measuring power requires knowledge of both voltage and current, or both voltage and resistance, or both resistance and current. Since audio meters only “know” voltage, their utility in measuring in dBm units depends on them also “knowing” either the circuit resistance (impedance) or current. Older audio voltmeters almost invariably were calibrated based on the assumption that they would be

measuring voltage across a 600 Ohm circuit. One milliwatt dissipated in a 600 Ohm resistor produces a voltage drop of 0.774597 Volts. Thus, older meters with the dBm unit displayed were actually measuring voltage relative to 0.774597 Volts and were in error if connected across any impedance value other than 600 Ohms.

Modern professional audio and broadcast equipment is rarely operated in matched mode for power transfer. Instead, output stages almost always have very low impedance outputs, ranging from near zero up through a few tens of Ohms. Inputs of audio devices are nearly always “bridging”, meaning high impedance relative to the driving source. Common input impedances are 10 kilohms or more for most line input stages and perhaps 1500 Ohms for microphone inputs. Thus, the entire open-circuit voltage developed by the driving device is applied across the input, which essentially does not load it down.

The dBu unit has been in common use, especially in Europe, for more than a decade. dBu is decibels relative to 0.774597 Volts, and says nothing about power, impedance/resistance, or current. If the circuit impedance is 600 Ohms, dBu and dBm are numerically equal. If the impedance is not known, dBu still makes an accurate statement about the voltage.

System One and Two provide accurate dBm unit calibration for both generator outputs and analyzer inputs for any arbitrary circuit impedances, since the user can specify to APWIN software what the circuit impedance value is. For most modern applications, however, the dBu unit is to be preferred. In many cases, in fact, equipment manuals call for level setting in dBm units at values which are unachievable from that equipment. Clearly the equipment designers or manual writers were thinking in dBu even though the dBm term was used.

## Ratio Units

% the % (percent) unit expresses the ratio of two numbers, multiplied by 100. Thus, a ratio of 0.7 is expressed as 70.0%

**dB** the dB (decibel) unit is computed from the equation  $\text{dB} = 20 \log_{10} V$  measurement over  $V$  reference

**PPM** the PPM (parts per million) expresses the ratio of two numbers, multiplied by 1,000,000. Thus, a ratio of 0.00005 is expressed as 50 parts per million.

**X/Y** the X/Y unit is the simple ratio between two numbers

## Frequency Units

**Hz** Hz (Hertz) is the basic frequency unit, expressing the number of total cycles of signal in one second

**F/R** F/R is a frequency ratio unit, obtained by dividing the measured frequency  $F$  by the reference frequency  $R$

**dHz** (delta Hz) dHz is the difference in frequency between the measured frequency and the reference frequency ( $F - R$ )

**%Hz** %Hz is obtained by dividing the measured frequency by the reference frequency and multiplying the result by 100 ( $100 * F/R$ ). For example, a measured frequency of 950 Hz would be expressed as 95.0% of a reference frequency of 1 kHz

**cent** the cent unit is 1/100 of a musical half-tone interval, which in turn is the 12th root of 2

**octs** (octaves) an octave is a frequency ratio of 2:1. For example, with a reference frequency of 1 kHz, a 2 kHz tone is expressed as +1.0 octave and a 500 Hz tone is expressed as -1.0 octave

**decs** (decades) a decade is a frequency ratio of 10:1. For example, with a reference frequency of 500 Hz, a 5 kHz tone is expressed as +1.0 decade and a tone of 50 Hz is -1.0 octave

**d%** (delta %) d% is 100 times the ratio of the difference between the measured frequency and the reference frequency to the reference frequency ( $100 * (F - R)/R$ ). For example, with a reference frequency of 1 kHz and a measured frequency of 950 Hz, the d% value is -5%.

**dPPM** the PPM (parts per million) unit is 1,000,000 times the ratio of the difference between the measured frequency and the reference frequency to the reference frequency ( $1000000 * (F - R)/R$ ). For example, a measured frequency of 1001 Hz with a reference frequency 1 kHz is 1,000 PPM.

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## Phase Units

**deg** the degree is the basic unit of phase measurement, representing 1/360 of a complete revolution of phase. See the Analog Analyzer chapter for a discussion of the phase meter ranges including Auto mode which “unwraps” phase plots.

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## Digital Domain Amplitude Units

All four of the digital domain amplitude units refer to digital full scale (FS), a precisely defined quantity for digital audio signals. For 24-bit resolution numbers, positive digital full-scale is defined as 7FFFFFF Hex, negative full scale is 800000 Hex, and the zero signal baseline is 000000 Hex

Since digital domain amplitude calibration has been defined by the Audio Engineering Society in terms of the RMS value of a sinewave whose peaks just reach those maximum digital codes, it is possible to obtain readings greater than 100% FS (or 1.000 FFS or 0.00 dBFS or 0.00 bits) when measuring high-level signals whose crest factor is less than the crest factor of a sinewave. For example, since a sinewave has a crest factor of 3.01 dB and a squarewave crest factor is 0.0 dB, a squarewave touching the maximum digital codes will be displayed as 141.4% FS (1.414 FFS, +3.01 dBFS, or +0.5 bits).

**FFS** the FFS unit is fraction of full scale. At lower levels common in digital audio applications, they may be displayed as mFFS (milli fractions of full scale, or FFS/1000) and  $\mu$ FFS (micro fractions of full scale, or FFS/1000000)

**%FS** the %FS (percent of full scale) unit is FFS multiplied by 100. For example, a signal at 0.8 FFS is 80%FS

**dBFS** the dBFS unit is defined as  $20 \log_{10} * \text{FFS}$ . For example, a signal at 0.1 FFS is -20 dBFS

**bits** the bits unit is computed from dBFS and the relationship  $\text{bits} = 1.76 + (\text{dB}/6.02)$ . For example, a signal at 0.0001 FFS (-80 dBFS) is -12.99 bits.

## Time Units

**sec** seconds are the basic time unit. At values common in audio, the display may be ms (milliseconds) or  $\mu\text{s}$  (microseconds)

**UI** (Unit Interval) the Unit Interval may be used in timing measurements on the AES/EBU or consumer digital interface signal. A Unit Interval is 1/128 of the frame length (sample period). Thus, at a 48 kHz sample rate a Unit Interval is 162.760 nanoseconds and at a 44.1 kHz sample rate a Unit Interval is 177.154 nanoseconds.

## DCX-127 Amplitude Units

**Vdc** Volts is the basic unit of DC (direct current) electro-motive force

**f(V)** f(V) is computed from the basic Volts measurement and a scaling and an offset value. The equation is  $f(V) = \text{scale} * (V + \text{offset})$

## Resistance Units

**Ohms** Ohms is the basic unit of electrical resistance

**f(O)** f(o) is computed from the basic Ohms measurement and a scaling and an offset value. The equation is  $f(O) = \text{scale} * (\text{Ohms} + \text{offset})$

## DCX-127 Digital Input and Output Units

**dec** the dec (decimal) unit converts the DCX-127 digital input and output between binary and decimal representation



**hex** the hex (hexadecimal) unit converts the DCX-127 digital input and output between binary and hexadecimal

**octal** the octal unit converts the DCX-127 digital input and output between binary and octal

**g(x)** (input) the g(x) unit is computed from  $g(x) = \text{scale} * x$

**h(x)** (output) the h(x) unit is computed from  $h(x) = x/\text{scale}$



## Sweeps and Sweep Setting

The Sweep panel and the Graph are closely related. The small version of the Sweep panel (which is the same size as the large version of the generator and analyzer panels) is used to select the parameter to be swept which is typically an instrument **Setting**, such as generator frequency or generator amplitude. However, the “swept” parameter may be a **Reading** of some input parameter from the device under test, such as the frequency of a

sequence of tones from a test tape, test CD, or remotely located generator. Using a Reading to control the process of measurements is called an **External Sweep**. Chart recorder-style measurements where one or more parameters on the Y axis are plotted versus time on the X axis are a specific form of External Sweeps called **Time Sweeps**. Sweeps of an internal setting are typically established by the user specifying start and stop values, number of steps, log/lin relationship, and permitting APWIN software to automatically compute the value of each intermediate step. It is also possible to use **Table Sweep** mode, in which each sweep setting value is looked up in a user-defined sweep table.

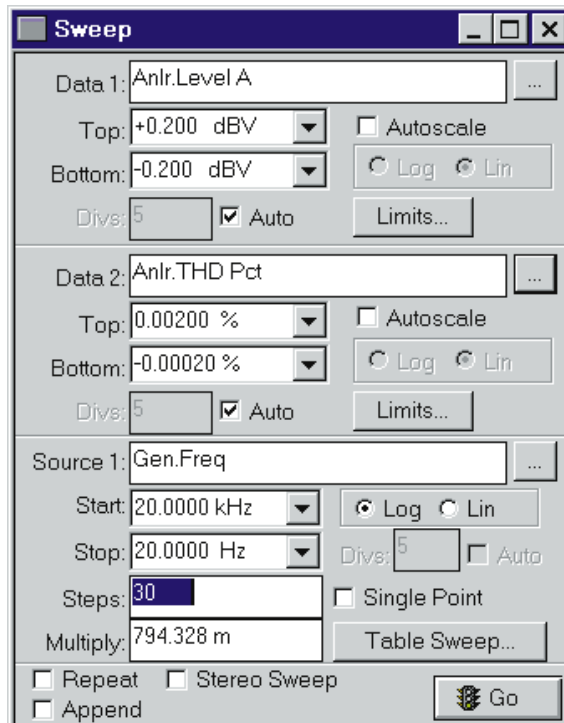


Figure 15-1 Sweep Panel, Small Version

The small version of the Sweep panel also permits selection of one or two parameters to be plotted, selection of auto-scaling mode, and control of characteristics of the graph such as horizontal and vertical scales, logarithmic vs. linear relationship, and number of divisions. The sweep panel also permits “attaching” upper and lower limits of acceptable performance to a sweep test. The large version of the Sweep panel permits adding up to four more traces, plotting measured data instead of the swept independent variable on the X axis, and setting up “nested sweeps”, where two parameters are swept in a coordinated fashion. The Graph displays all this data and provides methods of zooming in, returning to previous zoom levels, optimizing the graph scales to best fit the data, etc.

To bring the Sweep panel to the screen:

Mouse: click on the Sweep icon  on the toolbar, or select Panels Sweep via the menu bar

Keyboard: press  (for Panels Sweep).

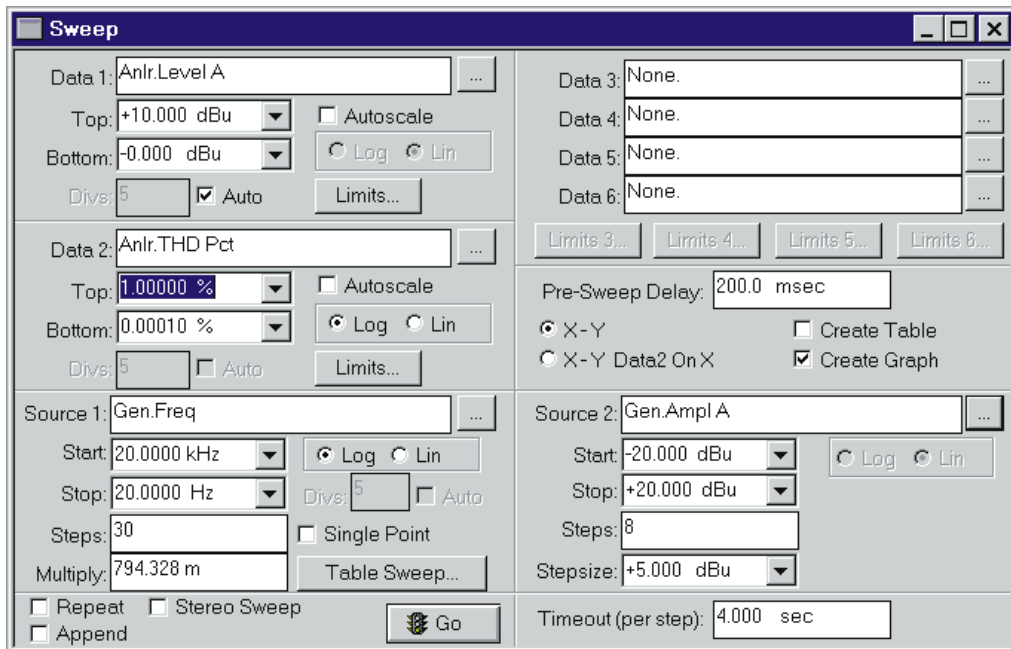



Figure 15-2 Sweep Panel, Large Version

The Sweep Panel can be displayed in small and large (double-width) versions. Clicking on the panel icon on the Panels Toolbar brings up the small version panel. Holding down the  key while clicking on the toolbar icon brings up the large version. To switch between the small and large panels, double-click anywhere in the Title Bar.


The normal default start-up mode of APWIN software shows the Sweep panel on the page two screen.

The small form of the Sweep panel is organized into three major sections—Data 1, Data 2, and Source 1. The Data 1 section controls the graphing of a dependent parameter, typically a measurement value, on the Y (vertical) axis with calibration at the left side of the graph. The Data 2 section permits a second dependent parameter to be plotted, with calibration at the right side of the graph. The Source 1 section controls the independent swept value, typically a stimulus parameter setting such as generator frequency or amplitude, and the X (horizontal) axis calibration of the graph.

## Source 1

Source 1 is the parameter that is deliberately changed as the independent variable during a sweep test, and normally provides the X-axis calibration of a graph.

To select a Source 1 parameter:

Mouse: Click on the button with the ellipsis (three dots)  at the right end of the Source 1 field. A “Browser” box will be displayed. In the Instrument column, click on the instrument module containing the parameter to be swept.

The choices in the Parameter column change with the instrument module selected. The check boxes below the parameter column

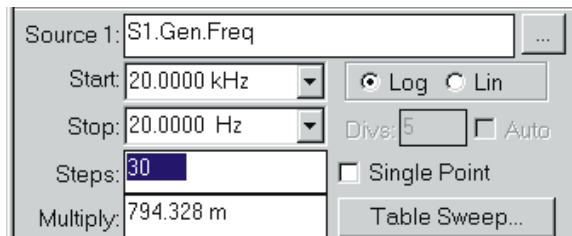


Figure 15-3 Source 1 Fields of Sweep Panel

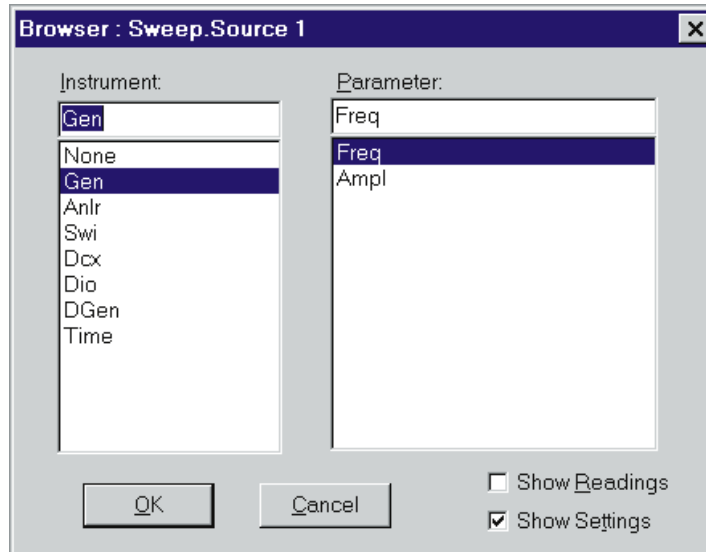


Figure 15-4 Source 1 Browser

permit only readings, only settings, or both readings and settings to be displayed in the Parameter column. In the parameter column, double-click on the desired selection or click on the selection and click on OK.

## Sweep Range, Log/Lin

The range across which the independent variable is swept during a test and the sweep function (logarithmic versus linear) are controlled by the Start and Stop fields and the Log/Lin radio buttons.

### Start-Stop

The beginning and ending values for a sweep are entered in the Start and Stop fields. All units for the parameter being swept are available for selection.

### Log-Lin

If neither Start nor Stop value is zero or a negative number, the Source 1 axis may be calibrated on either a logarithmic or linear scale. The Log-Lin buttons will be gray with linear selected if either value is zero or negative. To change the horizontal axis between logarithmic and linear calibration:

## Sweep Resolution

The span between the Start and Stop values will be stepped through in either equal size increments (Lin mode) or equal percentage increments (Log mode). The number of steps, size of each step, log-lin selection, and span between Start and Stop values are interrelated. Only integer values are accepted in the Steps field. For any span, if a value is entered into the Steps, the span will be exactly divided into that number of equal (Lin) or equal-percentage (Log) steps which determine all intermediate values of the Source 1 parameter. The field immediately below the Steps field will display the Step Size if a linear sweep or the step-to-step Multiplier if the sweep is logarithmic. It is also possible to specify a Step Size or Multiplier directly by entering a value into that lower field.

If the value entered does not result in an integer number of equal (lin) or equal-percentage (log) steps, the sweep will consist of N-1 steps of the exact value specified (where N = Steps value) plus a final, smaller step to the Stop value.

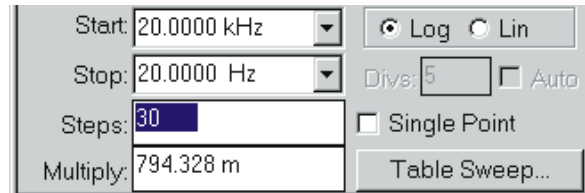


Figure 15-5 Log Sweep; Resolution Value is Constant “Multiply” Value

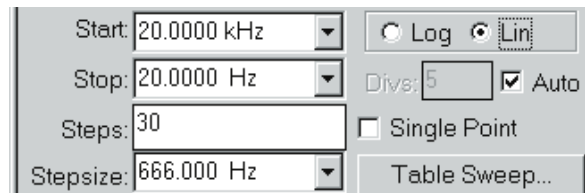


Figure 15-6 Lin Sweep; Resolution Value is Constant “Stepsize” Value

## X-Axis Divisions (Vertical Grid Lines)

On logarithmic axes, placement of graph divisions (grid lines) is pre-determined and the “Divs” field and Auto check box are gray and not functional. On linear axes, the user may specify a specific number of graph lines or divisions or may permit APWIN software to automatically select the number of divisions. When the Auto division mode is off, a

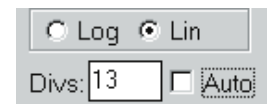


Figure 15-7 Manual Setting of Number of X-axis Grid Lines

new # Divs value may be entered or the existing value edited from the keyboard, followed by pressing .

## Single-Point “Sweeps”

It is frequently desired, particularly in manufacturing test and quality assurance environments, to make single-point measurements rather than a swept series of measurements across a range. To

accomplish this with APWIN, check the “Single Point” checkbox near the Go button at the bottom of the Sweep Panel. The Stop and Log/Lin control will become gray and the Steps and Multiply or Stepsize controls will disappear

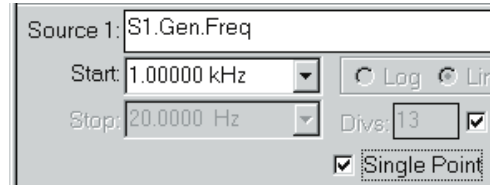






Figure 15-8 Source 1 Settings for Single-point Measurement at 1.0 kHz

since they are irrelevant in a single point “sweep”. When Go is clicked or  pressed, a single measurement will be made at the Start value entered in the Source 1 area of the Sweep panel. Since an X-Y graph cannot be drawn for a single point measurement, the Data table will automatically be displayed to show the one row of data when  is pressed.

## Go button

The Go-Stop button starts or stops a sweep. The Go button has the identical function to the Go icon in the toolbar and the  function key. When a sweep is in progress, the button displays Stop and will stop the sweep if pressed. This is the identical function to the Stop icon in the toolbar.

## Single vs. Repeated Sweeps

Pressing the Go button or  function key will result in a single sweep from Start to Stop values, or a continuing succession of sweeps until the Stop button is pressed, depending upon the Repeat checkbox. When Repeat is checked, a continuous series of sweeps will result until




manually halted. If Repeat is not checked, the sweep will automatically halt when the Stop value is reached.



Figure 15-9  
Repeat and  
Append  
Checkboxes

Repeating real-time sweeps and graphs may be interrupted at any time by the Stop button or Esc key. Only the data taken since the start of the sweep will be retained in memory, Data Editor, and on the graph (unless the Append box was also checked). When DSP-based units are performing FFTs or multitone tests, repeating sweeps should be interrupted only by unchecking the Repeat box and letting the current acquire-transform-process-display cycle complete itself. Interrupting a repeating FFT sweep by the Esc key or the Stop button will result in invalid data in the DSP unit and a subsequent zoom, reprocess, or retransform command will result in meaningless graphs.

## Appended Sweeps

The normal operation of APWIN is to discard any data remaining from previous sweeps each time the  key is pressed or the Go button is clicked. If it is desired to retain the previous data in addition to making a new sweep:

Any previous data will be replaced when a new sweep starts, or the new data will be appended to the existing data, depending upon the Append checkbox. When Append is checked, old data is retained and the new data is appended to it. When Append is not checked, old data is replaced whenever a new sweep starts.

## External Sweeps

It is frequently necessary to make and graph a series of measurements where some external, uncontrollable source is the independent variable. Common examples include frequency response measurements or other swept tests where the sweeping signal is pre-recorded on a test tape or test CD, or testing of a transmission link where a remote generator (not under control of APWIN software) is providing the signal. In these cases, APWIN software cannot control the values, direction of progression (high to low versus low to high), or

dwel times of the signal. APWIN can, however, measure the changing parameter of the incoming signal (usually frequency but sometimes level) and use those measurements as the X-axis calibration. This mode of operation, where a measurement (Reading) drives the data-taking process and calibrates the X-axis, is called External Sweep.

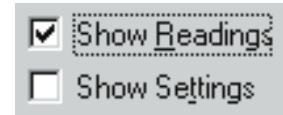


Figure 15-10 Readings & Settings Selection in Source and Data Browsers

**Readings** are measurements from an instrument. Examples include measured amplitude from a Level meter, measured Frequency from a Frequency counter, or measured Time from a clock. **Settings** are controllable values to which an instrument parameter can be set. Examples include generator amplitude, switcher channel number, or the type of dither sent to a digital output. To set up an External Sweep, Readings from an instrument suitable for measuring the parameter must be selected at Source 1, rather than the more conventional selection of a Settings parameter for an internal instrument. Thus, in the Sweep Source 1 Browser that appears when

the Ellipsis button at the end of the Source 1 box is clicked, the “Show Settings” check box should not be checked and the “Show Readings” box should be checked. For swept-frequency

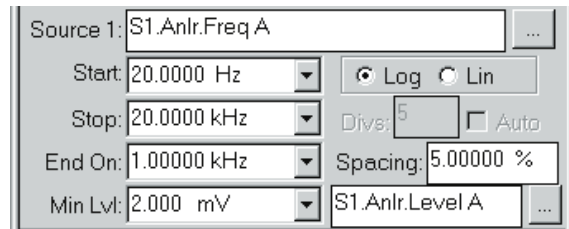


Figure 15-11 Source 1 Fields in External Sweep Mode (a Reading as Source Parameter)

measurements from a test tape or disc, for example, the Source Instrument selection might be an Analyzer (analog or digital as appropriate for the domain of the signal) with the selected measured Parameter as Frequency.

Sweep settling is operative during External Sweeps. Since the Source 1 parameter is also a measured parameter in External Sweeps, the Sweep Settling panel values for the Source 1 parameter must be appropriate for the signal being measured.

When a Reading parameter is selected at Source 1, several of the Source 1 control fields change. The Start, Stop, and Log/Lin controls

remain, new fields appear for “End On”, “Spacing”, and “Min Level”, and the “Steps” and “Multiply/Stepsize” fields disappear since those parameters are determined by the external signal source. The discussion of External Sweep setup and operation is organized into the following topics:

## Start, Stop, and Sweep Direction

---

The values entered in the Start and Stop fields at Source 1 on the Sweep panel not only determine the end values of the X-axis scale on the graph, but they define the sweep direction which APWIN software is expecting. If, for example, the Start value is 20 Hz and the Stop value is 20 kHz, APWIN expects the external signal to move continuously upwards in frequency. If the actual sweep direction of the external signal is opposite to the direction implied by the Start and Stop values, APWIN will not plot the measurements on the graph even though they are retained in memory and can be viewed in the Data table. The selection of Log or Lin only affects the X-axis calibration of the graph, with the actual progression through swept values being determined by the external signal.

## Min Level

---

The “Min Level” field in the Source 1 section of the Sweep panel permits the user to specify an input signal amplitude below which data will not be taken. On analog test tapes and analog disks, for example, there are frequently periods of “silence” between the test tones as the tape or disk plays through a series of tracks. However, “silence” in an analog medium may be only 40 to 60 dB below the recorded signal level. This may be a sufficient amplitude for the measuring instrument (usually the Frequency counter) to obtain occasional but spurious measurements that would result in nonsense points on the graph. By specifying a Min Level value less than the expected test tone level but above the expected noise level, the measuring and graphing process will be disabled during the interval between test tones. The measuring instrument for Min Level determination is selected in the field next to the Min Level value field. Click on the ellipsis button at the end of the field and use the Browser to select the instrument and parameter to supply Min Level readings. Normally, the selection should be the

Instrument and Level meter connected to the same channel as the Instrument whose Readings are selected at Source 1.

## Spacing


---

The “Spacing” field on the Sweep panel permits user control of how much the Source 1 parameter must change from its previous settled value before APWIN software will take another measurement set. If the Spacing value is set to too small a value, minor variations in the Source 1 parameter could result in multiple data points being measured for the same nominal value from the external signal. For example, momentary speed variations from an analog tape machine or analog disk player while playing a nominal 1 kHz tone could result in multiple measurements if the Spacing value were very small, such as 0.1% or less. If the Spacing value is too large, measurements will not be made from every value in a series of closely-spaced values of the external signal.


When the external signal continuously varies from its start to stop values (“glide tone”), the Spacing value determines the resolution of the measured sweep. Small values of Spacing cause APWIN software to capture measurements more frequently. Note that Sweep Settling parameters become more critical when the external signal is a “glide tone”. Typically the Points value for the Source 1 parameter (usually Frequency) will need to be reduced to “1” in order to capture measurements from a continuously-changing source.

## End On

---

The “End On” field in the Source 1 section of the Sweep panel permits the user who has advance knowledge of the external test signal to define a condition under which the sweep test will be automatically terminated at the end of the desired sweep data. This is important if an External Sweep test is one of a series of tests in a Procedure, since otherwise an External Sweep test is normally manually terminated by the user pressing the  key or clicking on a Stop icon when it is apparent that the sequence of test tones has ended.

The use of the “End On” field is based on the fact that many test tapes and test discs have a mid-band reference signal immediately following the end of a series of stepped-frequency test tones. Also, in

the case of a remote System One or System Two generator supplying the signal for transmission testing, the remote generator can be set up such that the generator dwells at its Generator panel setting (which can be a mid-band reference frequency) before and after a Source-1 frequency sweep. In either case, the swept parameter proceeds from some start value (usually a frequency at one extreme of the audio band) and moves to a stop value (usually at the other extreme), then reverses in direction to a mid-band reference value. If the value in the “End On” field equals the value of the swept parameter immediately following the last value of the sweep, and if that represents a reversal in the sweep direction, the test will be terminated and control will be returned to a procedure if a procedure is running. For example, if a test tape consists of a sequence of tones at 40 Hz, 100 Hz, 200 Hz, 400 Hz, 1 kHz, 2.5 kHz, 5 kHz, 10 kHz, 12 kHz, and 15 kHz, followed by another track of 1 kHz reference signal, the Sweep Panel should be set with a low frequency at Start, a high frequency at Stop, and 1 kHz entered in the “End On” field. As the tape plays through the increasing-frequency series of tones and then reverses the direction of frequency change from 15 kHz to 1 kHz, the test will automatically terminate, equivalent to pressing the  key or clicking on Stop.

## Time Sweeps

An External Sweep where measurements of time calibrate the X axis are called Time Sweeps. These produce “chart recorder” style measurements and are commonly used to measure instability and drift of audio parameters or to plot moment-to-moment variations such as wow and flutter as a tape is played, or phase jitter.

To set up a Time Sweep measurement, click on the ellipsis button at the right end of the Source 1 box on the Sweep panel. In the Browser that appears, select Time as the instrument.

As with other External Sweeps, check the “Show Readings” checkbox. Select “External Sweep Time” as the parameter and click the OK button.

You may now enter the desired Start and Stop times and either the number of Steps or Stepsize.

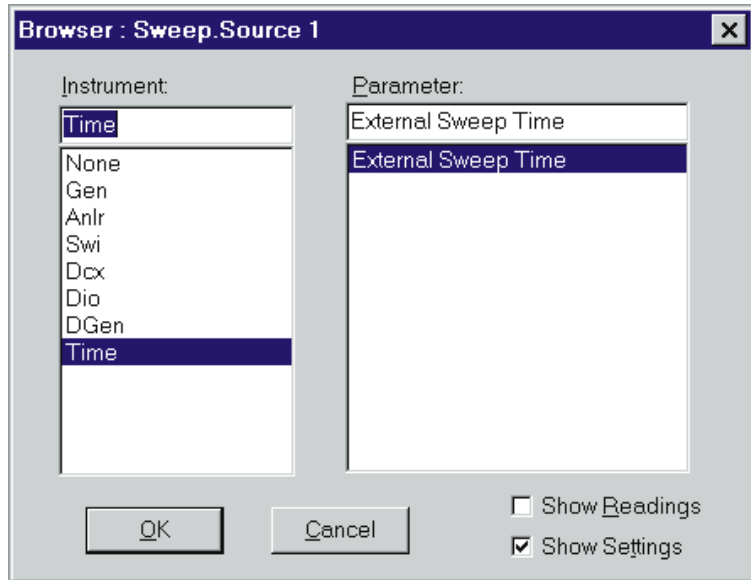


Figure 15-12 Source 1 Browser with Settings for Time “sweep” (Chart Recorder Mode)

A number of factors determine when, and how close together in time, readings are actually taken. Sweep Settling introduces considerable delay and variability in timing. Settling is often set to None for measurements such as phase jitter of wow and flutter versus time. The Reading Rate of the instrument plotted sets a minimum interval between readings. For example, at 8 readings/second, two consecutive readings cannot be closer than 125 ms. Windows and APWIN servicing of other panels and the graph or Data table (in Tabular display) slows down the ability to capture measurements close together. Thus, it is best to “kill” all panels, the graph, and the Data table, to be sure that the Utilities Configuration dialog is set for readings not to be active during sweeps, and to be sure that no other Windows applications are running when the most closely-spaced measurements are desired.

## Table Sweeps

Table Sweeps are sweeps where the Source 1 values are defined by a user-furnished tabular list of values from a “Sweep Table” or “Step Table” file (.ADS file type), rather than being internally computed from

the Start, Stop, and number of Steps values on the Sweep Panel. A Table Sweep permits testing at arbitrary Source 1 values and use of exact, “round” numbers which is often desirable when converting older, manual test procedures to automation. The Start, Stop, and Log/Lin selections serve only to define the X-axis of the graph in Table Sweeps. Some actual Source 1 values could lie outside the graph X-axis end points, depending solely on the values in the table used. The sequence of values in a Table Sweep need not even be monotonic (progressing continuously in one direction), but can jump back and forth between any arbitrary values as long as they are within the operating range of the instrument setting parameter being varied.

Creation of a “Step Table” or “Sweep Table” file is accomplished in the Data Editor, with further editing (if required) normally accomplished in the Attached Files Editor. The same basic techniques are used to create Sweep Table files and Limits files. In a Sweep Table file, the column of data to be used must be the same parameter as the Source 1 instrument setting parameter that will be controlled. Thus, for a generator frequency sweep via Table Sweep, the Sweep Table file must have a column of frequency values. See the Limits chapter for information on how to create Sweep Tables.

Set-up of Table Sweep function is starting by clicking on the Table Sweep button just above the Go button, near the bottom of the Sweep Panel. Note that the Table Sweep button will not be visible if an External Sweep (an instrument Reading selected instead of a Setting) is set up at Source 1. The Table Sweep dialog and Browser will appear, as illustrated in Figure 15-13.

Clicking on the down arrow near the right end of the Files box will display a list of any Sweep Table files already associated with the present test, including their entire path names.

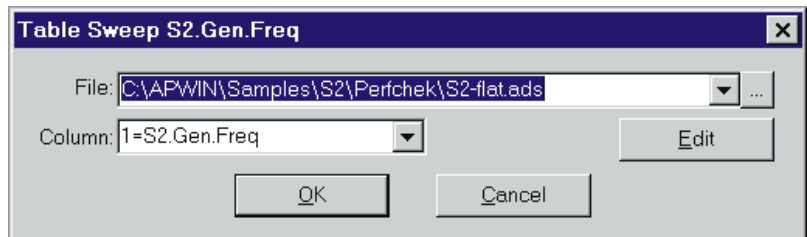


Figure 15-13 Table Sweep Dialog Box

If the desired Sweep Table file name cannot be found on the displayed list, clicking on the Ellipsis button at the very end of the Files box displays a dialog permitting navigation to the directory (folder) which contains the desired Sweep Table, and selection of the specific Sweep Table (.ADS) file.

When the Sweep Table file has been selected, click on the down arrow at the right end of the Column box. A list will be displayed of the column numbers and column headings of the selected file. Choose the column whose data is to be used as the Source 1 values, and click on OK to make the selection. If you do not remember the exact column contents of the Sweep Table, click the Edit box in the Sweep Table dialog to display the entire contents of the Sweep Table file in the Attached File Editor, in order to properly select the desired column. It is also possible to change the value of points or add or delete rows to and from the Sweep Table via the Attached File Editor. If any values are changed, a warning message will be displayed upon leaving the Attached File Editor, pointing out that the revised version must be saved if the changes are to take effect. Displaying the Sweep Table contents in the Attached File Editor will cancel the Sweep Table dialog, so it will be necessary to click on the Table Sweep button again to select the column to be used.

Note that when the Table Sweep selections have been completed, the Step size or Multiply field, number of Steps field, and the Single Point checkbox all disappear since they are not relevant in Table Sweep mode. A single point Table Sweep can be created, if desired, simply by using a Sweep Table file that has only one value. Clicking Go will now produce a Source 1 sweep using the exact values from the selected column of the selected file as Source 1 values.

---

## Data 1 and 2 Selection

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The Data 1 parameter is always plotted according to the calibrations of the left-hand graph vertical axis and Data 2 always plots against the right-hand axis. The color of the traces are initially assigned by APWIN defaults, but the defaults may be over-ridden by user selections in the Graph Legend; see page 16-10. To select a Data 1 or Data 2 value:



Mouse: Click on the button with the ellipsis (three dots) at the right end of the desired field. A “Browser” box will be displayed, illustrated in Figure 15-15. In the Instrument column,

click on the instrument module containing the meter or parameter to be plotted. The choices in the Parameter column change with the instrument module selected. The check boxes below the Parameter column permit only readings, only settings, or both settings and readings to be displayed. In the Parameter column, double-click on the desired selection or click once and then click on OK.

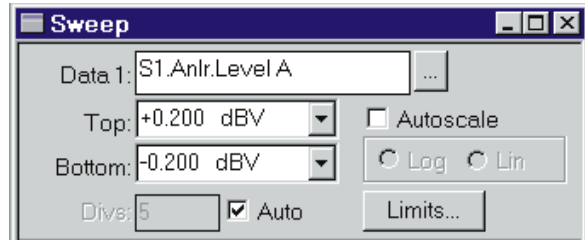


Figure 15-14 Data 1 Section of Sweep Panel

## Top and Bottom

The graph top and bottom values are determined by entries in the Top and Bottom fields. Any of the available units for the parameter being plotted may be selected.

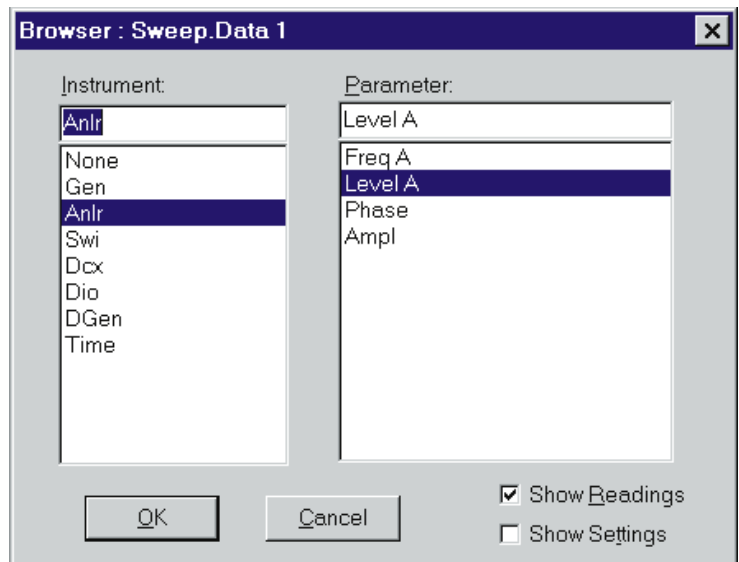


Figure 15-15 Browser Box for Data 1 Through Data 6

## Log-Lin

If neither the graph Top nor graph Bottom value is zero or a negative number, either Data axis may be calibrated on either a logarithmic or linear scale. The Log-Lin buttons will be gray with linear selected if either value is zero or negative.

## Number of Divisions (Grid Lines)

On logarithmic axes, placement of graph divisions (grid lines) is pre-determined and the # Divs field and Auto check box are gray and not functional. On linear axes, the user may turn Auto division mode off and specify a specific number of graph lines or divisions, or may permit APWIN software to automatically select the number of divisions.

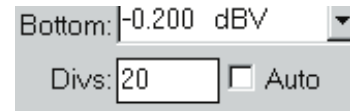


Figure 15-17 Manual Selection of Y-axis Grid Lines

## Autoscale

When Autoscale is not enabled, traces may plot off the top or bottom edge of the graph if the original Top and Bottom values were not properly chosen. With Autoscale enabled, APWIN software will automatically increase the graph span as necessary to accommodate data




Figure 15-16 Data 1 Autoscaling Enabled

values that exceed the originally-set graph Top or Bottom values. Autoscale mode only functions point-by-point as data values are acquired during a sweep; turning Autoscale on or off has no effect on data already in memory at the end of a sweep. (To optimize the graphic resolution for data already in memory following a sweep, use the Optimize function accessible by clicking the right mouse button on the Graph.)

## Limits

See the Limits chapter beginning on page 17-1 for complete information on attaching acceptance (Pass/Fail) limits to a test.

## Data 3 Through Data 6

The large version of the Sweep panel controls the plotting of data parameters 3 through 6. Clicking on the panel icon on the Panels Toolbar brings up the small version panel. Holding down the  key while clicking on the toolbar icon brings up the large version.

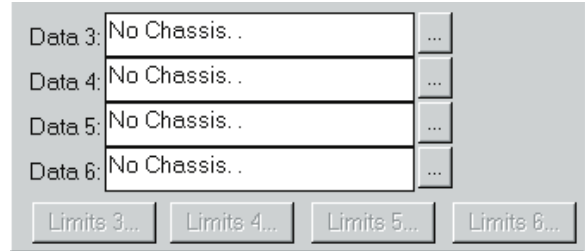


Figure 15-18 Data 3-6 Parameter and Limit File Selection Section, Large Version of Sweep Panel

### Plotting Data 3-Data 6

The Data 3 through Data 6 fields select parameters for the plotting of up to four traces in addition to Data 1 and Data 2 on the main Sweep panel. The color of the traces are initially assigned by APWIN defaults, but the defaults may be over-ridden by user selections in the Graph Legend; see page 16-10. Since the Graph has only two Y axis calibrations (left and right), Data 3 through Data 6 can be plotted only if they share a “measurement domain” with either Data 1 or Data 2. For example, if Data 1 has been assigned to THD % and Data 2 to Level, it will not be possible to plot Phase at Data 3, 4, 5, or 6 since the two vertical axis calibrations are taken up by % and amplitude (Level) units. If it is desired to simultaneously plot the level on channels A and B and the phase difference between them, Phase must be assigned to either Data 1 or Data 2 so that either the left or right vertical axis displays phase units. If Phase is assigned to Data 1 and Ampl to Data 2, for example, Level B can then be assigned to Data 3 and will use the Data 2 units, log/lin selection, graph top and bottom, autoscale, optimize, etc.

If one or more of the Data 3 through Data 6 fields are in use, the Title Bar at the top of the small version of the Sweep Panel will list the active Data fields.

To select a parameter to be plotted in any of the Data 3 through Data 6 fields, click on the ellipsis button at the right of the desired Data

field to view the identical Data Browser described above for Data 1 and Data 2.

## Stereo Sweeps

Stereo and two-channel devices are quite common in the audio field. System One and System Two can measure certain parameters simultaneously on both input channels—for example, level on both channels or acquisition and FFT on both channels. Other parameters, such as THD+N, can only be

measured on one channel at a time. In cases such as this, the Source 1 parameter must be swept twice with the appropriate analyzer input channel switching between sweeps. Stereo sweeps of parameters such as separation (crosstalk) require analyzer input channel switching between the two sweeps and also require corresponding switching of the generator output channel—drive A while measuring B, then drive B while measuring A. In any of these cases, measurement data from both channels will be combined onto a single graph in a single test file, with simplified sweep panel setup, by use of Stereo Sweep mode.

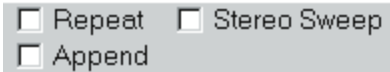


Figure 15-19 Stereo Sweep Checkbox

### Setting Up a Stereo Sweep

A Stereo Sweep is set up by:

- making the appropriate generator, analyzer, Sweep panel Source 1, and Sweep panel Data 1 settings to test one channel of the stereo device
- then checking the Stereo Sweep checkbox at the bottom of the Sweep panel (see the figure).

When the Stereo Sweep box is checked, APWIN software determines which parameter is the “stereo twin” of the selected Data 1 value and writes that parameter into Data 3. Both traces will share the Data 1 (left axis) selections for units, graph top and bottom values, log/lin calibration, and number of divisions. The Stereo box cannot be checked if the Data selection is a parameter with no “stereo twin” (for

example, if Phase or DCX-127 DMM Volts is selected at Data 1). If the Sweep panel is displayed in its large form when the Stereo Sweep box is checked, the “stereo twin” parameter in Data 3 will immediately be visible. If the Sweep panel is displayed in its small form, the title bar at the top of the panel will show that Data 3 is active, as shown in the figure.



Figure 15-20 Sweep Panel Title Bar with Data 3 Active

APWIN also determines whether a single sweep is sufficient (when both parameters can be measured simultaneously) or whether the sweep must be repeated after switching input and output channels when necessary. When successive sweeps are required due to measuring a parameter which can only be measured on one channel at a time, both the digital and analog generator output channels and digital and analog analyzer input channels are switched if the instrument is a Dual Domain version.

When the Stereo box is unchecked, the Data 3 parameter is automatically removed and only a normal, single sweep will result.

## Stereo Sweeps and Limits

If a limit file (.ADL) is selected at Data 1 or 2 before the Stereo box is checked, the Data 1 limits will automatically be used at Data 3 and the Data 2 limits will be used at Data 4. If it is desired to use a different limit file or no limits at Data 3 or 4, the Limits buttons for Data 3 and 4 (large version of the Sweep panel) may be used to change or disable the automatic selection. For more information on creation and use of Limits files, see the Limits chapter beginning on page 14-1.

## External Stereo Sweep

External sweeps, as discussed earlier in this chapter, are used to make swept measurements and graphs from an external, uncontrollable signal source such as a pre-recorded test tape or compact disc or a remote, sweeping oscillator. To make an External sweep of a stereo device, set up the Source 1 and Data 1 areas as described in the External sweeps section. Then, check the Stereo checkbox. The “stereo twin” of the Data 1 parameter will be written into Data 3 as described earlier.

If the parameters at Data 1 and 3 can be measured simultaneously, the External stereo sweep will then proceed exactly like a single-parameter External sweep. If the parameter to be graphed is one that can only be measured on one channel at a time (such as THD+N), a different form of sweep logic is used. The parameter will first be measured on the channel selected on the analyzer panel and plotted as Data 1. Then, the analyzer input will be switched to the opposite channel and the resulting measurement plotted as Data 3. The analyzer input is then switched back to the original channel and the Source 1 parameter is monitored, awaiting a change of at least the Spacing value before measuring another pair of points. This switching logic is based upon the fact that the measurement system is typically able to measure both channels in less time than the typical test tape or disc dwells at each frequency. Thus, a single pass through the recorded or remotely-originated sweep is sufficient to produce measurements of both channels even on parameters which must be measured one channel at a time.

If the rate of change of Source 1 parameter is an External Stereo sweep is so fast that the analyzer has not completed a settled measurement on the second channel before the Source 1 parameter steps to its next value, no Data 3 trace will be plotted. If the Data table is used to examine the data in tabular form, the symbols “—B” will be displayed in the cells that contain bad data. If cursors are used on the graph trace with missing values due to an excessively-fast sweep, the cursor readout box will show “—” and the delta-Y display box will disappear if the cursor is moved to one of the bad values.

## Plotting Measured Values X-Y

The large version of the Sweep panel permits plotting measured parameters on the Y axis versus another measured parameter on the X axis. Select the “**X-Y Data2 on X**” choice rather than “X-Y” which is the common mode for conventional graphs. With “X-Y Data2 on X” selected, the Data 1 parameter will be plotted as a Y axis displacement versus Data 2 measurements on the X axis. Data 2 values will calibrate the X axis and the Source 1 values and calibration will not

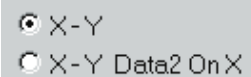


Figure 15-21 Plotting Data 2 on X Axis

be displayed on the graph at all. Source 1 will still be the independent variable, stepping through the range defined, and driving the test to create the changes in Data 1 and Data 2 variables that will be plotted.

Changes between conventional X-Y mode and X-Y Data2 on X may be made freely before or after a test is run, or even with a test file re-loaded from disk. Note that limits cannot be graphed nor data compared to limits when the graph is in X-Y Data 2 on X mode.

## Data Display Mode




Sweep test results may be displayed in real time either on an X-Y graph or as numbers in a table (or both). The Create Graph and Create Table checkboxes determine which forms of display will be used if neither is already visible on the current Workspace page when  is pressed.




Figure 15-22 Display Mode Controls

If **Create Graph** is checked, APWIN software will automatically display a graph when  is pressed and will plot onto that graph.

If **Create Table** is checked, the Data table will automatically be displayed when  is pressed and the data will be presented in tabular fashion in the Data table with the data appearing line by line as the sweep progresses. Since there is no issue of insufficient graph vertical axes available for calibration with tabular display, it is possible to have up to six data parameters of any mixture of unit domains (Volts, Hz, %, dB, etc., etc.) simultaneously displaying during a sweep with Tabular display..

If both boxes are checked, both forms of display will automatically be created. If neither checkbox is checked, the data is still taken and, if limits are in use, compared to limits for Pass/Fail decisions. This “no display” mode is actually the fastest and may be desirable in manufacturing test situations.

## Source 2 and Nested Sweeps

The large version of the Sweep panel permits “nested sweeps”, where a second parameter is stepped between repetitions of the Source 1 sweep. Clicking on the panel icon on the Panels Toolbar brings up the small version panel. Holding down the  key while clicking on the toolbar icon brings up the large version. If the Source 2 field is in use, the Title Bar of the small version of the Sweep Panel will indicate the fact.

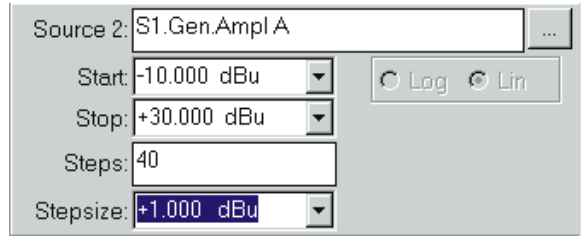


Figure 15-23 Source 2 Section, Large Version of Sweep Panel

A nested sweep controls two different settings parameters in an inter-related manner to produce a family of traces on the Graph. Examples include sweeping generator frequency at Source 1 and generator amplitude at Source 2 to produce a family of frequency response measurements at different levels, or sweeping generator frequency at Source 1 and switcher channels at Source 2 to produce overlaid frequency responses for many channels of a multi-track tape recorder or mixing console. The X-axis calibration and labeling of the graph is determined by the Source 1 parameter. The Source 2 parameter values are not intrinsically visible from the traces, but the Graph legend will display Source 2 values at the cursor locations along with the Data and Source 1 values. The sequence of sweeps is as follows: select the Source 2 Start value, and sweep through the Source 1 range from Start to Stop. Step the Source 2 parameter one step and sweep Source 1 again. Repeat for each defined step of Source 2, concluding the test by sweeping the Source 1 range at the Source 2 Stop value.

The Source 2 and Source 1 Browser operate identically. See the Selection of Parameter to be Swept (Independent Variable, Horizontal Axis) Topic for set-up instructions.



## Timeout

As discussed in the Sweep Settling Panel section starting on page 15-23, settling algorithms are normally used during sweeps to assure that the data values have

settled after any initial transients caused by the sweep so that measurements are repeatable. However, it is possible that the instability of the measured parameter is greater than the Tolerance value present used in the Sweep Settling algorithm. For example, if a Tolerance of 1% (about 0.1 dB) were in use while measurements were made of noise or an extremely noisy signal, it might never be possible to obtain the required number of consecutive points within 1% of one another in amplitude. The result could be complete failure to make a measurement (software hang-up) or, at the minimum, greatly slowed measurements.

The Timeout parameter is intended as a “safety valve” to avoid such hang-ups. At any attempted measurement point where settling has not been achieved within the time value specified in the Timeout field, APWIN software will compute the numerical value of the last six measurements, plot that value on the graph or display it in the Data table, and step on to the next point of the sweep. In a graph display, each timeout point is indicated by a white “T” at the upper margin of the graph, directly above the plotted point. In the Data table, each timeout point is indicated by the letter T following the data. In the Log File, the Pass/Fail message (if enabled) shows the total number of timeouts that occurred during a sweep and indicates “FAILED” if there were one or more timeouts, even though the eventual averaged data may have been within limits. The Log File (unless “None” is checked under “Sweep Data”) also includes a line for each measured point that timed out result in a row showing the measured value and a letter T.

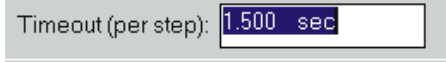


Figure 15-24 Settling Timeout Field, Large Version of Sweep Panel

## Sweep Settling Panel

Measurements by most meters in the instrument are taken continuously, often at rates as rapid as 32 or 64 readings per second. The real-time numeric meter displays and bar graphs show “raw”

	Tolerance	Floor	Points	Delay	Algorithm
THD+N Ratio	3.00000 %	0.00007 %	3	30.00 msec	Exponential
Level:	1.00000 %	25.00 $\mu$ V	3	30.00 msec	Exponential
Frequency:	0.50000 %	200.000 $\mu$ Hz	3	30.00 msec	Exponential
Phase:		+0.5 deg	3	30.00 msec	Exponential
DCV	0.20000 %	+0.00050 Vdc	3	30.00 msec	Exponential
Digital In:	0.00000 %	0 dec	3	30.00 msec	Flat
Genanlr.Level B	1.00000 %	1.000 $\mu$ FFS	3	30.00 msec	Exponential
Genanlr.2 Channel	1.00000 %	1.000 $\mu$ FFS	3	30.00 msec	Exponential
Genanlr.Freq B	0.50000 %	200.000 $\mu$ Hz	3	30.00 msec	Exponential


Figure 15-25 Sweep Settling Panel (GENANLR Loaded as Digital Analyzer Program)




unsettled data, with every measurement made by the meter being displayed on the panel. If transients, noise, or other momentary events affect a reading, the operator may choose to ignore the transient.

During a sweep, each change in stimulus amplitude or frequency introduces transients into the device under test, the generator, and the analyzer. Plotting such unsettled data usually results in jagged, unrepeatable graphs of doubtful value. Repeatable data is normally desired, taken after the device under test and the measuring instrument have settled to near-steady-state conditions. To obtain repeatable results, settled data must be used.

Settled data is normally obtained in APWIN by discarding early measurement points polluted by transients and plotting only the later, repeatable values after the device and instrument stabilize. The Sweep Settling process determines when data has stabilized sufficiently following a transient so that a point may be plotted and the sweep permitted to move on to the next step. The key Sweep Settling

parameters are under user control via the Sweep Settling panel. To display the Sweep Settling panel:

Mouse: Click on the Sweep Settling icon  on the toolbar or select Panels Sweep Settling via the menu bar.

Keyboard: Type    (for Panels sweep settLing).

## Settling Concepts and Algorithms

The general concept of the Sweep Settling Exponential and Flat algorithms is to discard all meter readings during the Delay interval, then to compare the number of successive meter readings equal to the “Points” value against the Tolerance or Floor values. Only when the specified (Points) number of consecutive meter readings agree with one another within the specified Tolerance or Floor values will the data be considered stabilized. It is then accepted for plotting and the Source parameter permitted to proceed to the next step.

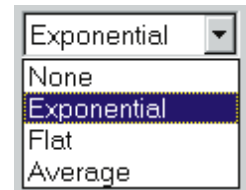


Figure 15-26 Settling Algorithm Selections

If the **Exponential** algorithm is selected, the newest reading (N) must agree with the immediately preceding reading (N-1) within the Tolerance value, with the reading before (N-2) that within twice the Tolerance value, with the reading before that (N-3) within four times the Tolerance value, etc. Figure 15-27 shows the “envelope” within which a succession of readings must fit if Tolerance is set to 1%. Exponential is the recommended settling algorithm for most audio applications, since typical device transients tend to die away in an exponential fashion. Exponential thus will usually provide repeatable results to the Tolerance acceptable to the user in the minimum length of time.

When the **Flat** algorithm is selected, the percentage difference between each set of two consecutive readings (N vs. N-1, N-1 vs. N-2, etc.) must be equal to or less than the specified Tolerance value, through the number of readings specified as the Points value. Illustrating the Flat algorithm for 1% Tolerance would result in an “envelope” bounded by two horizontal lines at the plus and minus 1% levels across the full number of Points. The Flat algorithm thus

guarantees that the transients have been settled to the specified Tolerance for some time, which tends to take longer than the Exponential algorithm.

When the **Average** algorithm is selected, points are first discarded for the duration of the Delay interval, as with Exponential and Flat. At the conclusion of the Delay period, the number of consecutive readings specified in the Points field is accumulated and their average value computed and plotted. Tolerance and Floor values are ignored when Average is selected. The Average algorithm is particularly useful when the signal is fundamentally noisy and might never settle within a practical Tolerance.

When **None** is selected as the algorithm, no settling process takes place for that meter. Measurements such as wow and flutter or phase jitter are examples of cases where no settling should be used, since it is desired to see the extreme variations in measurements.

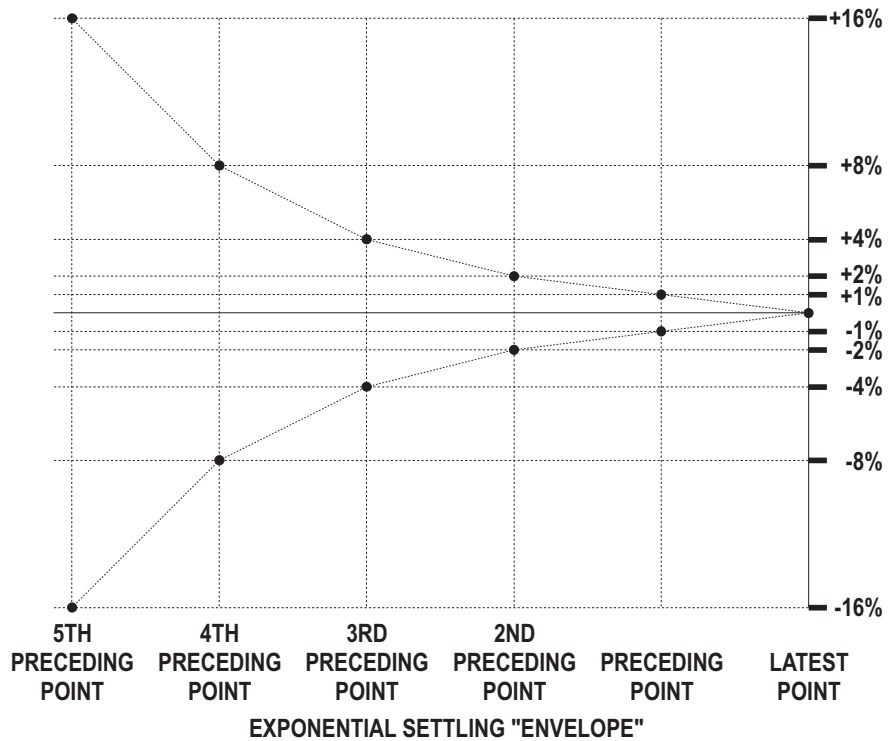


Figure 15-27 Graphic Representation of Exponential Settling Criteria, 1% Tolerance and 6 Points

## Tolerance and Floor

---

The Tolerance value which should be entered is the amount of variability the user is willing to accept from test to test. A Tolerance value of 0.1% (about 0.01 dB) or even slightly smaller may be appropriate when making frequency response measurements on the test system itself or on an external device known to be very flat and being measured under excellent signal-to-noise conditions. At the other extreme, Tolerance values of 10% to 25% (1 to 2 dB) may be required to obtain data under noisy conditions, or when making measurements with a random noise signal as the stimulus. The default value of 1% (about 0.1 dB) is a good starting compromise for most level measurements.

The Floor value is used by the algorithms instead of the Tolerance value whenever the Floor value is larger. When the measurements values are greater than a few percent of full scale on the instrument range in use, the Tolerance value is normally the determining parameter. If the measurements are very near the bottom of the instrument's dynamic range, use of only a Tolerance parameter could result in a "hang up" situation, since the percentage difference between two adjacent values (quantization levels) at the bottom of a meter's range is large. The Floor parameter thus serves as a "safety valve", avoiding slowing or hang ups in the highly resolution-limited situations where the signal is near the bottom of a measurable range. The default values of Floor for each meter are chosen to be approximately the resolution of that meter on its most sensitive range. Since resolution varies with reading rate (slower reading rates give more resolution), it may be appropriate to change the default values when reading rate is fixed at a given value.

## Delay

---

The Delay value is effective even when the Algorithm selection is "None". The Delay time will be taken at the beginning of each nest of a nested sweep, including nested FFT measurements with the FFT at Source 1 and another parameter such as generator amplitude at Source 2. Acquisition of signal into any of the FFT programs will not begin until the Delay value (or 200 milliseconds, whichever is greater) has passed. For Time sweeps where it is desired to make as many

measurements per second as possible, the Delay value should be set to zero in addition to selecting “None” for settling.

## Settling Parameter Independence Among Instruments

---

The settling for each meter in the System is independently controlled by its own row of fields on the Sweep Settling panel. All meters except the Phase Meter have fields for Tolerance, Floor, Points, Delay, and Algorithm. The Phase Meter does not have a Tolerance field and the variability acceptable to the user should be entered into the Floor field. The Reading Meter settling parameters on the top row are displayed only for the Reading Meter function presently selected on the Analog Analyzer panel.

## DSP Measurement Settling

---

The Settling concept does not apply to “batch mode” DSP-implemented measurements where a signal is acquired into memory, an FFT performed, and results graphed. Settling parameters for real-time DSP functions are displayed near the bottom of the Settling panel only for the DSP function presently selected on the Digital Analyzer panel.

Reading Meter and DSP function settling parameters are stored independently for each function. If, for example, different settling parameters are preferred for Amplitude and THD+N, it will not be necessary to re-enter them as the function is switched back and forth.

## Graphs and Printing

The parameters selected on the Sweep panel may be graphed as X-Y relationships by APWIN. The Graph window may be re-sized by dragging any corner or edge. The Graph may be displayed on more than one page and at different sizes on different pages, but the content is identical in all instances.

The user may zoom in, repeatedly if desired, on areas of the graph by dragging with the left mouse button. Zooming back to previous views, optimization of the plot, turning cursors on and off, and several other graph operations are controlled by a dialog box (see Figure 16-2) which appears when the **right** mouse button is clicked anywhere on the graph.

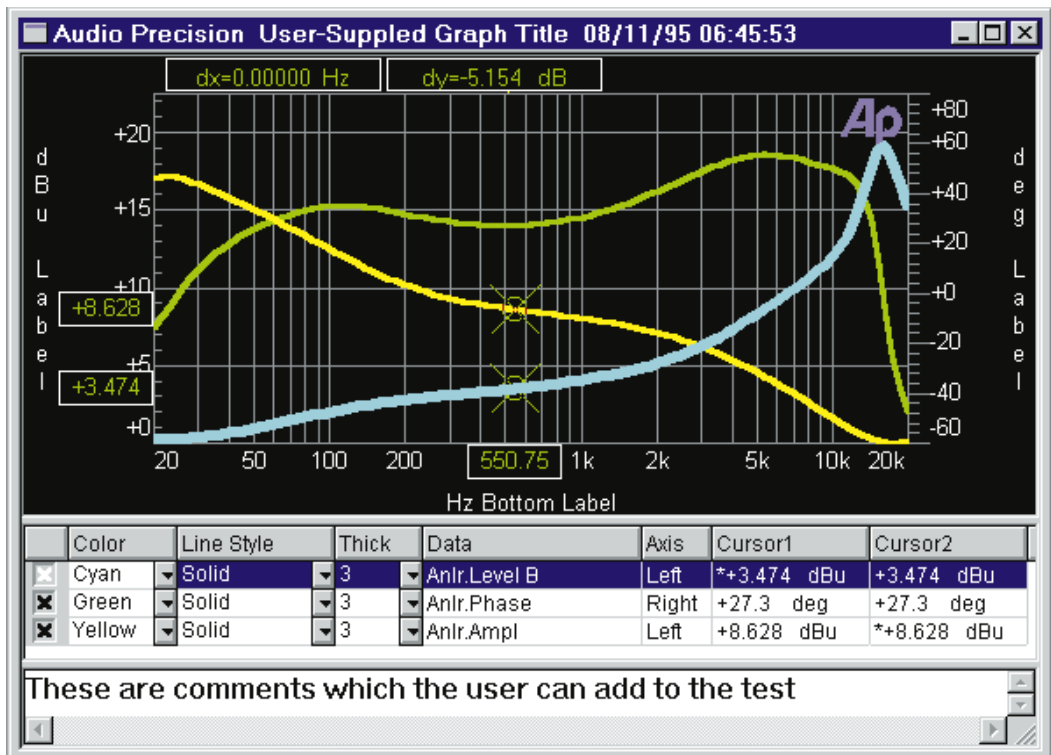


Figure 16-1 APWIN Graph Showing Cursors, Title and Labels, Legend, Comments

Two graph cursors may be dragged with the mouse or positioned by arrow keys to any points in the data for numeric display of data points and difference values. The cursors may both be attached to a single data trace, or each may be attached to a different trace if both traces are the same “domain” (both amplitude, for example).

The user may instruct APWIN to automatically choose X and Y coordinates for maximum resolution of the acquired data range (**Optimize**).

Scroll bars may be displayed to move up and down or back and forth on zoomed graphs.

A user-written title may be displayed at the graph top and labels may be placed in the top, bottom, left, and right margins of the graph.

An alphanumeric legend under the graph permits turning individual traces on and off, selecting trace colors, increasing trace thickness, changing line styles, explains which measured parameter is represented by each trace and which vertical axis calibration (right or left) is to be used, and provides simultaneous display of the value of all traces at the cursor positions.

A Comments editor box permits the user to freely type text which can be displayed on the screen and printed below the graph.

To bring the Graph to the screen:

Mouse: Click on the Graph icon  on the toolbar or select Panels Graph via the menu bar





Keyboard: Type    (for Panels gRaph).

## Zooming

You may zoom in on any area of the graph by clicking the left mouse button at one corner of the section to be magnified, then dragging (holding down the mouse button while moving the mouse) to the opposite corner of the section. A rectangular outline appears while dragging. When the mouse button is released, the graph re-draws to



expand the selected area. You may zoom repeatedly to magnify smaller and smaller areas.

When the graph is a frequency domain (spectrum analysis) or time domain (oscilloscope view) signal from one of the FFT programs, dragging a zoomed view also automatically sends the “Sweep Re-Process Data” () command to the DSP module so that the zoomed view will consist of the maximum possible resolution. Repeated zooms of an FFT spectrum analysis will eventually show the structure of the individual FFT bins. Zooming out also automatically sends the  command. This automatic  with zoom feature may be disabled in the Utilities Configure menu if it is not desired. Note that  can only be effective if the signal is still in DSP memory. A test loaded from disk, for example, will contain only the view of FFT results when the test was saved and increasing resolution during a zoom is not possible.

## Right Mouse Button Features

Clicking the right mouse button anywhere within the graph will display a selection of commands. The choices in the dialog box are: Zoomout, Zoom Out to Original, Optimize Individually, Optimize Together, Optimize Left Only, Optimize Right Only, Copy to Sweep Panel, Display Cursors, Scroll Bars, Title and Labels, and Comment.




### Zoomout

Clicking the normal (left) mouse button on the **Zoomout** choice in the dialog which appears when the right mouse button is clicked on a graph causes the most recent zoom view to be replaced with the previous one. If you have zoomed repeatedly, the coordinates of each zoom have been saved in sequence in a memory stack. You may



Figure 16-2 Mouse Right Button Dialog Box




then Zoomout repeatedly to work back up through the stack, viewing the series of zoomed views in reverse order.

When the graph is a frequency domain (spectrum analysis) or time domain (oscilloscope view) signal from one of the FFT programs, zooming in or out automatically sends the “Sweep Re-Process Data” () command to the DSP module so that the zoomed view will consist of the maximum possible resolution. Note that  can only be effective if the signal is still in DSP memory. A test loaded from disk, for example, will contain only the view of FFT results when the test was saved and zooming in our out with increased resolution is not possible. This automatic  with zoom feature may be disabled in the Utilities Configuration menu if it is not desired.

### Zoomout to Original

---

Clicking on this command in the dialog which appears when the right mouse button is clicked on a graph replaces the current zoomed-in view with the graph coordinates in use when the most recent sweep was started, or with the default initial graph coordinates if no sweep has yet been made since APWIN was launched.

When the graph is a frequency domain (spectrum analysis) or time domain (oscilloscope view) signal from one of the FFT programs, zooming in or out automatically sends the “Sweep Re-Process Data” () command to the DSP module so that the zoomed view will consist of the maximum possible resolution. Note that  can only be effective if the signal is still in DSP memory. A test loaded from disk, for example, will contain only the view of FFT results when the test was saved and zooming in our out with increased resolution is not possible. This automatic  with zoom feature may be disabled in the Utilities Configure menu if it is not desired.

### Optimize Features, General

---

Clicking the left mouse button on one of the **Optimize** commands in the dialog which appears when the right mouse button is clicked on a graph causes the graph to be automatically re-scaled with new top and bottom boundary lines for a maximum-resolution display of the actual range of data presently in memory. Optimizing after the

conclusion of a sweep thus sets new coordinates to best display the entire data set. If an Optimize command is clicked while a sweep is in progress, the graph will optimally display the partial set of data acquired up to that time, with new data taken in the remainder of the sweep then disappearing off the left or right of the graph. Optimize may be clicked as often as desired during a sweep or after manual changes in graph coordinates, but is most effective after the conclusion of a sweep. Four different forms of optimization are provided for better control of the results: Optimize Individually, Optimize Together, Optimize Left Only, and Optimize Right Only.

### Optimize Individually

The Optimize Individually command will be available only when both Data 1 and Data 2 are in use. Optimize Individually causes the Data 1 (left axis) graph top and bottom values to be automatically selected for best display of the actual values in the Data 1 data set, including any data in Data 3 through Data 6 which are mapping onto the Data 1 axis. Simultaneously, Optimize Individually selects new Data 2 (right axis) top and bottom settings for best display of the Data 2 data set and any Data 3 through Data 6 data which maps into Data 2. Even if both Data 1 and Data 2 axes are in the same units, the top, bottom, and number of divisions may not match following an Optimize Individually command, depending upon the actual data range in all the data sets which are calibrated against a particular axis.

### Optimize Together

The Optimize Together command will be available only when both Data 1 and Data 2 are in use. Optimize Together examines all the data in all data sets (Data 1 through Data 6) which map onto either the Data 1 (left) or Data 2 (right) axis, and then selects identical graph top and bottom values and division markings for both left and right axes which will show the entire range of data with maximum resolution and no data points plotting off the graph. Optimize Together thus preserves the ability to graphically compare traces plotted against both left and right Y axes.

## Optimize Left Only

The Optimize Left Only command is available whenever Data 1 is in use. It will optimize only the Data 1 (left) graph top, bottom, and number of divisions for best display of all data sets (Data 1 and potentially Data 3 through Data 6) which are plotted against the left Y axis.

## Optimize Right Only

The Optimize Right Only command is available whenever Data 2 is in use. It will optimize only the Data 2 (right) graph top, bottom, and number of divisions for best display of all data sets (Data 2 and potentially Data 3 through Data 6) which are plotted against the right Y axis.

## Copy to Sweep Panel

---

Zooming and Optimizing cause the displayed graph coordinates (top, bottom, left, and right values) to be different from the values specified on the Sweep panel. Clicking on the **Copy to Sweep Panel** command in the dialog which appears when the right mouse button is clicked on a graph writes the values presently in use on the Graph into the Sweep Panel fields, replacing the user-entered values. This permits then making a new sweep or acquisition across the zoomed-in data range.

## Display Cursors

---

Clicking on the **Display Cursors** command in the dialog which appears when the right mouse button is clicked on a graph turns the graph cursor function of and off. The function is on when a check mark is displayed alongside Display Cursors in the dialog box.

Cursor function consists of two cursors and up to seven cursor value numeric display fields on the Graph. The cursors are named Cursor 1 and Cursor 2. Their initial appearance is as vertical green cursor bars extending from graph bottom to top, positioned near the edge from which the sweep started.

When a trace is selected (by clicking on the desired row in the legend below the graph), the thickness of the selected trace becomes greater and the vertical cursor is replaced by a large “X” at the point of intersection with the trace. An asterisk (\*) appears in the cells at the intersection of the Cursor columns and the selected trace row. Either cursor may be dragged to a new horizontal location with the mouse.

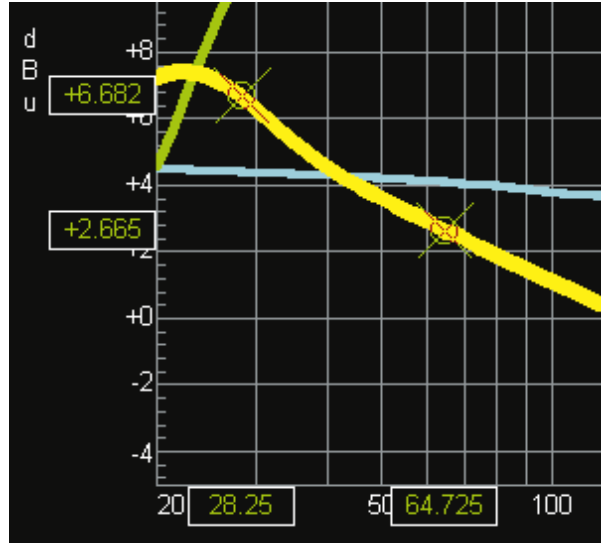





Figure 16-3 Cursors 1 and 2 Selected to Same Trace, Floating Numeric Readout Boxes in Line with Cursors. Cursor Positions May Be Dragged by Mouse on Display Boxes on Horizontal Axis

When the mouse cursor is positioned over a graph cursor or on the cursor value display box on the horizontal axis directly below the cursor, the mouse cursor changes from its normal arrow shape to  for cursor 1 and  for cursor 2. By holding down the left mouse button while the cursor has this shape and moving the mouse, the cursor may be dragged to any data point on the selected trace.

When the focus is on the graph, the selected cursor may also be moved with the horizontal arrow keys. The  with arrow key combination toggles arrow key control between Cursor 1 and Cursor 2. When a trace is selected, the cursor “snaps” from one data point to the next and cannot be positioned between data points.

The X and Y axis values at the intersection of each cursor with the selected trace are displayed in “floating” display boxes in the margins of the graph, aligned horizontally (Y value) and vertically (X value) with each cursor. The difference between the X values of the two cursors and the Y values of the two cursors are shown in display boxes

at the top of the graph. If a “nested sweep” (via Source 2) has been plotted, the value of the Source 2 parameter of the selected trace is also shown at the top of the graph. The Cursor 1 and Cursor 2 columns in the Legend below the graph show the values of cursor intersection with all traces, whether selected or not.

Cursor1	** Select **
*+10.855 dBr	*+10.855 dB
+6.612 dBr	+6.612 dBr

Figure 16-4 Legend Indication to Select Cursor to Second Trace

The two cursors may be selected to two different traces. To do this, first select one of the two desired traces by clicking in its row in the legend under the graph. Asterisks (\*) will appear in the cells in both cursor columns at the intersection with the row which was clicked. Then, click in the column heading box (Cursor1 or Cursor2) of the cursor which you wish to move to another trace (Cursor2 in Figure 16-4). The column head will change to **\*\*Select\*\*** and the entire column will be displayed in inverse video. Click in the cell at the intersection of this column and the row of the trace to which the cursor is to be attached. The asterisk will now appear in that cell and the cursor will be attached to that data trace. Note that cursors may be attached to two different traces only if they display compatible parameters, since the “delta Y” display field would otherwise be meaningless. For example, the cursors may be attached to two traces displaying amplitude, but not to two traces if one displays amplitude and one displays phase.

## Scroll Bars

Clicking on the **Scroll Bars** command in the dialog which appears when the right mouse button is clicked on a graph turns the check mark and Scroll Bars on and off. If the user has not zoomed in on the graph, scroll bars may be turned off for a larger graph. When zoomed, scroll bars permit looking at other portions of the data at the same magnification. The graph will scroll dynamically as the scroll button is dragged.

## Title and Labels

Clicking on the **Title and Labels** command in the dialog which appears when the right mouse button is clicked on a graph will display the Graph Title and Labels dialog box. The Title edit box at the top of

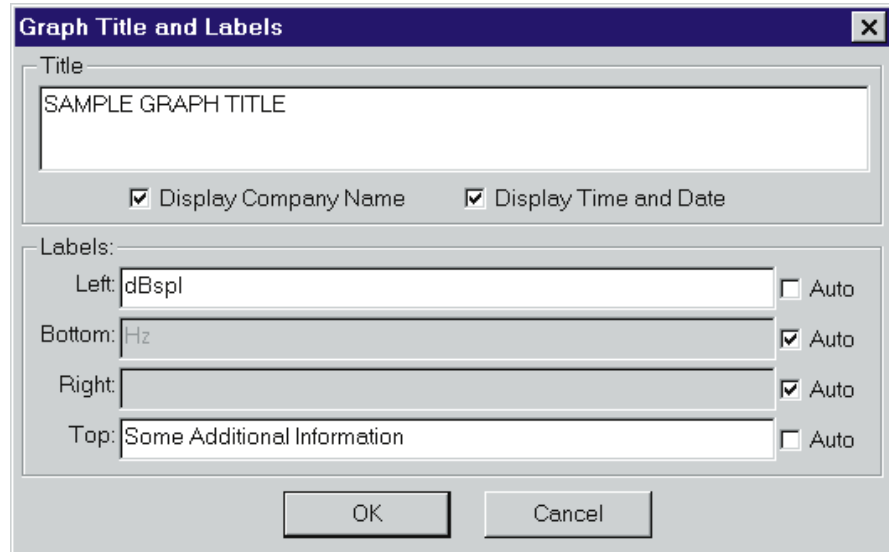



Figure 16-5 Title and Labels Dialog Box, Obtained by Clicking Right Mouse Button on Graph, Selecting Title and Labels

this dialog box permits typing in any desired text which will then appear in the top margin of the graph, following the Company Name (usually Audio Precision) text and preceding the date and time text. The text will dynamically appear in the graph title bar on screen as it is typed. The title bar on screen is limited to displaying one line of text, so the screen display of long titles will be truncated. However, the full title text typed in will display in Print Preview and will be part of a print-out. The “Audio Precision” text display at the left end of the title bar may be turned on and off by the “Display Company Name” check box. “Audio Precision” may be replaced by up to 32 characters of the user’s choice by use of the Utilities Configure menu command (see page 2-45). The date and time displayed at the right end of the title bar may be turned on and off by the “Display Time and Date” checkbox.

The left, bottom, and right labels normally are automatically supplied with the abbreviation for the measurement units graphed on each axis. This automatic label may be replaced or supplemented with additional text. To add custom text, un-check the “Auto” checkbox at the right end of the edit box to be changed. Then, you may delete or add to the text in the edit box. The top label has no automatic entry and may be used for any short text desired by the user.

## Comment

When the Comment line in the graph right mouse button dialog is checked, comments will be displayed below the graph. Comments may be typed directly into the Comment edit box, which is displayed under the Graph Legend as shown in Figure 16-6. The height of the Comment edit box may be adjusted by dragging the boundary line between the graph legend and the Comment edit box up or down. When the mouse cursor is positioned over the boundary, the cursor shape changes from its normal arrow shape to two closely-spaced horizontal parallel bars . With this cursor shape displayed, hold down the left mouse button and move the mouse vertically to drag the Comment edit box height to the desired value, then release the mouse button.

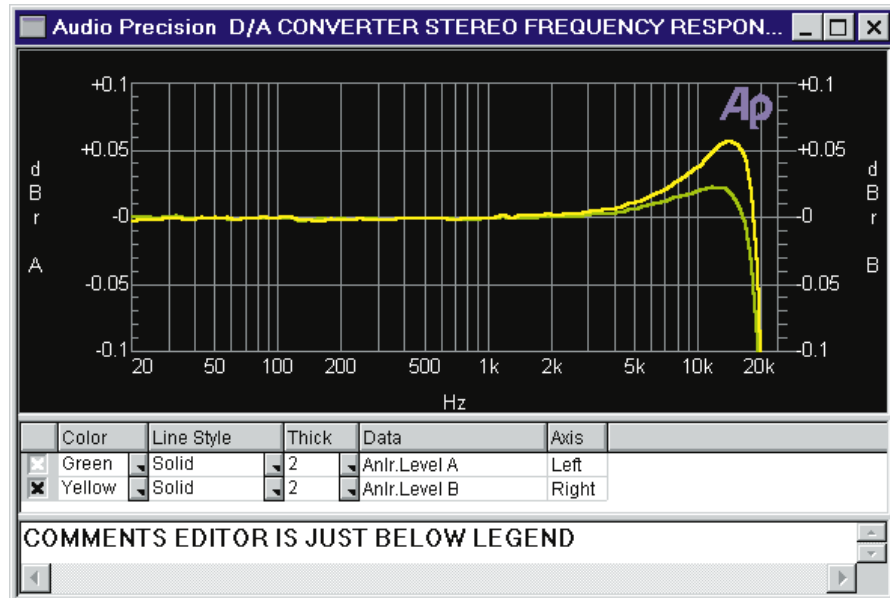


Figure 16-6 Comments Editor

## Graph Legend

The alphanumeric tabular display area underneath the graph is called the Legend. The column heads are Color, Line Style, Thick,



	Color	Line Style	Thick	Data	Axis	Source 2	Cursor1	Cursor2
<input checked="" type="checkbox"/>	Cyan	Solid	4	Anlr.Level A	Left	+20.000 dBu=Gen.Ampl	+23.181 dBV	+23.231 dBV
<input checked="" type="checkbox"/>	Green	Solid	4	Anlr.Phase	Right	+20.000 dBu=Gen.Ampl	+12.1 deg	-24.7 deg
<input checked="" type="checkbox"/>	Yellow	Solid	1	Anlr.BandPass	Left	+20.000 dBu=Gen.Ampl	*+23.062 dBV	*+23.227 dBV
<input checked="" type="checkbox"/>	Red	Solid	4	Anlr.Level A	Left	+10.000 dBu=Gen.Ampl	+13.177 dBV	+13.219 dBV
<input checked="" type="checkbox"/>	Magenta	Solid	4	Anlr.Phase	Right	+10.000 dBu=Gen.Ampl	+12.1 deg	-24.7 deg
<input checked="" type="checkbox"/>	Blue	Solid	1	Anlr.BandPass	Left	+10.000 dBu=Gen.Ampl	+13.051 dBV	+13.216 dBV

Figure 16-7 Graph Legend

Data, Axis, Cursor1, and Cursor2. When Source 2 is used (to create a nested sweep), another column is added to show the Source 2 value of each trace. When limits are attached to a test and displayed on the graph, each limit trace will also have a row in the legend and the color, thickness, etc. of the limit traces may be changed. In addition to these columns, a check box at the beginning of each row permits displaying or not displaying each trace.

### Color (Screen Display)

The cells in the Color column show the color of each trace. The down arrow at the right of each cell displays a selection of seven different colors which may be selected to distinguish one trace from another. If a color is changed in the legend, new sweeps made after that time will plot with the new color, for the current operating session of APWIN. Quitting from APWIN and

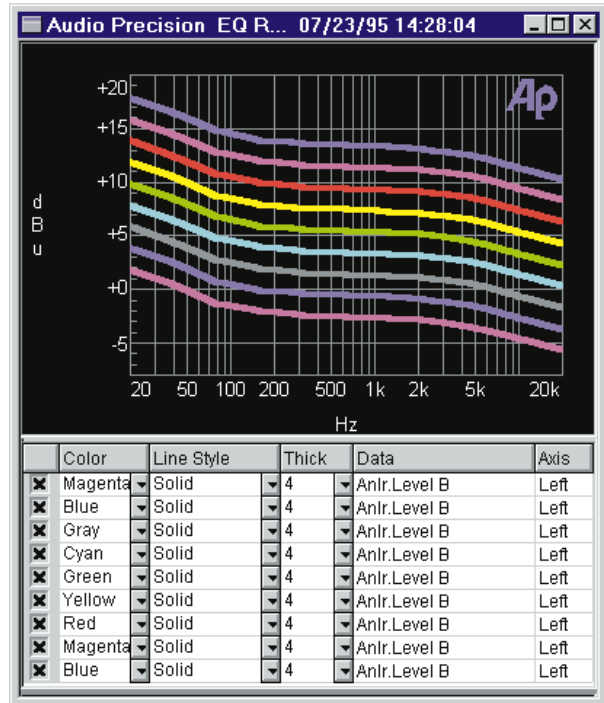



Figure 16-8 Traces Automatically Change Through Color Sequence for Appended or Nested Sweeps


re-starting will result in color assignments for new tests going back to their original default values. If you wish to always start with your personal choice of default colors, save a “default” test with those colors. Open this “default” test every time you start APWIN. Nested sweeps and appended sweeps will plot each trace with a different color (up to six, excluding gray), automatically cycling through the list of available colors.

Colors assigned in the legend are retained through saving a test and subsequently opening the test and re-displaying the data with the  key. Note that colors for each trace in a hard copy print-out (assuming that a color printer is used) may be selected independently from the trace colors on the screen display by use of the File Page Setup command and dialog box.

### **Line Style (Screen Display)**

---

The Line Style column permits selecting solid lines or several combinations of dashes and dots to discriminate between traces. The selected line style will be displayed only when the trace thickness is one unit; thicker traces are always plotted with the Solid line style. If a line style is changed in the legend, new sweeps made after that time will plot with the new line style, for the current operating session of APWIN. Quitting from APWIN and re-starting will result in line style assignments for new tests going back to their original default values.

Line styles assigned in the legend are retained through saving a test and subsequently opening the test and re-displaying the data with the  key. Note that line styles for each trace in a hard copy print-out may be selected independently from the trace line styles on the screen display by use of the File Page Setup command and dialog box.

### **Line Thickness (Screen Display)**

---

The Thick column permits selecting the thickness of the displayed trace from 1 to 30 units. A scroll bar is provided to gain access to this full range of thickness values. The actual thickness obtained for a given selection depends upon the resolution of the computer screen, so some experimentation may be required to obtain the desired appearance. If a thickness value is changed in the legend, new sweeps made after that time will plot with the new thickness, for the current

operating session of APWIN.

Quitting from APWIN and re-starting will result in thickness assignments for new tests going back to their original default values.

Thickness values assigned in the legend are retained through

saving a test and subsequently opening the test and re-displaying the data with the **F7** key. Note that thickness values for each trace in a hard copy print-out may be selected independently from the trace thickness values on the screen display by use of the File Page Setup command and dialog box.

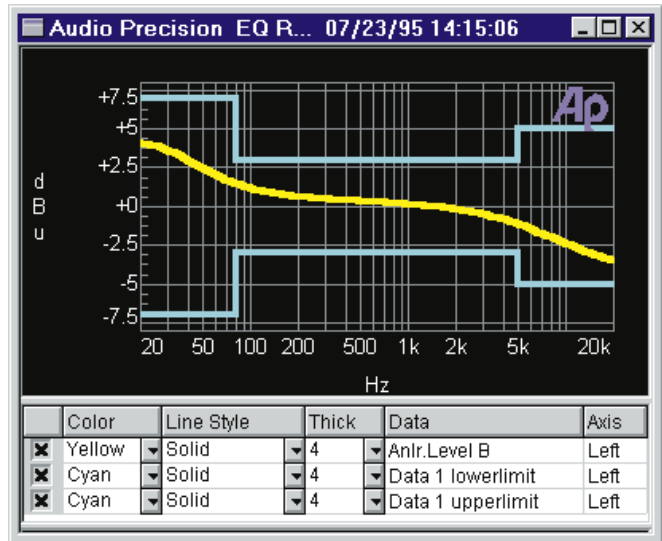


Figure 16-9 Graph and Legend Showing Data Trace and Upper and Lower Limits

## Legend Data Column

The Data column lists the parameter (instrument and meter) plotted by that trace.

## Legend Axis Column

The Axis column tells whether the trace is calibrated by the left or right vertical axis and tic marks.

## Legend Source 2 Column

The Source 2 column, if displayed, shows which instrument setting or meter is being varied as Source 2 in a nested sweep, and what the value of that parameter is for each trace.

## Legend Cursor Columns

---

The two Cursor columns show the values on each trace at their intersections with the cursors.

## Trace Selection


---


The “selected” trace, which is displayed on the graph with greater trace thickness than the other traces, is the trace which the cursors will be attached to and whose values at the cursor positions will be displayed in the “floating” numeric display boxes at the margins of the graph. The selected trace is indicated in the Legend by an inverse video highlight. A different trace may be selected by clicking on its row in the Legend or using the vertical arrow keys. To de-select a trace, click again on the Legend row presently selected. If limit traces are displayed, it is not possible to select them.

## Rows, Legend Size, Column Width

---

When there are more traces on the graph than can be displayed in the Legend at its present size, vertical scroll bars appear so that all rows in the Legend may be examined and selected. As traces are added due to appended sweeps or successive progressions through a nested sweep, the newest trace information appears at the bottom of the Legend and older data is pushed up.



The proportion of the total Graph window occupied by the Legend may be adjusted by dragging the boundary line between legend and graph or legend and Comments Editor up or down. When the mouse cursor is positioned over the boundary, the cursor shape changes from its normal arrow shape to two closely-spaced horizontal parallel bars . With this cursor shape displayed, hold down the left mouse button and move the mouse vertically to drag the Legend height to the desired value, then release the mouse button.

Individual column widths in the Legend may also be changed by positioning the mouse cursor over any column boundary until it changes from its normal arrow shape to two closely-spaced vertical parallel bars . Hold the left mouse button down and drag the column width to the desired value. All columns to the right will

maintain their previous size, so it is best to work from left to right when planning to re-size several columns.




See the Sweep panel discussion for information on how to assign parameters to traces, set the graph top and bottom values and number of divisions and tic marks, etc.

## Keyboard Control of Graph

Zooming by dragging intrinsically requires the use of a mouse. Nearly all other graph-related features are available from keyboard alternatives. For all of the control methods described in this section to function, the Windows focus must be on the Graph window as indicated by the graph title bar being blue rather than gray. If the focus is not on the graph, use the   keystroke to move the focus to the graph.

To select among the graph itself, the graph legend, and the comments editor, use the Tab key. When the graph legend is selected, the top left box (left of “Color” and above the “X” column to turn traces on and off) will be blue. When the Comments editor is selected, a blinking text cursor appears in the editor. If neither of these is true, the graph itself is selected.

### Keyboard Access to “Right Button” Features

When the graph itself is selected, the   keystrokes will cause the same dialog to display which is also accessed by the mouse right button. The up and down arrow keys then move an inverse video selection bar among the rows. Pressing the  key makes the selection and causes the dialog to disappear.

If cursors are displayed and “attached” by selecting a trace (see Legend section below), the horizontal arrow keys move the selected cursor from point to point. To shift keyboard control between Cursor 1 and Cursor 2, press the Shift key while pressing either horizontal arrow key once.

When the Title and Labels dialog is selected, the  or   keys move among the various fields.

## Keyboard Control of Legend

When the graph legend is selected, the horizontal arrow keys move the selection through the columns of the legend. The vertical arrow keys move the selection up and down through the legend row. When the first column is selected, the Space bar toggles between displaying or hiding the selected trace (row). When the Color, Line Style, or Thickness columns are selected, the Alt Down Arrow keystrokes cause the list of selections to be displayed. The Up Arrow and Down Arrow keys then move the selection up and down the list, and the Enter key makes the selection.

## Bar Graphs

A new Bar Graph can be created on the screen by:

clicking on the Bar Graph icon  on the toolbar

or

selecting Panels Bar Graphs from the menu bar

or

from the keyboard, pressing    (Panels Bar Graphs)

By default, each newly-created bar graph will display the Reading meter parameter of the Analog Analyzer. To change to any other measurement or setting, click the Setup button on the bargraph (not visible if the bargraph has been dragged to a small size), or click the



Figure 16-10 Readings Bargraph, Showing Max-min Bar and Target Area

right mouse button anywhere on the bar graph. A Bargraph Setup dialog box will appear. A “browser” control permits selection of a chassis, an instrument, and a reading or setting parameter of that instrument. Units to be displayed are chosen by clicking on the down arrow of either the “Left” or “Right” field, then selecting the desired unit. The values for bargraph end points are then free-form numeric entries into the Left and Right fields. Press



or the (or ) keys to complete the entry. Log calibration may be selected if both end values are positive numbers.

A bargraph may be set to any size desired by dragging any of the corners or edges. The size of the digital display scales with the size of the bargraph.

## Readings Bargraphs

The present value of the parameter measured is displayed by a magenta graph. The Autoscale box may be checked to cause the bargraph end points to automatically be set to values sufficient to keep the largest measured data values on the graph.

## Readings Max/Min

A thin gray stripe in the center of the magenta area shows the minimum and maximum values which have been reached by the

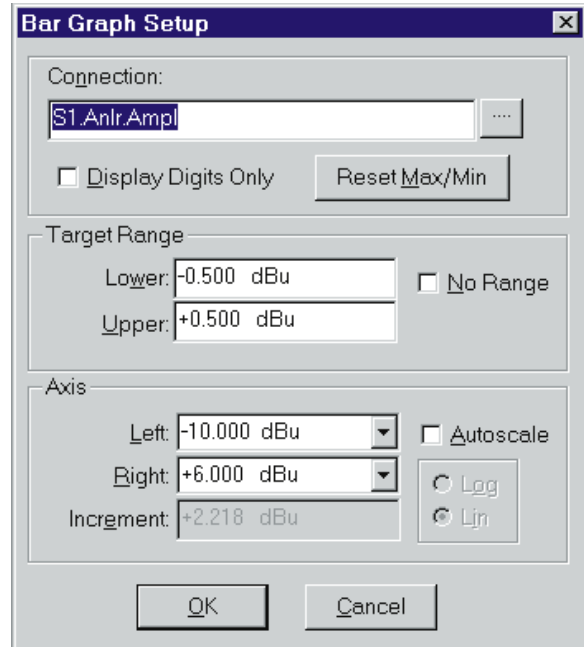


Figure 16-11 Bargraph Setup Dialog, Obtained by Mouse Click on Setup Button or Right Click on Bargraph

parameter since the bargraph was created or reset. The min/max feature may be reset by pressing the **F2** function key when the Windows focus is on the bargraph (title bar is blue) or by clicking the right mouse button on the bargraph, then clicking Reset Max-Min, followed by clicking OK.

## Readings Target Range

The Target Range feature of a bargraph may be used to mark a range as acceptable or to set a target value or range for operator manual adjustments. Note that this visual Target range on a bargraph has nothing to do with Acceptance Limits and Pass/Fail testing during a sweep. To enable the Target feature, un-check the “No Range” checkbox on the Bargraph Setup dialog box which appears when the right mouse button is clicked on a bargraph. Then, enter the desired upper and lower values for the Target Range and click OK.

## Settings Bargraphs

A settings bargraph is created by selecting a setting rather than reading in the Bargraph Setup dialog box. End points, units, and log/lin (if neither end has a zero or negative value) may be selected exactly as in a Readings bargraph.

The Increment field of a Settings bargraph permits selection of the increment size in which a parameter will be varied when the slider on the settings bar graph is dragged the minimal distance or when the left or right arrow keys on the keyboard are pressed. Clicking in the

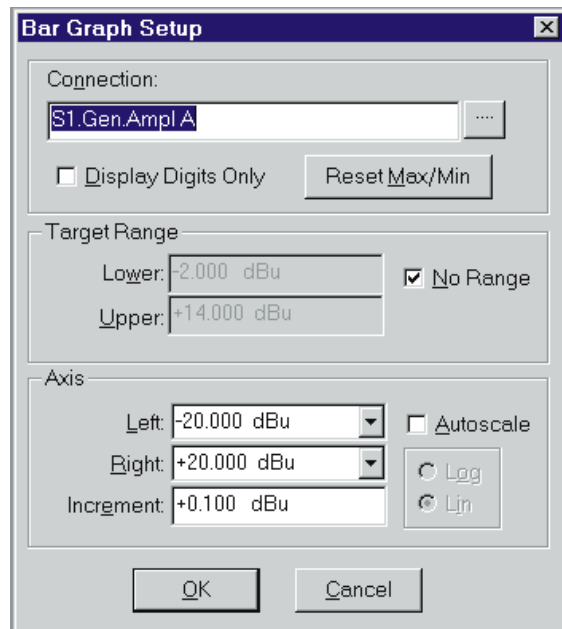


Figure 16-12 Bargraph Setup Dialog Box for a Setting Bargraph



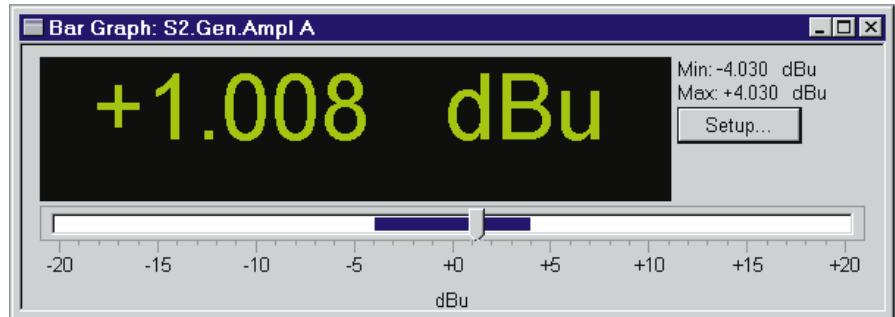


Figure 16-13 Settings Bargraph Showing Max-Min Range

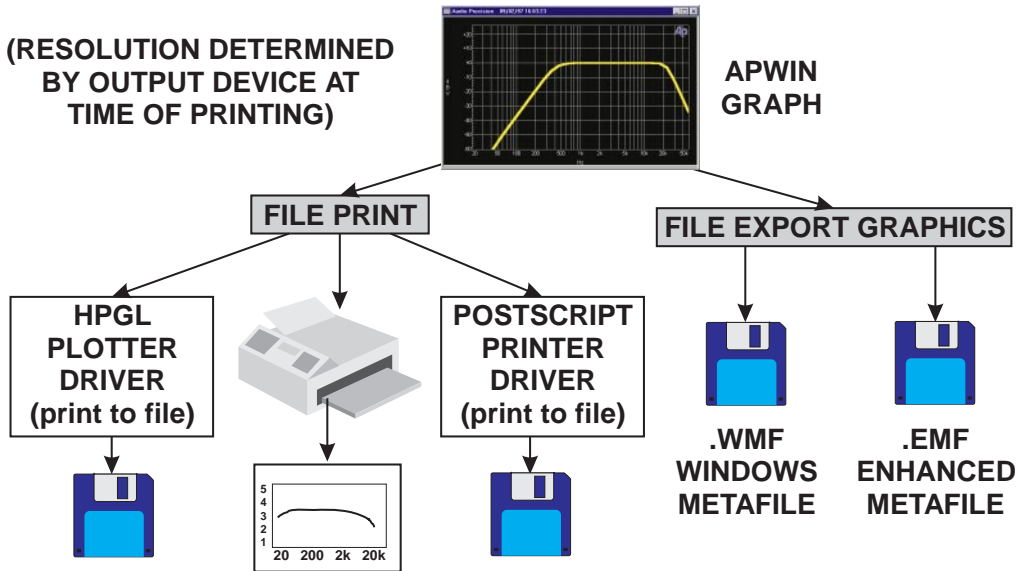
“slider track” to the right or left of the slider control will cause the setting to increase or decrease, respectively, in larger steps. A dark blue coloring of the “slider track” shows the minimum and maximum values to which the parameter has been set.

## Printing

Several different techniques are available for making paper copies of APWIN graphs and, in some cases, other panels. Each has a different set of advantages and disadvantages. The methods are:

- **File Print Graph.** Prints the current graph in vector format to an attached Windows-compatible printer. Resolution of a vector file is dependent only on the eventual output device, not on the graphics resolution of the computer. This technique is fast and flexible, supporting any printer (monochrome or color) which is installed to Windows. Only graphs may be printed by this method, not bargraphs, instrument panels, etc. Graphs may also be “printed” to an HPGL or Postscript file by this technique, with the file then loaded into another program for editing or printing
- **Edit Copy Panel to Clipboard.** Copies the selected APWIN window (graph, bargraph, instrument panel, dialog, etc.) in bitmap format to the Windows clipboard. It may then be pasted into another Windows application for editing or printing. As an exact bitmap copy, graphs have a black background which may not be desired in some printing applications.

### VECTOR GRAPHICS



### BITMAP GRAPHICS

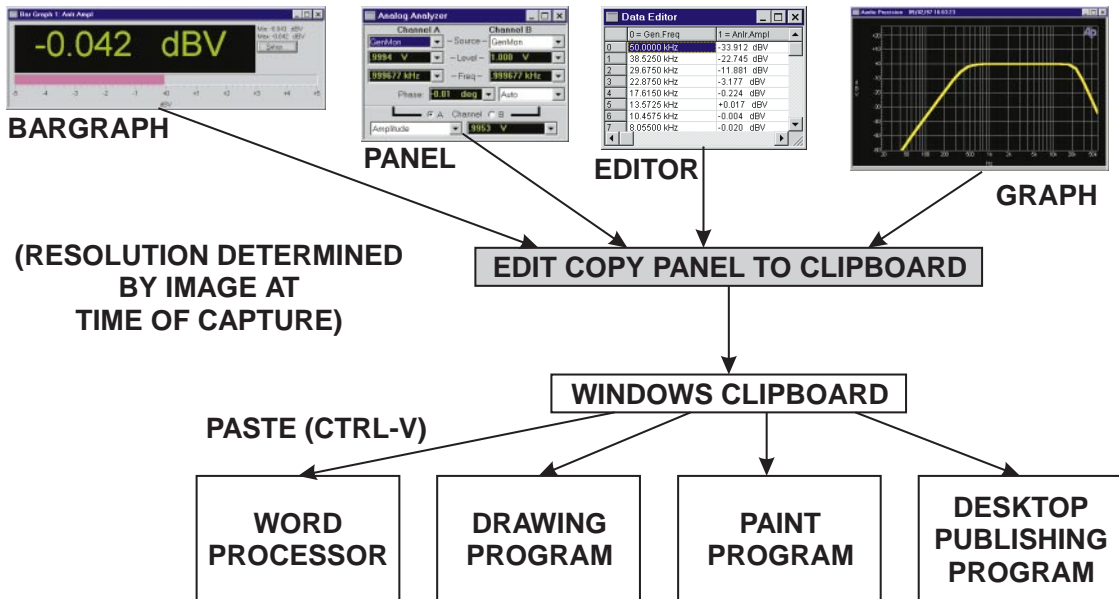


Figure 16-14 Schematic Representation of Various Hardcopy Techniques

- **File Export Graphics.** Creates a Windows Metafile (.wmf) or Enhanced Metafile (.emf) copy of the graph as a disk file. This file may then be loaded into another Windows application for editing or printing. A metafile is a vector format whose resolution is dependent only on the eventual output device, not on the graphics resolution of the computer.

**Print Screen.** The stand Print Screen key on the computer keyboard makes a bitmap copy of the entire screen to the Windows clipboard. From there, the bitmap may be pasted into many other Windows applications for printing or editing.

The remainder of this chapter provides details on the the File Print method. The Edit Copy Panel to Clipboard and File Export Graphics methods are described in more detail in the Menus chapter.

## File Print

---

APWIN graphs may be printed to color or monochrome printers or to an HPGL or Postscript file. The appearance of the graph may be previewed by the File Print Preview command before actually making the print-out. The print-out may be set up by the user to have different line styles or trace widths and different colors from the graph displayed on screen, and the various text elements such as comments, title, legend, and labels may be included or excluded from a print-out independently from their use on screen. These selections are made in the File Page Setup dialog. File Print Setup allows the user to change between different printers.

To print a graph:

select the File Print command under the File Menu

or

click on the Print icon 

or

from the keyboard, press the   key combination

or




from the key board, press    (for File Print).

Clicking on the Print icon causes an immediate print-out according to the present settings of the Page Setup screen, without giving the user the opportunity to make changes. The File Print command (and its keyboard alternatives) displays the standard Windows Print dialog box. The features of this dialog box vary somewhat between Windows 3.11 and Windows 95. Both permit selection of the number of copies and show which printer will be used. The Setup button of the Windows 3.11 Print dialog and the Properties button of the Windows 95 both give access to additional features including Landscape versus Portrait orientation of the graph on the page (which will be overridden if the Portrait/Landscape selection is changed on the Page Setup screen), and the paper tray to be used on multiple-tray printers..

When all selections are made, click on the OK button or press the Enter key. A message will be displayed while APWIN sends the data to Windows and the printer.

### File Print Setup

---

The File Print Setup command permits selection of the printer to be used (or of an HPGL or Postscript driver installed for purposes to “printing” to an HPGL or Postscript file; see the Printing to File section on page 11-27. Print Setup also controls the paper orientation (landscape versus portrait) and the paper size and source on printers that have multiple paper trays. To display the Print Setup dialog box, select Print Setup under the File menu or press the    keys (for File pRint setup).

### File Page Setup

---

The File Page Setup dialog (see Figure 16-15) permits flexible control of the size, margins, line styles, colors, grid line appearance, and various modules of text information on graph print-outs. The settings in the Page Setup dialog can track the on-screen graph settings, or they can be independent of those in the Graph Legend, permitting one scheme to be used for screen display and another scheme for paper print-out. Achieving the desired appearance of a graph print-out often involves an iterative process of viewing the preview results, making changes to the Page Setup dialog, and viewing the results

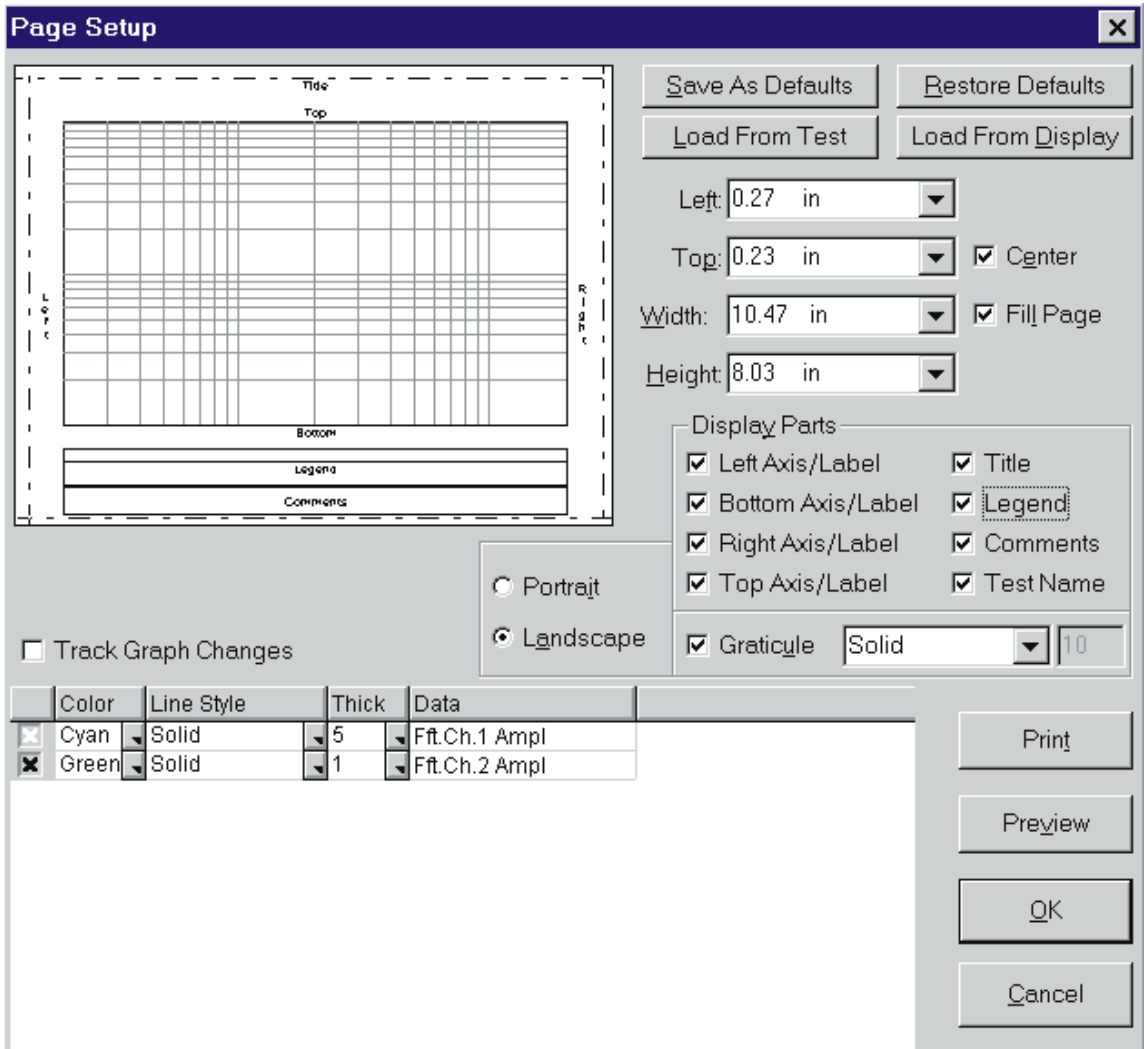


Figure 16-15 Page Setup Screen

again. Thus it is often desirable to start preparation of a graph for printing by selecting Page Setup since you can then jump directly to the Print Preview screen by clicking on the Preview button on the Page Setup screen and can jump back from Print Preview to Page Setup by pressing the Esc key on the keyboard. The actual printing of a hard copy may be initiated by clicking on the Print key of either the Print Preview screen or the Page Setup screen.

To display the Page Setup dialog, click on the Page Setup command under the File menu.





## Landscape vs. Portrait


The top left of the Page Setup dialog window shows a graphic representation of the page in the selected Landscape or Portrait mode, with the size of the graph and its location on the page also represented. The selection between Landscape or Portrait is made by clicking on the appropriate button at the right center of the window.

## Graph Margins and Size

The margins and size of the graph may be set by entering numbers in the Left, Top, Width, and Height boxes at the upper right of the Page Setup dialog. Units may be selected as inches or millimeters by clicking on the down arrow at the right of any of these boxes, with the units for all four fields then changing simultaneously.

Check boxes to the right of the margin and size fields permit automatically centering the graph on the page (Center checkbox) or maximizing the size of the graph on the page (Fill Page checkbox).

The graph may also be stretched or compressed in size and dragged to another location on the page by use of the mouse. When the mouse cursor is positioned over the top or bottom margin of the representation of the graph, the cursor shape changes to a vertical two-headed arrow  and the graph may be stretched or compressed vertically by holding down the left mouse button while moving the mouse (dragging). When the cursor is placed over either the left or right margin, the cursor shape changes to a horizontal two-headed arrow  and the graph may be stretched or compressed horizontally. When the cursor is placed at any of the four corners of the graph, the cursor shape changes to a diagonal two-headed arrow  whose direction depends on the  corner selected, and both height and width can be changed simultaneously by dragging with the mouse. None of these actions can be taken if the Fill Page checkbox has been checked.

The graph may be dragged to another location on the page with the mouse. With the cursor anywhere over the central portion of the graph, the mouse cursor shape changes to an X shape with arrowheads . Holding down the left mouse button while moving the mouse will then drag the graph, at its present size, to a new location. The graph location cannot be changed if either the Center or Fill Page checkboxes have been checked.

## Displaying Text Elements

The Display Parts section of the File Page Setup dialog box controls whether eight specific text elements are printed or not. The individually controllable parts are the **Left Axis/Label**, **Bottom Axis/Label**, **Right Axis/Label**, **Top Axis/Label**, **Title**, **Legend**, **Comments**, and **Test Name**. The **Legend** consists of the information displayed below the screen graph, showing which color and line style or thickness trace plots each data parameter and which vertical axis is the calibration reference for the trace. **Comments** is a free-form text editor and display for additional information of any type. The **Test Name** selection causes the name of the test file (.AT1 or .AT2) to be printed at the bottom of the graph, below the bottom axis/label. **Left Axis/Label** refers to the left-hand Y-axis calibration (Data-1) and any additional label information that may have been added by the user. **Bottom Axis/Label** is the X-axis (usually Source 1) calibration and optional added label. **Right Axis/Label** is the right-hand Y-axis (Data-2) calibration and optional label. The **Top Axis/Label** is purely optional for user-added information. The **Title** appears in the blue Title Bar at the top of the screen-displayed graph and at the top left of a printed graph. Optional data is entered into any of these last five areas via the Title and Labels choice of the right mouse button menu obtained by clicking on the screen display of the graph.

## Graticule Appearance

In addition to being able to turn the graticule (vertical and horizontal grid lines on the graph) on and off via its checkbox, the grid lines may be selected as any of five standard and one custom line styles. Clicking the down arrow at the right end of the field to the right

of the Graticule check box will display a list of dot and dash line styles to be used for the grid lines. The last selection, “Dots per inch”, operates in conjunction with the numeric entry field to its right to permit the user to specify the dot density of the printout.

### Trace Appearance

The controls at the bottom left and center of the Page Setup dialog permit selecting which data traces are to be printed and to select the Color (if a color printer has been selected via the Print Setup menu command), Line Style (solid, dash, dot etc.), and trace Thickness of each. All of these settings are independent from the settings of the same parameters for the computer screen display, as controlled in the Graph Legend. Thus, the screen display can be set for trace colors while the print-out on a monochrome laser or dot matrix printer can use various dash and dot line styles to distinguish between different graphed parameters.

Trace color and line style or width selections may be made individually for a large number of traces such as can result from appended or nested sweeps. If those settings are saved as defaults and a later test made which only uses one or a few traces, the colors and styles will be assigned from the saved defaults in sequence. For example, assume a six-trace page setup is made with the colors in the sequence (top to bottom in the legend on the Page Setup dialog) blue, red, yellow, magenta, green, and cyan. Assume the “Save As Defaults” button is clicked. If another test is now set up or loaded from disk that uses only two traces, the printed colors of those two traces would be blue and red (but could be changed if desired). If still another test is loaded or set up with five traces, the colors automatically assigned would be blue, red, yellow, magenta, and green.

The checkbox at the left end of each row controls whether that trace will be printed.

Clicking the down arrow at the right end of the Color field causes a list of available colors to be displayed, and the desired color can be selected. Note that the Gray color selection will actually result in a black trace on color printers, since gray does not print reliably on all color printers.



Clicking the down arrow at the right of the Line Style box displays a list of available line styles (solid, dash, dot, etc.) which are particularly useful with monochrome printers to distinguish between traces. Note that line styles other than solid will only print at a trace Thickness of one unit; thicker traces are always printed as a solid style.

Clicking the down arrow at the right of the Thickness cell produces a list of available line widths from 1 to 30 units. The vertical scroll bar in this box must be used to gain access to all values. The actual trace thickness on the final printed page will depend on the resolution of the particular printer, so some experimentation may be required to obtain the desired result. As noted above, Thickness values greater than one will print as a solid trace even if some other line style has been selected.

Particularly on lower resolution displays, the width of the Page Setup box may not be sufficient to display the full width of the Data, Color, Line Style, and Thickness columns including the down arrows at the end of the Thickness column. In this case, the mouse cursor can be used at the ends of the column heading cells to drag the columns to a narrower width.

### **Independence of Printed from Screen Graph**

Certain aspects of the printed graph (including which parameters are graphed, graph top and bottom and left and right calibrations, etc.) depend totally on the screen display and cannot be changed from the Page Setup dialog. Other aspects (including trace color, line style or thickness, and whether or not each text element is displayed/printed) can be set independently on the Page Setup dialog from the screen display, if the "Track Graph Changes" checkbox is not checked. Preferred settings for print-out may be saved as defaults to the APWIN.INI file so that they become the user's standard settings. These standard defaults may then optionally be overridden with either the current Graph Legend settings or with the settings stored to disk as part of a test file and re-loaded. If the "Track Graph Changes" box is checked, the color, line style, thickness, etc. controls disappear from Page Setup and the Graph Legend settings are used for printing.

## Save As Defaults

Clicking the “Save As Defaults” button causes all current settings of the Page Setup dialog to be written into the APWIN.INI file. These become the Page Setup dialog settings each time APWIN is started, and may replace any current settings at the user’s option by clicking the “Restore Defaults” button.

## Restore Defaults

Clicking the “Restore Defaults” button replaces all current settings of the Page Setup dialog with the settings from the APWIN.INI file. These would be the settings in use the last time the “Save As Defaults” button was clicked.

## Load From Test


All information from the Page Setup dialog is saved to disk along with all instrument panel settings, test data, etc., in a test file (.AT1 or .AT2 file extension). If the user wishes to print with the Page Setup settings which were in use when a test file was saved, rather than the current defaults from the APWIN.INI file, click the “Load From Test” button.

## Load From Graph

Many parameters may be set independently for the screen display of graphs and paper print-out controlled by Page Setup. These parameters include the trace color, line style or thickness, and whether or not certain text elements are displayed. If the user wishes to replace the current settings of the Page Setup dialog box with the current Graph settings, click on the “Load From Graph” button.

## File Print Preview

---

To see the results of a graph print-out before actually printing, click the Print Preview icon  , select the Print Preview command in the File menu, or from the Page Setup dialog of the File menu, press the Preview button.

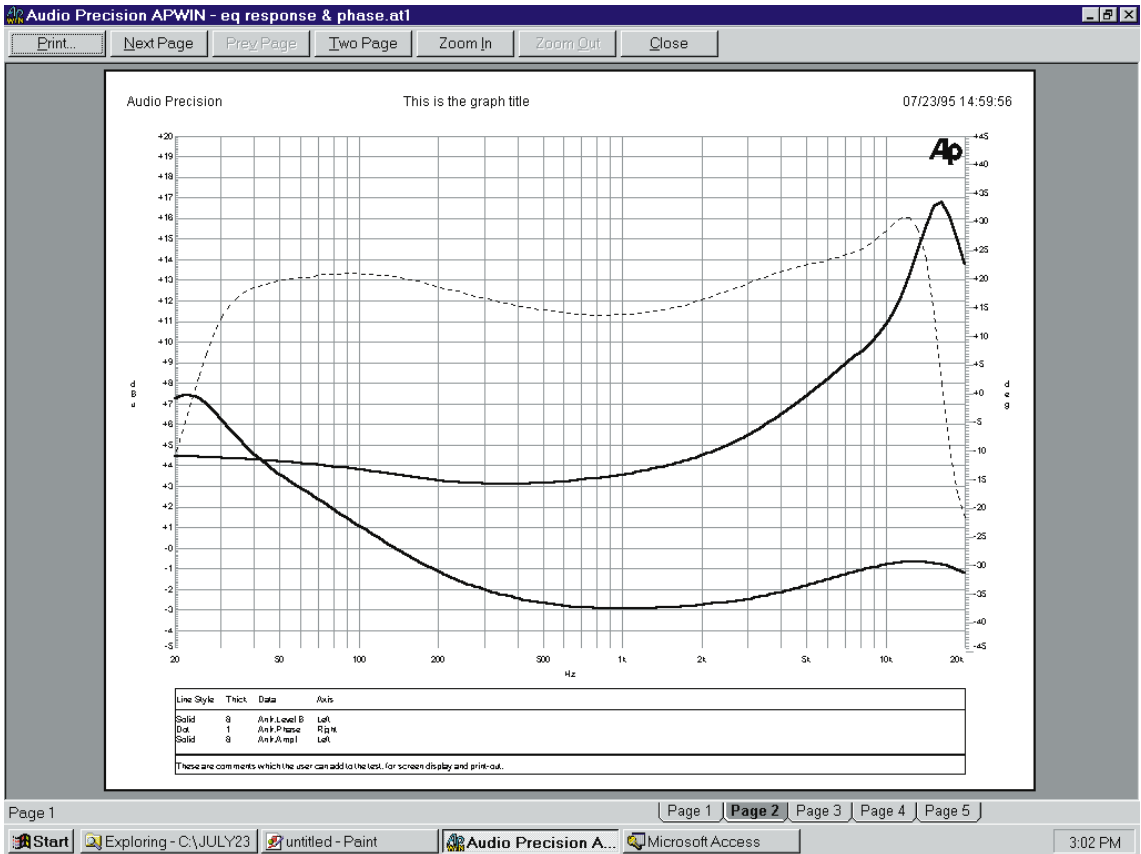


Figure 16-16 Print Preview Screen

If there is a graph on screen, Print Preview displays on the computer screen the page very much as it will appear; see Figure 16-16. The Zoom In and Zoom Out buttons at the top of the screen may be used to look at portions of the page in more detail. When the mouse cursor is positioned over the graph area, the cursor takes on the shape of a magnifying glass; in this condition, clicking the left mouse button once has the same effect as clicking the Zoom In button once. A third click of the left mouse button, when the preview graph has been zoomed to the maximum magnification, returns the view to the original view.

Achieving the desired appearance of a graph print-out often involves an iterative process of viewing the preview results, making changes to the Page Setup dialog, and viewing the results again. **Thus**


***it is often desirable to first select Page Setup, since you can then jump directly to the Print Preview screen by clicking on the Preview button on the Page Setup screen and can jump back from Print Preview to Page Setup by pressing the Esc key on the keyboard.*** The actual printing of a hard copy may be initiated by clicking on the Print key of either the Print Preview screen or the Page Setup screen.

## Printing to a File

---

APWIN and Windows can print a graph to an HPGL or Postscript disk file. Many types of word processing, graphics, and desktop publishing software will import HPGL and Postscript files.

To print a graph to a file, first use the printer installation feature of Windows Control Panel (Windows 3.1) or the Settings Printers command from the Start button of Windows 95 to install a driver for an HPGL plotter or a Postscript printer, as desired. In the “Connections” dialog of the installation, select “File” rather than a real-time output port such as LPT1 or COM2. With this printer selected under APWIN’s Print Setup command, a disk file will result from the APWIN Print command or icon.

Note that an alternative method exists for bringing graphs (and bargraphs, panels, etc.) into other Windows applications. Selecting the desired APWIN panel or graph followed by use of the Edit Copy Panel to Clipboard menu command will copy the selected graph or panel to the Windows clipboard. It may then be pasted into another Windows program such as a word processor or desktop publishing program by use of the Paste () command.

# Limits, Data Editor, and Attached File Editor

Limits are the extreme values of a performance parameter which are considered by the responsible persons to be acceptable. Acceptance test limits (often also referred to as Pass/Fail limits or Go/NoGo limits) are commonly used in quality assurance, manufacturing production test, incoming inspection, and routine maintenance applications.

In APWIN, one or more limit data files (.ADL files) contain the numerical values to be used as limits. These files are commonly created in the APWIN Data Editor and later edited, if necessary, in the Attached File Editor. A single .ADL file may contain both upper and lower limits if desired, and may contain limits for several different parameters if desired. Alternately, a separate .ADL file could be used for each limit. The Limits buttons on the Sweep panel are used to establish cross-reference instructions on which .ADL file (and which column of data within the file) is to be used for each measured parameter in a sweep test, and to gain access (via the Edit button) to the Attached File Editor.

When a test is run with limits attached, several courses of action can be taken if a measurement falls outside limits. The test may be made to halt at the first out-of-limits condition. Results may log into a log file. Failure indications will appear in a tabular listing (Data Editor). If the test is running under control of an APWIN Procedure, the full flexibility of the Basic programming language can be used on failure.

## Using Data Editor

The Data Editor is used to create .ADL files for use as Limit data files for Pass/Fail testing, or .ADS files for use as Sweep Tables with the Table Sweep feature. To open the Data Editor:




click on the Data Editor icon 

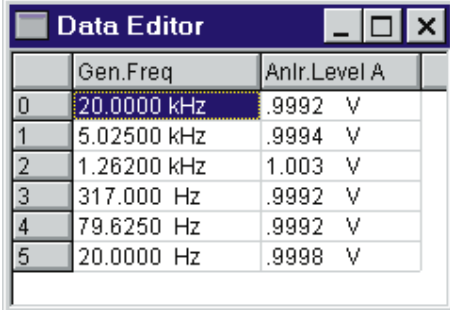
or

select Panels Data Editor

or







from the keyboard press

   (for Panels data editOr).



	Gen.Freq	Anlr.Level A
0	20.0000 kHz	.9992 V
1	5.02500 kHz	.9994 V
2	1.26200 kHz	1.003 V
3	317.000 Hz	.9992 V
4	79.6250 Hz	.9992 V
5	20.0000 Hz	.9998 V

Figure 17-1 Data Editor

The mouse can be used to drag any edge or corner of the Data Editor to bring it to any desired size. The Data Editor can be toggled between a full screen size and the user-dragged size by clicking on the “large/small version” icon or using the   or   keys. The “large/small version” icon for Windows 95 is , and for Windows 3.11 is .

If there is presently data (from a sweep) in memory, you will see a tabular listing of the data. The first column shows the Source 1 values and the remaining columns (if the test was not a nested sweep) the Data 1 through Data 6 values. If the test was a nested sweep, the second column displays the Source 2 values and the remaining columns the Data 1 through Data 6 values.

## Selecting Rows

Individual data rows or groups of rows may be selected for deletion or inserting new rows. To select an individual row, click the button at the left end of the desired row or hold down the Shift key while clicking anywhere in the row. Multiple rows may be selected by individually clicking the button at the left of each desired row, or clicking and dragging down across a range of buttons, or holding down the Shift key while dragging in the cell area across a group of row. The multiple rows need not be adjacent.

## Clearing Data, Adding and Deleting Rows

Click the **right** mouse button anywhere within the Data Editor window. A menu (see Figure 17-2) will display the following choices:

New Data, Insert Row Before, Insert Row After, Add Row to the End, Delete Row. Select the desired action with the left mouse button.

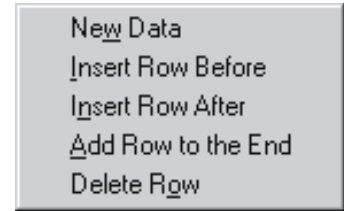


Figure 17-2 Data Editor Right Button Menu

**New Data** clears all the data presently in memory. Do not select this command unless you wish to replace all the data from the last sweep.

**Insert Row Before** is effective only if data is visible and a data row has been selected. In that case, Insert Row Before causes a row to be inserted immediately before the selected row. The row inserted will initially contain values in each cell which duplicate the values of the row that was selected. If multiple rows are selected when Insert Row Before is executed, that number of rows will be inserted.

**Insert Row After** is effective only if data is visible and a data row has been selected by clicking on it. In that case, Insert Row After causes a row to be inserted immediately after the selected row. The row inserted will initially contain values in each cell which duplicate the values of the row that was selected.

**Add Row to the End** will cause an additional row to be added at the bottom of the existing data. The initial contents of each cell will duplicate the values in the row above. If the Data Editor is empty, the Add Row to the End selection will add an initial row with values of one in each cell. The Data Editor will contain a column for each Source and Data parameter defined on the Sweep panel, and the column headings and units will correspond to the parameter and units selected at each set of Source and Data control fields.

A new row can also be added to the end by pressing the down arrow on the keyboard if the last row is selected. The values in each of the new rows will duplicate the values in the last row. If the Data Editor is empty, clicking the left mouse button anywhere within an empty Data Editor and pressing the down arrow will add new rows. The values in each row in this case will be a nominal value “volunteered” by APWIN. To obtain new rows of user-entered data, edit the cells in the last row to the desired values. Then, press the

down arrow, which will duplicate those values in the new rows. This is normally the fastest way to create several rows, since the down arrow can be quickly pressed repeatedly.

**Delete Row** is effective only if data is visible and a data row has been selected by clicking on the button at the left end of the row, or pressing the Shift key while clicking anywhere in a row. In that case, the selected row is deleted. The Delete Row selection of the right mouse button will then delete all selected rows.

## Entering and Modifying Values

A cell may be selected by clicking on the cell with the mouse cursor, or by using the horizontal and vertical arrow keys to move to the desired cell. The selected cell is indicated by blue inverse video. Typing from the keyboard and pressing the **ENTER** key replaces the present contents of the selected cell. Pressing the **F2** key while a cell is selected enables a character-by-character editing mode, permitting replacement of individual digits.

## Viewing Data Graphically

If the Data Editor and the Graph are simultaneously visible on the same page, the graphic representation of the data points is shown simultaneously as each data value is entered or changed (see Figure 17-3). This is particularly useful

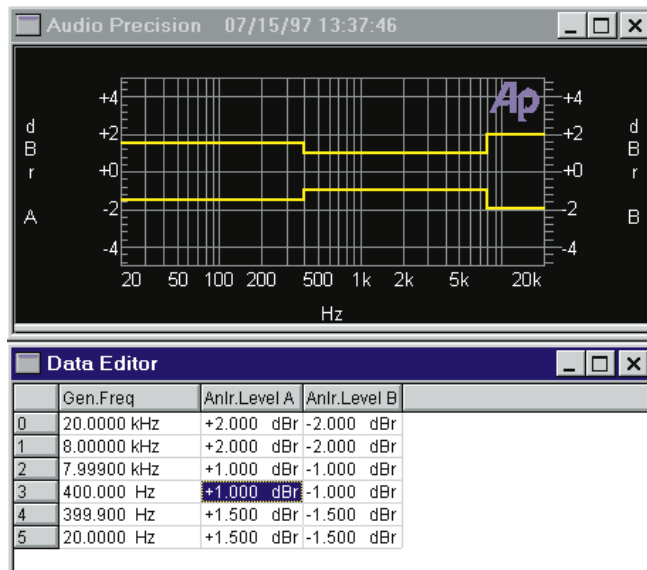


Figure 17-3 Data Editor and Graph Viewed Simultaneously While Creating Limits File



when creating Limits files.

## **Saving Data, Sweep, and Limit Data Files**

---

When the Data Editor is complete for the intended Sweep Table or Limit application, use the Save As command, select Limit (.ADL) or Sweep (.ADS), as appropriate, as the file type, navigate to the directory (folder) where the file is to be stored, and supply an appropriate name for the file. It may then be “attached” (cross-referenced) to the desired test file or files by Opening the test file and using the Table Sweep button at Source 1 or the Limits button at any Data section.

To save a data file, use the File Save As command. Select the “Sweep Data (.ADA)” choice to save the Data Editor contents and certain Sweep Panel information, but not all the instrument panel setups as are saved in a test (.AT1 or .AT2) file. Select the “Limit Data Files (.ADL)” choice to save the Data Editor contents to be used as acceptance limits for pass/fail testing. Select the “Sweep Table (.ADS)” choice to create a file to control the Source 1 parameter when Table Sweep mode is selected on the Sweep panel.

## **Exporting ASCII Data**

---

To save the Data Editor contents in a format compatible with spreadsheets, word processors, and most other computer programs, select the File Export menu command. An Export file (.ADX) is an ASCII, comma-delimited file containing only the Data Editor contents including column headings.

## **Creating Limits**

---

A limit file is an APWIN data file (.ADL file type) used as a standard for comparison of measurements in order to make Pass/Fail decisions. The Source 1 column of the data file must correspond to the Source 1 parameter of the test with which it will be used. For example, a frequency-swept test will have generator or FFT frequency at Source 1 and any limit files used with this test must also have Frequency at Source 1.

## Create Data File Columns

For each limit, there must be a data file column heading and numbers corresponding to the type of measurement to which it will be compared. The limit (data) file need not be created with the exact same units, but they must be in the same “domain”. For example, a data file stored with dBV units may be used with a test which plots amplitude in Volts, dBu, dB<sub>r</sub>, or any other analog amplitude units, but a data file stored with % units could not be used with a measurement in dBV or degrees.

Upper and lower limits, if both are used and are to be stored in the same data file, must have separate columns in the data file. Alternately, two different data files could be used as upper and lower limits.

To create the column structure for a limit data file, set up the Sweep Panel by selecting as Data 1, Data 2, etc. the parameters and units that are to have limits. Each Data n selection will produce one column in the data file, following the first column, which contains the Source 1 parameter.

Use two Data parameters for a measurement that will have both upper and lower limits if it is desired to have both limits in the same data file. One of these columns may then be used for upper limit values and the other for lower limit values.

Only one Data parameter needs to be selected for a measurement that will have only one limit (for example, distortion and noise measurements normally have only an upper limit).

If two measurements are to have exactly the same limit values (often the case for the two channels of a stereo device), additional columns need not be created for each since a single column in the resulting Data file may be used as a limit for multiple measurements. When one channel of a stereo device has been selected as Data 1 and the desired limit files connected via the Data 1 Limits button, checking the Stereo Sweep checkbox at the bottom of the sweep panel automatically selects the other stereo channel as Data 3 and automatically connects the same limits to Data 3 that have been

specified for Data 1. See the Stereo Sweeps section in the Sweeps chapter for more details.

## Create Data File Rows

The data file used as a limit must have a row for each “break point” in the defined limit. If the limit values are to be constant across the entire sweep range, a two-row data file is sufficient. If the limit value is to change values at a single Source-1 value (step up or down at a specific value), there must be two rows for each segment in the Limits file. For example, if the limit is to be 0.02% from 20 Hz to 100 Hz, 0.01% from 100 Hz to 10 kHz, and 0.05% from 10 kHz to 20 kHz, three segments must be defined which will require a six-row data file. Values would then be typed in to create a Data file as follows:



<b>Gen. Freq</b>	<b>Anlr THD Pct</b>
20 Hz	0.02 %
99.999 Hz	0.02 %
100 Hz	0.01 %
9.9999 kHz	0.01 %
10 kHz	0.05 %
20 kHz	0.05 %


New rows in the Data Editor may be created by pressing the down arrow when the cursor is in the last data row, or by clicking the right mouse button on the Data Editor and selecting the “Add Row to the End” choice.

Limits do not necessarily have to extend across the full Source 1 range swept by the test. For example, it might be desired to sweep from 20 Hz to 20 kHz but to only apply Pass/Fail limits between 50 Hz and 10 kHz, with the data outside that range measured and graphed only for informational purposes.

When editing is complete, use the Save As command, click the down arrow at the end of the “Save as type” box, and select “Limit Data files (\*.ADL)”. Click in the File Name box and type in the name you wish to assign to this limit file, and click the Save button.

## Attaching Limit Data Files to Tests

A cross-reference must be established in order for a test file (.AT1 or .AT2 file type) to use the proper limits file or files (.ADL tile type) as limits during a sweep. This cross-reference is established by use of the “Limits . . .” buttons in the Data 1 and Data 2 areas of the Sweep Panel and the “Limits 3 . . .”, “Limits 4 . . .”, “Limits 5 . . .”, and “Limits 6 . . .” buttons on the large form of the Sweep Panel for Data 3 through Data 6. The Sweep Panel may be switched between large and small forms by clicking on the central of the three icons at the right of the Title Bar of the Sweep Panel, or press the **CTRL**  or **CTRL**  keys.

Clicking on any of the “Limits” buttons displays a dialog box (see Figure 17-4) which permits specifying the file and the column of data within that file to be used for Upper and Lower limits. It is not necessary to specify both Upper and Lower limits; either may be left as “None”. Clicking on the ellipsis button  at the right of the File box will display a browser box which permits navigating to the disk directory (folder) where the .ADL limits file is located. Double-click on the desired file name, or single-click and then click on the Open button. If the Upper and Lower limit values are stored in the same data file, check the “Upper and Lower Limits are from Same File” box and the file name will be automatically selected in the other File box.

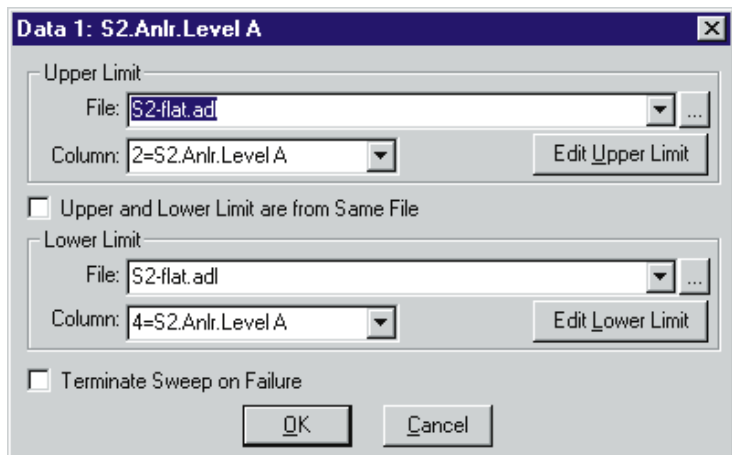


Figure 17-4 Limits Dialog, Obtained via Limits Button for each Data Parameter. Note Edit Upper Limit and Edit Lower Limit Buttons to Display Attached File Editor

When the file has been selected, clicking on the down arrow at the right of the Column box produces a list of the column head labels in that file. The desired column can be selected by clicking on it. If you need to view the limit file contents in order to know which column to select, click on the Edit Upper Limit or Edit Lower Limit button to bring up the Attached File Editor.

## Attached File Editor

The Attached File Editor is similar in format to the Data Editor, but is intended only for viewing and modifying existing Limit files and Sweep Tables. The Attached File Editor is accessible only by clicking the “Edit” button in the Table Sweep dialog box or the “Edit Upper Limit” or “Edit Lower Limit” button in the Limit dialog box. Clicking the Edit button will automatically load the attached limit file or sweep table. The column corresponding to the Edit button will be marked with two asterisks (\*\*) in the column head. For example, if the Edit button in the Lower Limit section of the Data 2 Limits dialog is clicked, \*\* will appear in the heading of the Data 2 Lower Limit. Exact numerical values may be determined from the tabular display and values may be changed or added exactly as described above in the Data Editor sections.

If one or more values are changed in the Attached File Editor, a dialog will be displayed when the Attached File Editor is closed, warning the user that the modifications will have no effect unless the file is saved. If the “No” option is chosen, the changes will be lost and the file will not be modified. If the “Yes” option is chosen, the old version of the file will be over-written with the new version. There is no provision for saving an edited file in the Attached File Editor to a new name.

### Actions Upon Failure

**Sweep Halt:** The user can choose to have a sweep test halt immediately if a data point is taken outside the acceptable limits by checking the “Terminate Sweep on Failure” checkbox on the dialog which appears when the Limits button is clicked. If this box is not checked, the sweep will continue to its conclusion independently of whether data points are outside limits.

**Tabular Display Codes:** When Tabular display is checked on the large version of the Sweep panel, or when the Data Editor is used to examine data, measured points above the upper limit are indicated with an “L” and measurements below the lower limit are indicated with a “U”.

**Log File:** When the Log Errors feature of the Utilities Configuration menu is enabled, two different actions may be taken related to measurements failing limits. The “Pass/Fail Message” feature causes a text message to be written into the log file, telling whether or not all points passed limits or at least one point failed. The message also summarizes the number of failures of the upper and lower limits, and may include the test name and the date and time the test ran. The “Log Sweep Data” feature, with either “All” or “Failed Only” options selected, will write the actual measured values into the log file with an indication of whether the upper or lower limit was failed and the value of the limit which was failed. An example of a log file is shown below:

```
C:\Program Files\APWIN\SAMPLES\sample.at1
Sample Response Test w/Limits 07/09/95 11:10:13
FAILED : Lower Limit 3, Upper Limit 4, Timeouts 0
Gen.Freq(Hz)      Anlr.Level A(dBr)
20000             5.000000(>2)
10022.5           4.200000(>2)
5025              2.100000(>1)
2517.5           1.800000(>1)
79.625           -1.800000(<.5)
39.9             -2.100000(<.5)
20              -2.700000(<.5)
```

The first line is the test file name and path. The second line is the Graph Title, date, and time. The third line is the “Pass/Fail Message” which summarizes failures. The “Failed Only” option was in use, so the remaining lines consist of the points which were outside limits. The points below 2517.5 Hz and above 79.625 Hz were within limits and thus are not shown. The “greater than” (>) and “less than” (<) symbols and numbers at the end of each row show where the value was above the upper limit or below the lower limit, and the value of the limit itself.

For more information on log files, see the Log Files section of the Menus chapter, starting on page 3-39.





## Compute Menu

The Compute Menu consists of commands to operate a number of algorithms which perform computations on the data set resulting from a sweep. These various computational utilities can “push” data curves up or down the graph so as to go through a specific data value at a specific horizontal point, invert a curve, fit a straight line to a data set and subtract all data values from that straight line, compute the difference between the data in memory and the data in a user-specified data file on disk, and a number of other useful operations. The Compute Menu consists of the following commands: Normalize, Invert, Smooth, Linearity, Center, Delta, 2-Sigma, Average, Minimum, Maximum, Equalize, Clear All.

### General Operation, Computational Utilities

Selecting any Compute command except Clear All will display a dialog box generally similar to the example shown in the figure. A column of six check boxes labeled Data 1 through Data 6 is common to all compute utilities. Checking the box next to any Data trace selects that trace for input to the computational algorithm. If the error message “No Compute Performed” is received when a computation is attempted, it often is because no box was checked to select the data to be computed. The computed results will replace the original data, in the same trace. Several traces can be checked if desired and all will be computed in one operation of the Compute (or Compute and Close) button if the data in all traces is compatible with the

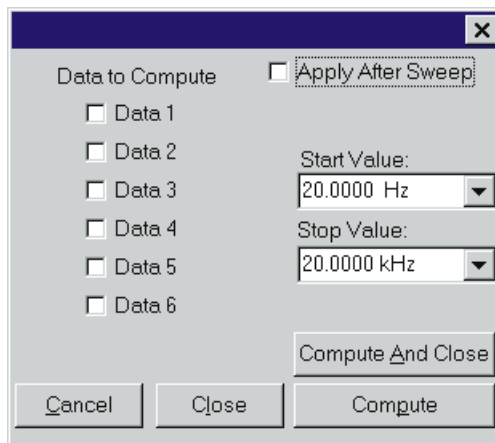


Figure 18-1 Typical Compute Dialog

particular computation. Any data traces not checked will not be computed.

If it is desired to retain the original data and view both original and computed data, the test should initially be set up to graph the same parameter on two traces such as Data 1 and Data 3, for example. Then, the Compute dialog checkbox for Data 3 can be checked and the Compute button clicked, resulting in computed results as Data 3 with the original data still in Data 1.

When the box or boxes have been checked to select the Data sets which are to be computed, the computation can be triggered by clicking on either the Compute or the Compute and Close button. The Compute button triggers a computation and leaves the Compute dialog box visible for additional operations, while the Compute and Close button triggers the computation and removes the dialog box. Several different computations may be manually made in sequence by selecting the same trace in different Compute dialogs and operating the Compute or Compute and Close button. Different computations may also be automatically “chained”; see below. The Close button may be used at any time to remove the Compute dialog from view while retaining any changes which have been made. The Cancel box will remove the dialog from view without retaining changes made since it was opened.

It is also possible to set up an automatic computation or sequence of computations each time a sweep is run by checking the “Apply After Sweep” checkbox at the top of a dialog or several dialogs. When this box is checked and the Close button is clicked, each time a sweep is run the selected computation will automatically take place following the last data point of the sweep. If upper and/or lower limits have been attached to a test and the “Apply After Sweep” box is checked, the data will not be compared to limits until after the computation is made. Computations can be chained in any sequence desired. First, execute the Compute Clear All button to clear all previous settings. Then, set up each desired compute function in the intended sequence of operation, checking the “Apply After Sweep” box and clicking the Close button. When the test is operated, it will perform each compute operation in the sequence in which they were set up.

Many of the Compute utilities have one or two additional numeric fields to define which point or range of the input data will be used as a reference for the computation.

## Compute Normalize

The Compute Normalize command moves a trace (data set) up or down the graph so that the trace goes through a user-specified Y-axis value at a user-specified X-axis value. A typical use for Compute Normalize is to convert actual measured data from a “golden unit”, such as a reference loudspeaker, into upper and lower limit curves for use in repetitive production testing. For example, a test result could be pushed upwards by 3 dB and saved as an upper limit, then pushed downwards 3 dB from the original value and saved as a lower limit.

The Compute Normalize dialog includes two numeric entry fields: Horizontal Value and Target Value. The Horizontal Value field is used to specify at which point across the X-axis the Target Value will be achieved in the computed result. The Target Value field is used to specify the desired new Y-axis value at the specified Horizontal Value. Both fields have the full range of units available which apply to the data set being computed.

If two or more Data boxes are checked in the Compute Normalize dialog before the Compute button is clicked, all checked traces will be normalized to the entered Target Value at the entered Horizontal Value if all are in the same “domain”; for example, if all are amplitude or level readings. If any of the traces checked contain incompatible data (for example, level data on the first selected trace and phase data in degrees on another trace), an error message

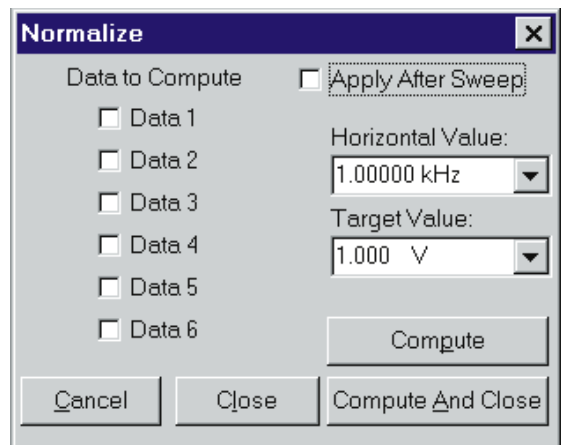


Figure 18-2 Compute Normalize Dialog

will be received when the computation is attempted since a single Target Value cannot work for two domains.

## Compute Invert

Compute Invert turns a curve upside down. A typical application of Compute Invert is to turn over the result of a frequency response measurement on the test system itself, including connecting cables with their response variations, in order to use the result as an Equalization curve and thus correct for systematic response variations.

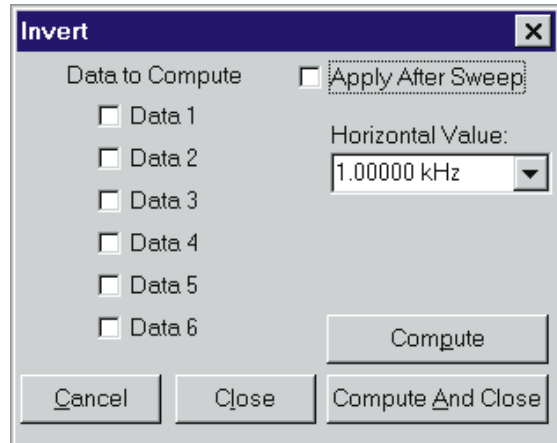


Figure 18-3 Compute Invert Dialog

The Compute Invert dialog has a single numeric entry field for Horizontal Value. The data inversion will be performed around the value at the specified Horizontal Value as a pivot point.

## Compute Smooth

The Compute Smooth utility performs a running 3-point smoothing on the selected data set. Smoothing is sometimes used for loudspeaker sinewave frequency response measurements, based on the assumption that sharp peaks and dips are probably the effects of room reflections and cancellations rather than actual speaker response variations and are thus better removed.

The Compute Smooth dialog contains a numeric entry field for “Passes” and an “Auto” checkbox. One pass means going through the data set once, replacing each measured value with the computed average of that point and the two adjacent horizontal measurement values. If the entry in the “Passes” field is greater than one, the

smoothing process is repeated the specified number of times. Checking the “Auto” box causes the Passes field to gray and the software selects a number of passes based upon the number of measurement points in the data.

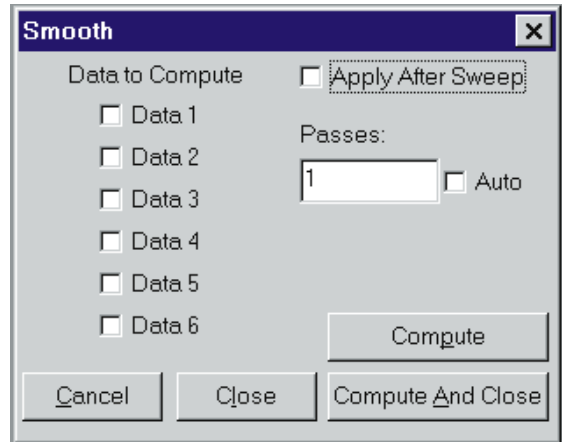


Figure 18-4 Compute Smooth Dialog

## Compute Linearity

The Compute Linearity utility performs two mathematical operations in sequence. First, the best straight line is fitted to the data values within a user-specified range of data, using the least squares method. Second, every data value in the set is subtracted from the straight line just computed. The result is a graph of deviation from perfect linearity. The Compute Linearity utility is normally used on measurement data such as an output-amplitude-vs.-input-amplitude sweep of an A/D or D/A converter, or on the output-input phase shift through a device under test such as a loudspeaker.

The Compute Linearity dialog has two numeric entry fields, labeled Start Value and Stop Value. Entries into these fields specify the horizontal range of data values to which the straight line will be fitted. Especially in the case of an output amplitude

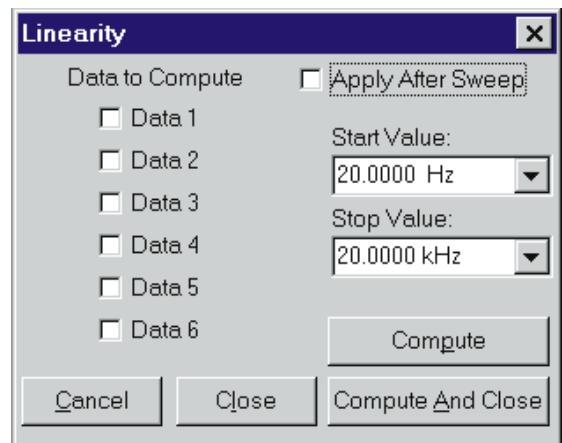


Figure 18-5 Compute Linearity Dialog

versus input amplitude sweep across a wide dynamic range, measurements at the higher amplitudes may be compressed or clipped and measurements at the lower amplitudes may be noise-limited. The Start and Stop Value fields permit specifying a reference range where these effects are likely to be minimal. The Start and Stop Values are used only in the straight line fitting portion of the process; after the line is derived, every data point—inside and outside that range—is subtracted from the straight line.

The results of Compute Linearity, showing only deviations from perfect linearity, will normally span a much smaller Y-axis range of values than the original measurement values. The Optimize commands available by clicking the right mouse button on the graph will select new graph top and bottom values in order to re-display the data with the best possible resolution.

## Compute Center

The Compute Center utility can only be used with data sets that have both upper and lower limit files attached. See the Limits chapter for more information on limits. The purpose of the Compute Center utility is for cases where the shape of a device's frequency response curve is much more important than the exact absolute amplitude of the measurements. This is commonly a case in microphones, for example, where small variations in output level (sensitivity) are easily compensated for in actual usage by adjustment of a gain control. The Compute Center utility will compare all measurement values within a specified horizontal range to the upper and lower limits and then push the data up or down the Y-axis to

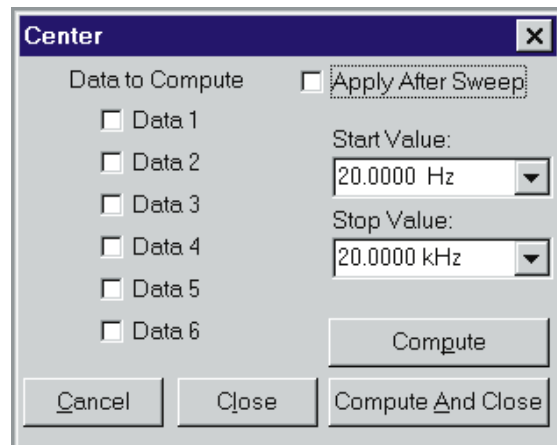


Figure 18-6 Compute Center Dialog

best center it between the limits.

The Compute Center dialog includes Start Value and Stop Value numeric entry fields. These are used to enter horizontal axis values between which the measured data and limits are compared. Data outside that range will be not be compared to the limits for purposes of moving the data vertically for best fit within the limits. However, all data within the range defined by the limit files themselves will then be used by APWIN software for the PASS/FAIL decision making following a Compute Center if the “Apply After Sweep” box is checked.

## Compute Delta

The Compute Delta utility will subtract the data values in a user-specified data file (.ADA, .ADL, .ADS, or .ADQ) on disk from the measurement data presently in memory, or will subtract the data in one column in memory from the data in another column. This feature can produce after-the-fact equalization when the disk file is an EQ curve, can subtract out an offset if the disk file is a constant value, or can subtract one channel of stereo data from another to display inter-channel difference

The Delta Source must first be selected in the field near the bottom of the dialog. If the source is to be a data file previously saved to disk, click on the ellipsis button at the right end of the field and use the

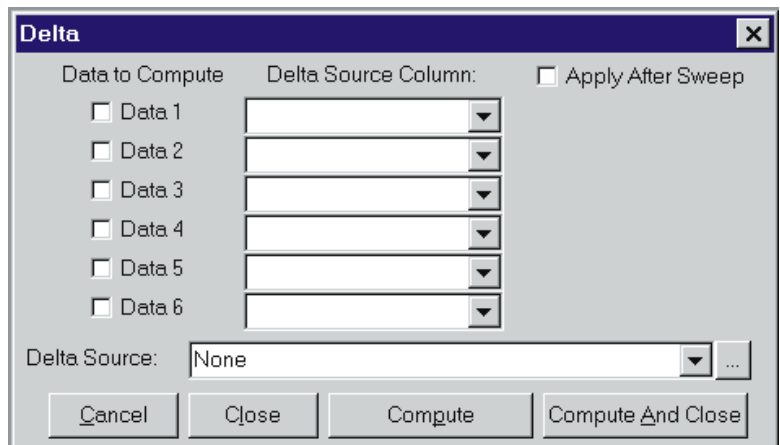


Figure 18-7 Compute Delta Dialog

browser to navigate to the desired directory and file name. Only data files (.ADA, .ADL, .ADS, or .ADQ) may be selected. If data in a test file (.AT2) is to be used as a Delta Source, load the test file, use the File Save As Data command to save the sweep data contents into a data file, and select that new data file as the Delta Source.

It is also possible to use the measurement results data presently in computer memory as the Delta Source. Click on the down arrow (not the ellipsis button) at the right end of the Delta Source field and select “Sweep Data” instead of a data file name. This can be useful in order to subtract (for example) right channel response from left channel response of a stereo device to obtain a graph of the difference in response of the channels.

When a Delta Source has been selected, it is then necessary to select a particular column of data in the file or computer memory which is to be subtracted from the measurement data. The Delta Source Column field at the right of each checkbox permits selection of any of the columns in the specified Delta Source. Clicking on the Compute or Compute and Close buttons will then replace the data of the selected trace with the difference between the original trace values and the selected Delta Source Column of data.

## Compute 2-Sigma

The Compute 2 Sigma utility is used with a series of wow and flutter measurements versus time (chart recorder style) when it is desired to have a single number representing the wow and flutter performance. The 2-Sigma value for a series of wow and flutter measurements is defined as the magnitude value

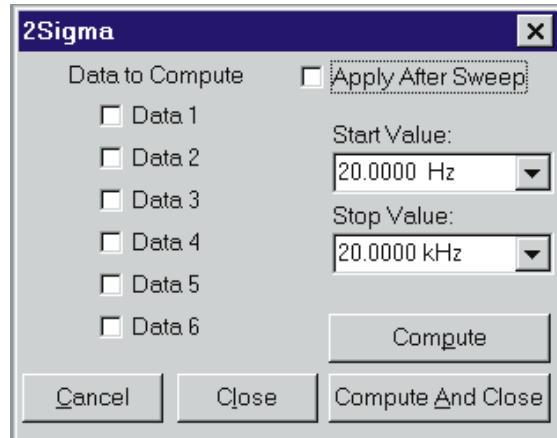


Figure 18-8 Compute 2-Sigma Dialog



that is exceeded exactly five percent of the time. This is based on the fact that five percent of the values in a Gaussian distribution will exceed the two standard deviations (two sigma) value above the center. In fact, there is no reason to suppose that a series of wow and flutter measurements will have a Gaussian distribution, but the 2-Sigma standard for wow and flutter is defined in this fashion.

The Compute 2 Sigma dialog includes numeric entry fields for Start and Stop values. Entries in these fields specify the horizontal range of data that will be included in the calculation. Particularly in the case of wow and flutter measurements, an initial large transient may exist for several seconds due to the long time constants of wow and flutter detectors. The Start and Stop fields may be set to exclude such an initial transient.

Note that the Compute 2-Sigma utility replaces every original measured value with the computed result, so a horizontal line on the graph results and the Data Editor view of the data will show every value to be identical.

## Compute Average

The Compute Average utility will compute the average value of all the points in the data set between the Start Value and the Stop Value and replace every measured value with that computed average. Data values outside the range will not be included in the computation of average value, but all values in the entire data set will be replaced with the value computed.

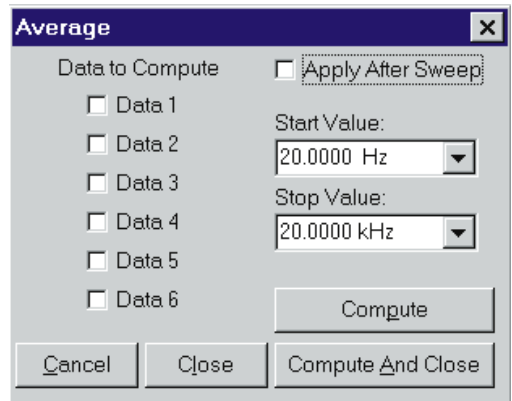


Figure 18-9 Compute Average Dialog

## Compute Minimum and Compute Maximum

The Compute Minimum and Compute Maximum utilities will find the minimum or maximum value in a sweep (data set) and replace every measured value with that value. The Start Value and Stop Value fields may be used to define a horizontal range of data to be searched. Data values outside the range will not be included in the search for minimum or maximum value, but all values in the entire data set will be replaced with the value found.

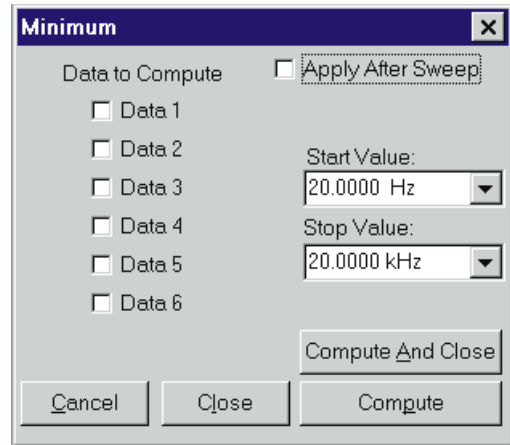


Figure 18-10 Compute Minimum Dialog;  
Compute Maximum is Identical

## Compute Equalize

The Compute Equalize function multiplies the value of every measurement point in the selected data set by the value at the corresponding horizontal (frequency) point in a specified .ADQ file on disk. Compute Equalize thus has a similar result, following a sweep, to the EQ Sine “waveform” available for the analog and digital generators, except that the EQ Sine functions make real-time changes in generator output levels

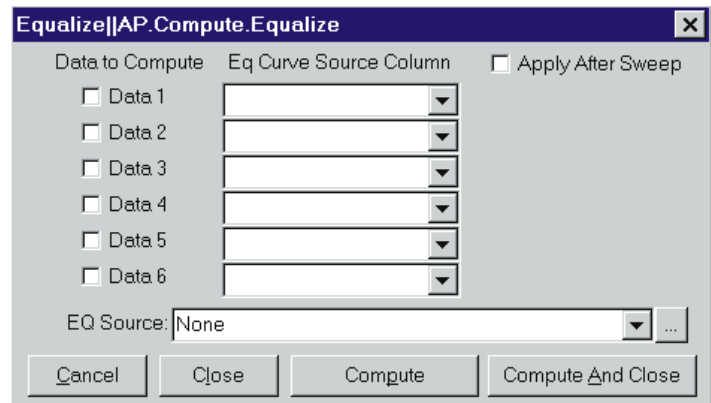


Figure 18-11 Compute Equalize Dialog

while Compute EQ is a one-time computation normally made following a sweep test. Compute Equalize interpolates between data points in the specified .ADQ file as necessary to produce values precisely at the data point horizontal values.

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## **Compute Clear All**





The Clear All command of the Compute menu clears all settings of all Compute functions to their normal default settings.



# Automating Tests

Automation speeds up testing activities and reduces or eliminates the normal boredom when humans perform repetitive actions. There is a significant degree of automation in simply saving a test file and later opening it to recreate the same setup. Much greater degrees of automation are available in APWIN through Procedures (.APB files), sometimes also referred to as Scripts or Macros.



Procedures are computer programs that define sequences of operations. A procedure may load test files from disk and may make changes directly to any field of any panel. Procedures may run linearly from top to bottom or may branch to different sections or to other procedures, depending upon measurement results or operator input from keyboard or mouse. Procedures may display menus and dialogs to the users, with a variety of methods including buttons, checkboxes, and text entry areas for the user to enter data or control the program flow. Procedures may send graphs or tabular data to printers, may save information to disk, and may communicate with other Windows programs. Procedures are normally written in APWIN Basic, furnished as a standard part of APWIN. APWIN Basic is compatible with Microsoft Visual Basic for Applications. Since APWIN Basic complies with the Microsoft OLE (Object Linking and Embedding) specification, it is possible to control System Two or System One from other Windows programs such as a spreadsheet or word processing program.




An existing procedure may be loaded from disk by the File Open Procedure command. The resulting dialog box permits navigation among directories (folders), displaying the procedure (.APB) files in the selected directory. When an .APB file is opened, the program (listing) will be shown in the Procedure Editor if the Procedure Editor is visible. Once loaded, the procedure may be executed by the Run Procedure  icon on the Procedure Toolbar or from the Menu by the Procedure Run command (  ).

Changes may be made to the procedure in the Procedure Editor. Trouble-shooting of a procedure that is not functioning satisfactorily may be aided with several de-bugging tools. These include break points, single-step operation, and evaluation of expressions. User dialogs may be created which include buttons, prompting messages, and user input of text and numbers. For more details on de-bugging and editing procedures, see *Getting Started with Procedures*, the APWIN Basic Programmers Manual, or *Help APWIN Basic Editor*.

New procedures, or additions to existing procedures, may be generated by two different techniques. One method, suitable for those with some experience with programming techniques and knowledge of the specific syntax and commands of APWIN Basic or other forms of Visual Basic, is by typing and modifying text in the Procedure Editor. The second method, suitable even for users with little or no experience in programming or APWIN Basic, is via the LEARN mode (Procedure Learn Mode menu command). Starting LEARN mode causes each ensuing user mouse click and keyboard entry to write a line of APWIN Basic code into the Procedure Editor. Simple procedures may be completely generated in LEARN mode. More sophisticated procedures with branching, calling of sub-procedures, processing of data results, etc., can have their core created in LEARN mode but will typically require further commands to be added in the Procedure Editor. For more details on these topics, see *Getting Started with Procedures*, the APWIN Basic Programmers Manual, or *Help APWIN Basic Editor*.

## LEARN Mode

LEARN mode is started with by clicking on the Learn icon  on the Learn toolbar , or by the Procedure Learn Mode menu command (Alt R R) or Utilities Learn Mode. Once Learn mode has been started, every user action will result in one or more lines of program code written into the Procedure Editor until Learn mode is halted. If a procedure has already been loaded into the Procedure Editor, the commands created by Learn mode will be inserted at the cursor position in the Procedure Editor. If no procedure has been loaded, the Procedure Editor will be opened with a new (blank) procedure ready for recording of the Learn mode commands. To stop

Learn mode, click on the Stop Learning icon  or use the Procedure Learn Mode or Utilities Learn Mode command again to toggle Learn Mode off. To temporarily suspend the learning of commands, hold down the   keys while clicking the mouse to make changes which will not be learned.

For a Learn Mode example, assume the following list of user actions:

- Click on Start Learn mode icon.
- Click on New Test icon.
- Click on analog generator On/Off control.
- Click on analog analyzer Ch A input and select Gen Monitor instead of XLR Bal.
- Click on page 2 tab.

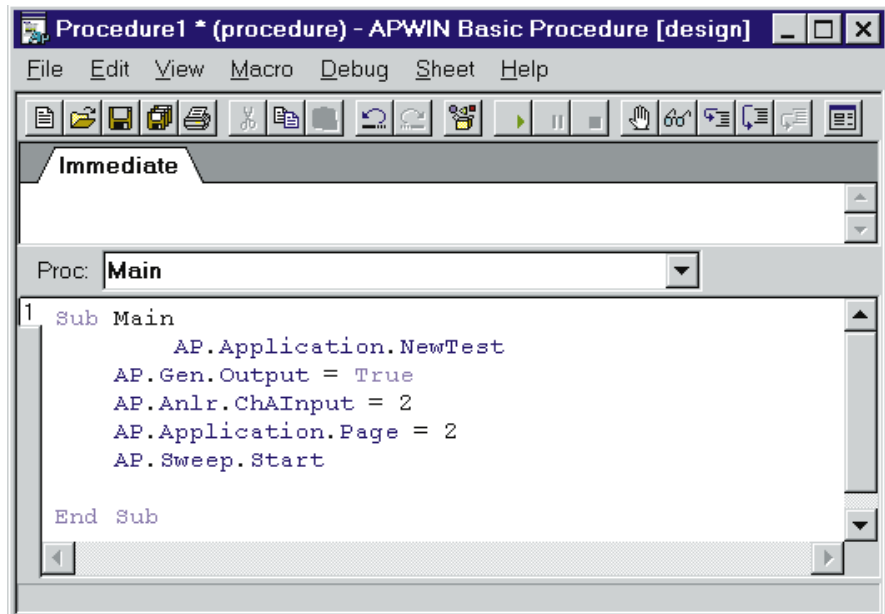



Figure 19-1 Learned Procedure Example

- Click on GO (or press ) .
- Click on Stop Learn mode icon.

Opening the Procedure Editor should show the program listing as illustrated in Figure 19-1. This procedure will duplicate all the actions above if the Run Procedure icon is clicked.

## Optimizing Automation Speed

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Many considerations affect testing speed. Some of the more important factors are listed below, with suggestions on alternatives that may result in faster testing.

### Computer Characteristics

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The computer processor type, clock rate, amount of memory, type of graphics card and amount of graphics card memory all have a strong effect on the speed of most tests. The computer specifications particularly affect “compute intensive” operations such as the speed of “painting” panels and graphs on screen when a test is loaded, time required for the Compute utilities, etc. Use faster computers with more memory and fast graphics cards with more graphics memory to speed up operation.

### Other Programs Running

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The number and complexity of other programs running simultaneously on the same computer affect testing speed. For the fastest audio testing speeds, shut down as many other programs as possible while running APWIN. This is particularly true for programs which may respond to external stimulus (e-mail, etc.).

### Number of Sweep Points

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200-point sweeps take longer than 10-point sweeps. For fastest testing, take data at only the number of points truly required. If the required point density is greater across some portions of the spectrum than others, consider use of Table Sweeps via an .ADS file which contains a list of specific points at which measurements will be made.



## Type of Measurement

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For example, THD+N sweeps are slower than Level sweeps due to the necessity of tuning the bandreject filter to each new frequency and waiting for measurements through the filter to stabilize. One cannot normally substitute a level measurement for a THD+N measurement, of course, but it may be appropriate to compromise to a small number of points in a THD+N sweep in recognition that each point is relatively slow.

## Single vs. dual channel (stereo)

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The System Two analog and digital analyzers have two Level meters and so can make level and response measurements of a two-channel device in the same time as for a single channel. Likewise, the FFT-based programs are two channel and have almost no speed penalty for measuring both instead of only one. However, a number of the real-time measurements made with the main (reading) meter, such as THD+N, can be made only on one channel at a time. For the greatest speed with such measurements on a two-channel device, a stereo sweep automatically sweeps one channel, changes input and output connections, and sweeps the second channel in less time than loading and running separate tests for each channel. For measurements that must be made sequentially on the two channels, speed is improved by minimizing the number of measurement points. An alternate technique such as multitone, based on two-channel FFTs, should be considered for the greatest speed on stereo devices.

## Autorangeing in analog analyzer vs. fixed range selections

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When signal levels change, autorangeing takes time to find the correct new range. When a test is loaded with ranging set to auto, the analyzer must adjust itself to the present value. If the test is loaded before the device under test is connected or stabilized, the analyzer may first range down to the noise level, then range up again after signal appears. When the signal level is known within reasonable limits, fixed range operation is much faster. The risk is that an unexpected change in signal level may cause clipping or noise-limited measurements in the analyzer.

## **“Keep all readings active” feature**

---

When the “Keep all readings active” option is selected in the Utilities Configuration dialog, APWIN during sweeps must pass readings from every measurement section of the instrument to its display location on the computer screen, even though that particular measurement is not being plotted. Sweeps will be slower as a result. For fastest speed, turn off the “keep all readings active” feature unless it is important to be able to watch some measurements on an instrument panel or bargraph while others are being plotted on the Graph.

## **Sweep Settling**

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The values on the Sweep Settling panel for the particular measuring instruments being graphed may have a strong impact on sweep speed. If the Delay value is 100 ms instead of 30 ms, an additional 70 ms is spent at every step of the sweep. If Tolerance is 0.1% instead of 2%, it will take longer at every step of a sweep for the stream of measurements to stabilize within that narrower range. For the fastest sweep speeds, use the largest value of Tolerance consistent with the repeatability desired for the data. Use the smallest number of Points and the shortest Delay which provide repeatable data. However, when testing devices with significant time delay such as satellite links or three-head tape recorders, the Delay value must be at least as long as the known time delay through the device under test. Use the Exponential rather than Flat Algorithm if repeatability is acceptable.

## **Multitone vs conventional swept sinewave testing**

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Multitone testing normally provides much faster results than swept sinewave techniques. For example, a 31-tone multitone signal can provide response, distortion, noise, and phase measurements on a stereo or mono device in a few seconds, compared to perhaps 45 seconds or one minute for the same types of measurements at the same 31 frequencies via conventional swept sinewave techniques. Response and phase measurement results are normally identical between swept sine and multitone techniques. Distortion and noise measurements are made by fundamentally different methods in the

two techniques, so there is no simple answer to the question of correlating results from the two types of measurements. It is typically necessary to characterize a known-good device with multitone techniques in order to determine suitable test limits.

### Analog hardware analyzer vs. DSP audio analyzer

The digital analyzer with the filter-based DSP Audio Analyzer program (ANALYZER.AZ2) selected and Low Bandwidth (1x) A/D as the Input can make many of the same measurements as the analog hardware analyzer. The DSP-based program has several features that provide greater speed than the analog analyzer. The Fast RMS detector is a synchronous detector that measures across a small number of complete cycles of the signal rather than for a longer, fixed measurement period. Reading rates up to 128 readings per second are available. Response measurements with the Fast RMS detector may be several times faster than response measured with the analog hardware analyzer. The compromises are in response flatness, repeatability, residual distortion and noise, and bandwidth.

	<b>Analog Hardware Analyzer</b>	<b>DSP Audio Analyzer, Low BW A/D, RMS Detector</b>	<b>DSP Audio Analyzer, Low BW A/D, Fast RMS Detector</b>
<b>Frequency Range</b>	10 Hz-200 kHz	10 Hz-22 kHz @48 kHz rate	10 Hz-22 kHz @48 kHz rate
<b>Frequency Response Flatness 10 Hz-20 kHz</b>	±0.008 dB (typically ±0.003 dB)	±0.04 dB typical	±0.04 dB typical
<b>Max. Data Spread, 10 sweeps, 20 Hz-20 kHz</b>	±0.003 dB typical, settling tolerance 0.03%	±0.04 dB typical, settling tolerance 1.0%	±0.04 dB typical, settling tolerance 1.0%
<b>Stereo Response Sweep Speed, 31 Pts 20-20k</b>	7 seconds typical, analog generator	5.3 seconds typical, sine (D/A) generator	1.5 seconds typical, sine (D/A) generator

<b>Residual</b>			
<b>THD+N</b>	Typical	Typical	Typical
<b>1kHz, 2 V,</b>	0.00017%	0.00054%	0.00054%
<b>22/20 kHz</b>	(-115 dB)	(-105 dB)	(-105 dB)
<b>BW</b>			

## Bandpass/Bandreject Filter Tracking Methods

DGen tracking, AGen tracking, or Sweep Track will all produce faster sweeps in THD+N, Bandpass, Bandreject, or Crosstalk measurement functions than the Counter Tuned alternative. The various “Track” selections avoid the time necessary to produce a stabilized frequency reading at each step of a sweep before sending the frequency value to the BP/BR filter. This is true of both analog hardware and digital analyzers.

## Mouse activity during a sweep

Keeping track of mouse movements on the screen during a sweep uses part of the PC processor’s capability and consequently slows down the sweep.

## Analog vs. DSP-D/A Generator

The DSP generator and D/A converters are selected by the several analog generator waveform selections with “(D/A)” as part of the waveform name. This DSP generator has no frequency range switching and stabilizes in amplitude faster at each new frequency than the analog hardware generator, so should be used for the fastest sweeps when its distortion and flatness are acceptable. The major performance specifications are compared below.

	<b>“Sine” Waveform Selection</b>	<b>“Sine (D/A)” Waveform Selection</b>
<b>Frequency Range</b>	10 Hz-204 kHz	10 Hz-47% of sample rate (22.56 kHz @ 48 kHz)
<b>Frequency Response Flatness, 10 Hz-20 kHz</b>	±0.008 dB (typical ±0.003 dB)	±0.03 dB to 18 kHz +0.03, -0.15 dB 18 kHz-20 kHz

<b>Residual THD+N, 20 kHz BW (system spec, including analog analyzer)</b>	0.0004% (-108 dB)	0.0015% (-96.5 dB)
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**Loading set-ups as tests vs. changing parameters from a procedure.**

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In a procedure, changed instrument set-up conditions can be achieved by loading test files (.AT1/.AT2) from disk for each new or modified setup. Changes can also be implemented by sending commands directly to the individual instrument parameters. Where only a few settings or values must be changed from the previous setup, it is normally faster to send commands from the procedure since several lines of a procedure can be executed in less time than it takes to load a complete test. If many changes must be made, it may be faster to load a test. Loading a test has the advantage of guaranteeing a new, fixed set-up. Incremental changes depend upon the instrument condition at the start of the changes, so modifications to earlier portions of the procedure or to a test loaded earlier will produce a different result.

**Time taken by the PC and Windows to draw the various panels and graphs on screen**

---

The PC takes a finite amount of time to draw each window (panel, graph, etc.) on the currently displayed page of workspace. It is not necessary for any panels, or even a graph, to be displayed during completely automated testing if limit files are being used to produce pass/fail results. The fastest test loading will occur if no windows are displayed. While this can be achieved by “killing” all panels and graphs before saving a test, that test is then awkward to modify or use in an interactive mode. The “AP.Application.Visible” group of procedure commands may be used as global commands at the beginning of a procedure to inhibit the display of all panels or of certain types (panels, graphs, bargraphs, etc.) in all tests of the procedure. User dialogs will still be displayed.




# Switchers

## APWIN Panel




The SWR-122 and SWR-2122 switchers are high-performance 2 x 12 relay switching matrices, stackable up to 16 units (192 channels). SWR-122 family switchers are no longer available, so only the SWR-2122 switchers will be described in this manual. SWR-2122 family switchers are available in four versions:

- a balanced Output Switcher (SWR-2122M) with 12 male XLR connectors, to fan out the analog generator's two outputs to as many as 12 device inputs. Complement mode may also be selected which drives all except one output connector.
- a balanced Input Switcher (SWR-2122F) with 12 female XLR connectors, to fan into the analog analyzer's two inputs from as many as 12 device outputs
- a balanced Patch Point, or insertion, switcher (SWR-2122P) with 12 5-pin XLR connectors carrying both input and output circuits. Each output pair of pins is "normalled through" to the input pins on the same connector when the control relay is not operated, but can break the path when desired to insert a signal
- an unbalanced switcher (SWR-2122U) with female BNC connectors instead of XLR connectors. Via rear panel switch settings, it can be made to function as an Input or Output switcher.

To bring the Switcher panel to the screen:

Mouse: click on the Switcher icon  on the Panels Toolbar, or the Panels Switcher menu command.

or

Keyboard: press the    keys (for Panels sWitcher)

The Switcher panel controls from 1 to 32 switcher modules (up to 16 input and 16 output). Each unit is an independently powered external accessory, connected in

“daisy chain” fashion to the same Audio Precision interface bus as the measurement instrument and computer.

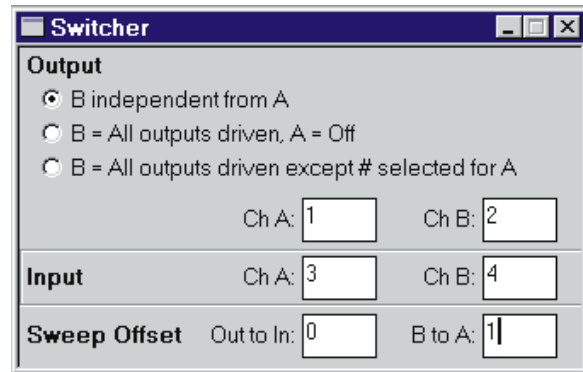


Figure 20-1 Switcher Panel

The Switcher panel permits selecting any desired input and output connector. The switchers may also be controlled as a swept parameter in order to scan a measuring instrument across many device outputs, or to scan a stimulus parameter across many device inputs. The Sweep panel, not the Switcher panel, determines this type of operation. The Switcher panel settings will determine the switcher conditions before a switcher sweep begins and after it ends.

## Input Channel

The channel numbers entered in the Input Channel A and Input Channel B fields determine the Input switcher connections unless a switcher sweep is in progress. The acceptable range of numbers for each field is from 1 to 192, since up to 16 Input switchers may be used. The practical range depends upon how many switchers are connected and how the rear panel switches are set on each switcher. Typically, the rear panel switches are set at installation time so that the first Input switcher responds to control as channels 1-12, a second Input switcher (if present) is set as channels 13-24, etc. Entering a zero or a number beyond the range of Input switchers actually present causes there to be no connection.

To sweep across switcher channels during a sweep test, the Sweep Panel Source 1 or Source 2 Browser must be used to select **Swr** as the



instrument and the particular channel and sweep mode desired as the parameter.

## Output Channel

The channel numbers entered in the Output Channel A and Input Channel B fields determine the Output switcher connections unless a switcher sweep is in progress. The acceptable range of numbers for each field is from 1 to 192, since up to 16 Output switchers may be used. The practical range depends upon how many switchers are connected and how the rear panel switches are set on each switcher. Typically, the rear panel switches are set at installation time so that the first Output switcher responds to control as channels 1-12, a second Output switcher (if present) is set as channels 13-24, etc. Entering a zero or a number beyond the range of Output switchers actually present causes there to be no connection.

To sweep across switcher channels during a sweep test, the Sweep Panel Source 1 or Source 2 Browser must be used to select Swr as the instrument and the particular channel and sweep mode desired as the parameter.

## Output Modes (Switcher)

Switcher output channels A and B may have certain specific relationships defined between them, controlled by the three “radio buttons” at the top of the Output section of the Switcher Panel. These three relationships are:

- B independent from A
- B = All outputs driven, A = off
- B = All outputs driven except # selected for A

When **B independent from A** is selected, channels A and B may be independently set to any channel number within their range. This is the normal mode for most operation.

When **B = All outputs driven, A = off** is selected, the switcher B common input is connected to all 12 outputs on each Output

switcher module and the A common input is disconnected. Both the A and B output fields will be gray and unavailable for settings in this mode since all connections are defined by the mode itself. This mode enables connection of a single signal to all device inputs, which may be a requirement of a “burn-in” rack or a life test.

When “**B = all outputs driven except # selected for A**” is selected, the A common input connects to the channel number entered in the A output field. The B common input connects to the remaining 11 channels on that switcher and to all 12 channels of all other Output switchers connected. The purpose of this mode is for “worst-case crosstalk” measurements, so that all channels except one of a multi-track or multi-channel recorder or mixing console are driven while the output signal from the one undriven channel is measured. This mode is normally used with a nested sweep with Source 2 on the Sweep panel set to scan channel A input and output through all possible device channels while Source 1 is commonly set for a frequency sweep to measure selective crosstalk across the audio spectrum.

## Switcher Sweep Offsets

When the switchers are selected as the instrument at Source 1 or Source 2, a number of Parameters are available in which both Channels A and B, or both Input and Output switchers, may be simultaneously swept. These complex switcher sweeps often require that Channel A and Channel B are not connected to the same channel, or that the Input and Output switchers are not connected to the same channel. The two offset fields on the Switcher panel set the relationship between the switcher parameters being swept. Before a sweep starts or after a sweep is completed, the switcher channel selection will be determined only by the Output and Input sections of the Switcher panel.

**Out to In Offset:** this field determines the channel number difference between the Output Switcher and the Input Switcher at any moment during a sweep. For example, if the value “1” is entered in the Out to in Offset field, the Output switcher will always be connected to the next higher channel number than the Input switcher connection (if Input is at 7, Output is at 8, etc.) An Out-to-In Offset of “1” would

permit crosstalk measurement scans of a multitrack tape recorder, measuring the adjacent higher channel. A value of “-1” would permit measurement of the adjacent lower channel during a scan.

**B to A Offset:** this field determines the channel number difference between Channel B and Channel A at any moment during a sweep. An A-to-B Offset of “1” is often used, along with a Step Size of 2, when testing stereo devices which are connected with stereo pairs going to adjacent switcher channels (device 1 L and R to switcher channels 1 and 2, device 2 L and R going to switcher channels 3 and 4, etc.).

## Switcher Control During Sweeps

The switchers may be controlled in a variety of useful ways during a sweep, by selecting the parameters at Source 1 or Source 2 in the Parameter column for the type of sweep desired. The available sweep control parameters and their functions are as follows.

**Ch. A Input:** The Channel A connector on the Input Switchers will be scanned across the range of switcher input connectors specified by the Start and Stop values at the Source settings of the Sweep panel. All other switcher connections will remain as set on the Switcher panel. This is a common selection to check the same audio parameter on a number of single-channel devices in sequence.

**Ch. B Input:** The Channel B connector on the Input Switchers will be scanned across the range of switcher input connectors specified by the Start and Stop values at the Source settings of the Sweep panel. All other switcher connections will remain as set on the Switcher panel.

**Ch. A Output:** The Channel A connector on the Output Switchers will be scanned across the range of switcher output connectors specified by the Start and Stop values at the Source settings of the Sweep panel. All other switcher connections will remain as set on the Switcher panel. This is a common selection to drive the inputs, one at a time, of a number of single-channel devices in sequence.

**Ch. B Output:** The Channel B connector on the Output Switchers will be scanned across the range of switcher output

connectors specified by the Start and Stop values at the Source settings of the Sweep panel. All other switcher connections will remain as set on the Switcher panel.

**Ch. A Input/Output:** The Channel A connector on the Input Switchers and the Channel A connector on the Output Switchers will both be scanned across the range of switcher output connectors specified by the Start and Stop values at the Source settings of the Sweep panel. If the Input/Output Offset field on the Switcher panel is set to a zero value, both input and output switchers will be set to the same connector number at any given moment. If the Input/Output Offset value is not zero, the selected Output channel number will be greater or smaller than the Input channel number by the value of the I/O Offset field and the Input channel number will correspond to the present command from the Sweep panel Source section. For example, if the I/O Offset is set to 1, the Output switcher will be set to the next higher connector number than the Input switcher at any time during a sweep, and the Sweep panel effectively “drives” the Input channel number. All other switcher connections will remain as set on the Switcher panel. This is a common selection, using an I/O Offset value of zero, to perform tests across the channels of a multi-track tape recorder or a number of monaural devices.

**Ch. B Input/Output:** Identical to Ch. A Input/Output, except that it is the Channel B connectors which are swept. All other switcher connections will remain as set on the Switcher panel.

**Ch. A + B Input:** The Channel A and Channel B connectors on the Input switchers will both be scanned across a series of input connectors. If the A/B Offset field on the Switcher panel is set to zero, both channels A and B will connect to the same input connector. If the A/B Offset field is not zero, the Channel B connection will always be to a connector higher or lower in number than the Channel A connection, by the value in the A/B Offset field. For example, if the A/B Offset value is set to -2, the Channel B Input connector will always be connected to the Input connector two steps below where the Channel A connection is. All other switcher connections will remain as set on the Switcher panel. This selection could be used to sequentially monitor the output signals from a group of stereo power amplifiers on a burn-in rack, all of whose inputs are constantly driven. For that

application, the A/B Offset value would be set to 1 and the Stepsize parameter would be 2. Thus, at the first sweep step channel A connects to switcher channel 1 (device number 1 left channel output) and B connects to switcher channel 2 (device number 1 right channel output). At the next sweep step, channel A connects to switcher channel 3 (device number 2 left channel output) and B connects to switcher channel 4 (device number 2 right channel output), etc.

**Ch. A + B Output:** Identical to Ch. A + B Input, except for the Output switchers. All other switcher connections will remain as set on the Switcher panel.

**Ch. A + Ch. B Input/Output:** Both the A and B connections of both Input and Output switchers are swept across the defined sequence of input and output connections. The difference between Channel A and Channel B settings and between Input and Output settings will be as determined by the two Offset fields on the Sweep panel. This is a common selection to test audio parameters of a number of stereo devices such as power amplifiers in sequence. For this application, the Input/Output Offset value would be zero, the A/B Offset value would be 1, and the Step Size parameter would be 2. Thus, at the first step of the sweep, A input and output are connected to switcher channel 1 (device number 1, left channel) and B input and output are connected to switcher channel 2 (device number 1, right channel). At the second step of the sweep, A input and output connect to switcher channel 3 (device number 2, left channel) and B input and output connect to switcher channel 4 (device number 2, right channel), etc.

## Frequency or Amplitude Sweeps on Many Switcher Channels

A common application is to measure some characteristic, such as frequency response or distortion or noise versus frequency, on a number of nominally identical switcher channels. The results can all overlay on the same graph for channel-to-channel comparisons. Any individual channel traces or combinations of traces may be displayed or printed. This type of test is set up as a “nested sweep” with the switcher selected at Source 2 (on the large form of the Sweep panel). To measure frequency response on all 24 tracks of a tape recorder, for example, select Swr as the instrument and A I/O (Channel A, Input and

Output) as the parameter. Enter 1 for the start, 24 for stop, and 1 as the step size. For frequency response, Source 1 would be Analog Generator Frequency and Level A would be plotted at Data 1. When the sweep is started, Source 2 causes the switcher to connect the channel A input and output to channel 1 and frequency is swept between the Source 1 start and stop values. When data at the stop point is measured, Source 2 steps channel A input and output to channel 2 and the Source 1 sweep repeats. This process continues until all 24 tracks have been measured and plotted. The Graph Legend may be used to turn on and off or highlight by color or trace width particular traces of interest.

### Switcher with Multi-Track Recorder

Multi-track recorders can be quickly tested by use of switchers and “nested sweeps” using both Source 2 and Source 1 on the Sweep panel. The diagram shows typical switcher connections to test multi-track recorders. 32 and 48 track recorders would require three each and four each, respectively, of both input and output switchers.

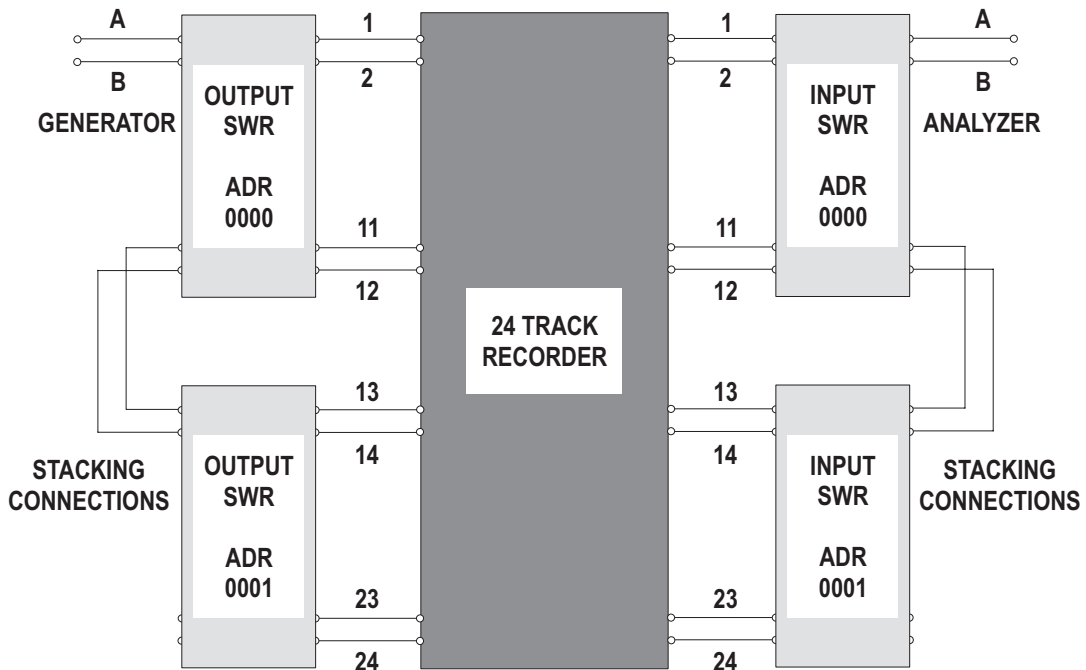


Figure 20-2 Switcher Connection and Address Setting Diagram, Multi-track Recorder

The generator drives the selected recorder input channel via Output switchers and the analyzer selects the output of the same channel via the Input switchers. The second switcher and any others necessary to handle all channels are “stacked” by having their rear panel A and B channels connected in parallel. The “8-4-2-1” rear panel binary address switches must be set to the proper code to make the switchers respond to address 1-12, 13-24, etc., as required.

A typical frequency response measurement across all tracks would be set up with:

- a Level meter connected to analyzer input channel A
- that Level meter then selected at Data 1 on the sweep panel
- a generator frequency sweep set up at Source 1 on the sweep panel
- a switcher channel A input/output “sweep” set up at Source 2 on the sweep panel, starting on switcher channel 1 and stopping at the last channel of the recorder, with a step size of 1

The response curves for all channels would then be overlaid on the same graph and can be compared for level balance, same response shape, etc. The Graph Legend can be used to temporarily turn off unwanted traces or to highlight the trace for a particular channel in a different color or by making the trace thicker.

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## Switcher Hardware

Audio Precision’s switcher family is designed for use with System One and System Two in the testing of multiple input-output devices such as mixing consoles, multi-track tape recorders, audio routing switchers, distribution amplifiers, or multiple units of simpler units during burn-in, environmental test, and similar activities. The family consists of balanced input, output, and patch-point switchers plus an unbalanced input/output switcher. The patch point switcher permits signal measurements at connection points or allows breaking the

normal path, measuring the unloaded output of the previous stage, and driving the following stage with the generator.

## Functional Description

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The switchers differ from one another primarily in connector configuration and in attributes set by internal jumpers or rear-panel switches which define them as an input switcher (connecting multiple device-under-test outputs to the analyzer inputs), an output switcher (connecting generator outputs to multiple inputs of devices under test), or a patch-point switcher.

Each of the switchers is a balanced 12 x 2 crosspoint matrix. Either of the two common points can be connected, under software control, to any of the twelve selectable points. The circuit boards are of balanced design but may be used with unbalanced circuits with the BNC connector version (floating shells). Up to 16 modules of the input and/or output switcher may be stacked and connected in daisy-chain fashion to permit testing devices with up to 192 inputs and/or outputs (up to 96 stereo pairs). Rear panel programming switches permit setting the address of each module in a stacked-module system for the desired channel number selections 1 through 12, 13 through 24, etc. A simplified diagram of the input and output switchers is shown in Figure 20-3. The relay contacts shown as single switches are actually multiple relay contacts connected in a more complex arrangement. This is necessary to obtain the isolation and crosstalk required for practical professional and high-grade consumer applications.

## Input Switcher

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The input switchers function basically as the simplified diagram indicates. Any of up to 12 balanced circuit points per module (192 with the maximum 16 modules stacked) may be connected to either the A or B connector, which would in turn normally be connected to the A and B inputs of the analyzer. The balanced version of the input switcher has two male XLR connectors for connection to the analyzer inputs, and 12 female XLR connectors for connection to device-under-test outputs. The unbalanced BNC version may be configured as an input switcher via a rear-panel switch.



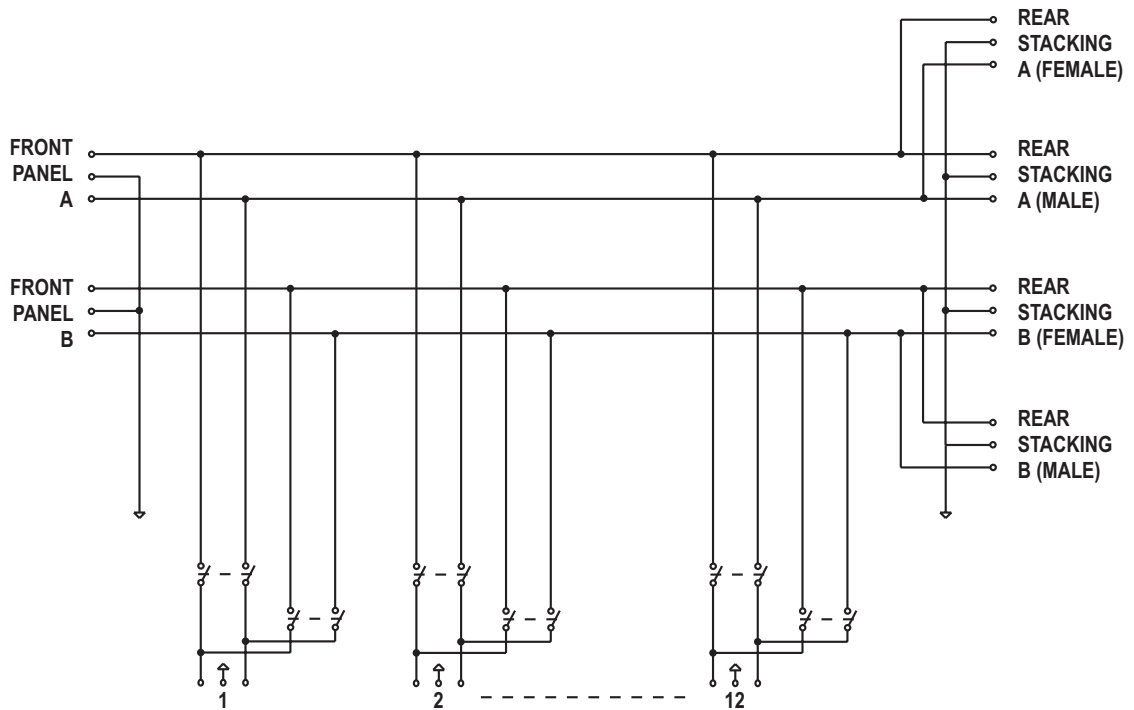


Figure 20-3 Input and Output Switchers, Simplified Functional Diagram

## Output Switcher

The usual function of the output switcher is to connect System Two generator outputs to a number of device inputs. It is electrically different from the input switcher in that it also has the capability of connecting one output of the generator to all but one input of a device, such as a routing switcher. The output of that one non-driven channel may then be measured to obtain a worst-case crosstalk measurement. The balanced version has two female and 12 male XLR connectors. The unbalanced BNC version may be configured as an output switcher by a rear-panel switch.

## Patch Point Switcher

The patch point switcher differs considerably from the input and output switchers. The two “common” connectors of the patch point switcher are a female connector 3-pin XLR (INPUT) for connection to

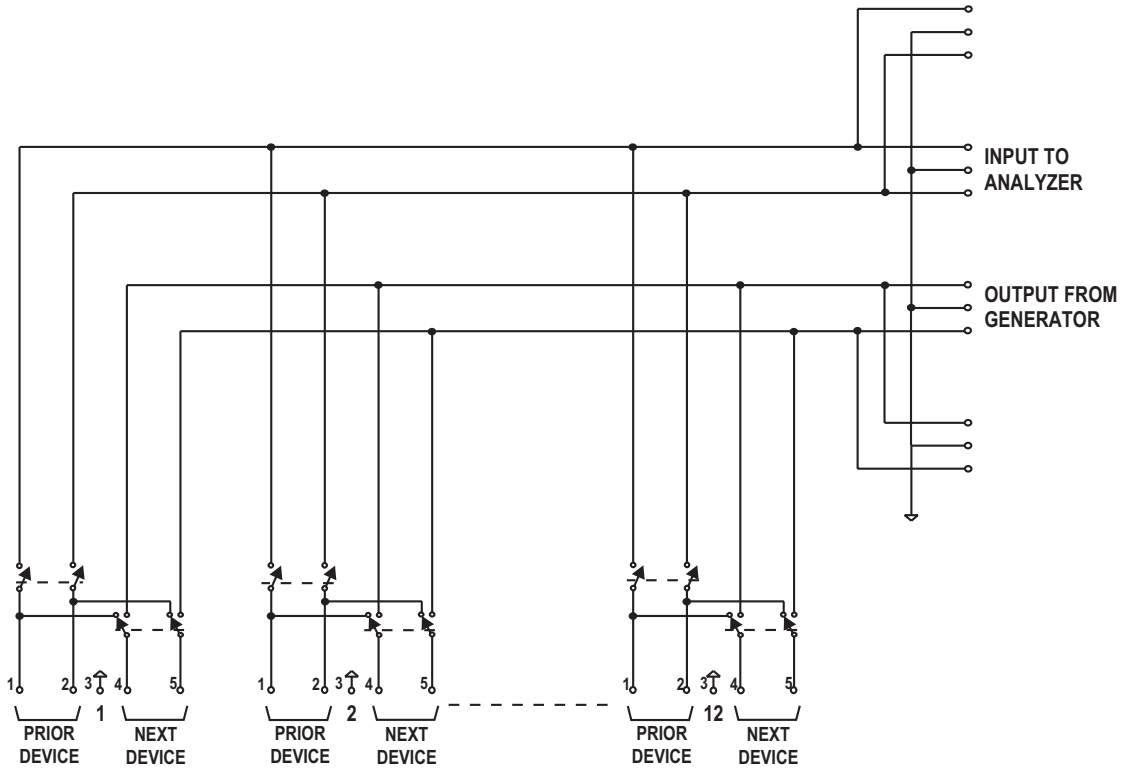


Figure 20-4 Patch Point (Insertion) Switchers, Simplified Functional Diagram

an output of the generator and a male 3-pin XLR connector (OUTPUT) for connection to an input of the analyzer. The twelve connectors of the patch point version are 5-pin XLR connectors. They are designed for connection at major circuit nodes of a console, or between series-connected devices in an audio chain in a studio, transmitter, or sound reinforcement system. Pins 1 (high) and 2 (low) connect from a balanced output of the preceding device. Pin 3 is ground. Pins 4 (high) and 5 (low) are the signal to a balanced input of the following stage or device. Figure 20-4 is a simplified diagram of the patch-point switcher.

The patch point switcher is “fail safe” in that in its normal mode (power off, or no channel selected) the previous stage or device is directly connected to the input of the following stage or device.

## Switcher Jumper Selection

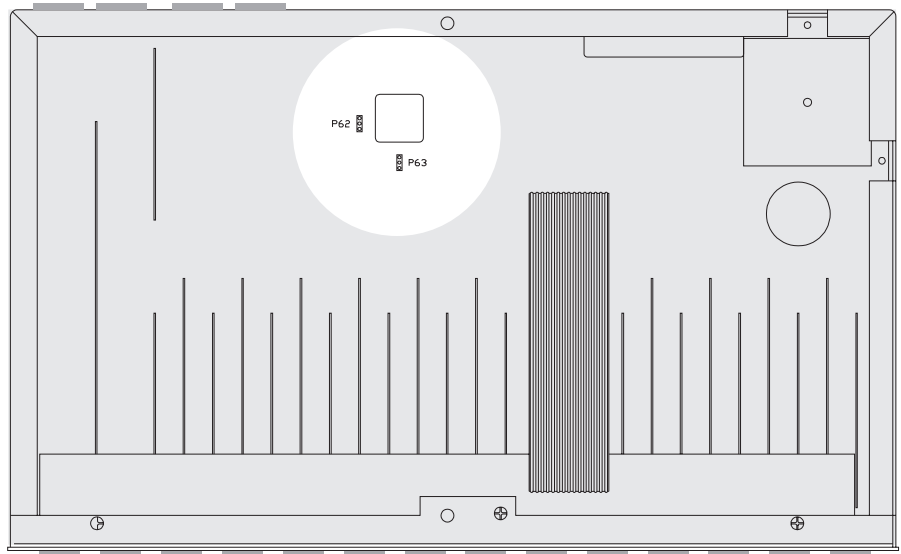


Figure 20-5 P62 and P63 Jumper Locations Inside SWR-2122 Switchers

The switcher circuit board has two jumpers on it labeled P62 and P63. The setting of these jumpers determines whether the switcher works as an input, output, patch point, or unbalanced switcher. Figure 20-5 shows the overall switcher circuit board with the location of the jumpers. Figure 20-6 shows the settings of the four jumpers for each of the four models of switcher.

### Unbalanced Switcher, Selection of Input vs. Output

The unbalanced switcher uses BNC connectors that may be used for generator output or analyzer

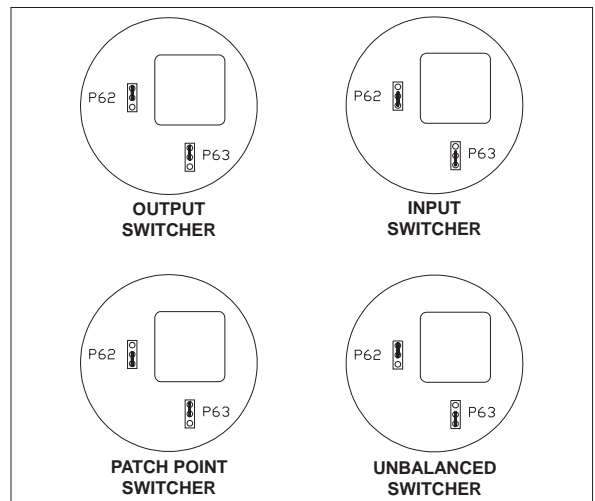


Figure 20-6 P62 and P63 Jumper Placement for Output, Input, Patch Point, and Unbalanced Switchers

input switching. A miniature switch on the rear panel of the switcher accomplishes selection between these two modes. When switch 6 (the right-end switch as viewed from the rear) is in the UP position, the unbalanced switcher operates as an input switcher for use with the analyzer. When the switch is in the DOWN position, the unbalanced switcher operates as an output switcher for use with the generator. Note also the labeling on the rear of the unbalanced switcher, describing the difference in cable connections to the four rear-panel BNCs when used as an input versus output switcher.

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## Switcher Installation

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Switcher installation consists of physically mounting the switcher modules, setting the ac mains voltage range assembly, providing proper ac mains power, electrically interconnecting them, setting rear panel switches for address, and (on unbalanced switchers only) setting rear panel switch 6 for input or output function. Physically, the switcher modules may be either rack-mounted or simply stacked on top of one another. Ac mains connections are made via standard IEC connectors. A rear panel voltage change and fuse assembly can be set for 100, 120, 220, or 240 Volts AC. No power switch is provided; the units are designed for continuous service.

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### Switcher Address Settings

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Each switcher module consists of 12 channels. Up to 16 modules may be stacked to provide up to 192 channels. Rear panel address switches must be set to tell each switcher which channel commands from the software it should respond to. For example, the first switcher is normally set to be channels 1-12, the second module to channels 13-24, etc. The four left-hand switches on the six-switch binary switch bank on the rear panel set the module address. The table below shows relationships between address switch position, binary code, and channel number on the computer software “panels”.

REAR SWITCH SETTINGS				BINARY CODE	CHAN #
#8	#4	#2	#1		
up	up	up	up	0000	1-12
up	up	up	down	0001	13-24
up	up	down	up	0010	25-36
up	up	down	down	0011	37-48
up	down	up	up	0100	49-60
up	down	up	down	0101	61-72
up	down	down	up	0110	73-84
up	down	down	down	0111	85-96
down	up	up	up	1000	97-108
down	up	up	down	1001	109-120
down	up	down	up	1010	121-132
down	up	down	down	1011	133-144
down	down	up	up	1100	145-156
down	down	up	down	1101	157-168
down	down	down	up	1110	169-180
down	down	down	down	1111	181-196

## Cable Connections

Electrically, the digital control connectors of the switchers must be connected in “daisy-chain” fashion. Normally, the computer will be connected to the first switcher with a digital interface cable, the first switcher connects to the second, etc., and the last switcher connects to the System digital interface (APIB) connector. Each switcher has both a male and female digital interface connector on its back panel for these connections.

Audio signal connection must be connected in daisy-chain fashion via the rear panel XLR connectors (balanced) or BNC connectors (unbalanced). Normally, the generator and analyzer outputs and inputs will connect to the front panel inputs and outputs of the first pair of switcher modules (assuming multiple units of both input and output switchers are being used). The rear panel connectors will then be used to connect channels A and B of the first input switcher module to A and B of the next input switcher, etc.

## Patch Point Switchers

Rather than the common channel A and common channel B connector of Input and Output switchers, patch point switchers have a common input and a common output connector. Rear panel switches must be set on a Patch Point switcher to tell it to respond to software commands for Channel A, Channel B, or both. The two right-hand switches on the rear panel binary switch band must be set as shown in the table below to accomplish the desired control.

### REAR SWITCH

A	B	PATCH POINT MODE
down	down	OFF
down	up	CHAN B
up	down	CHAN A
up	up	EITHER A or B

These switches are disabled by the I/O jumper settings in input or output versions of the switcher.

Patch point switchers, input switchers, and output switchers may all be combined in the same system. Input and output switchers may share the same address. A patch point switcher must not have the same address as either an input or output switcher. Two patch point switchers may be set to the same address if one is set as Channel A and the other as Channel B.

## DCX-127 Multifunction Module

### Panel


The primary functions of the DCX-127 include an autoranging 4 1/2 digit volt-ohmmeter, two independent dc voltage outputs variable across the  $\pm 10.5$  Volts range, 21-bit (plus sign bit) low-speed digital inputs and outputs, and three 8-bit digital output ports. The DCX-127 also provides pulses and dc levels whose timing depend upon key software actions such as end of sweep, analyzer channel selection, or data settling. These may be used for synchronization and sequencing of external devices or portions of the test system such as the triggered sine burst generator. See the hardware discussion later in this chapter.

The DCX-127 Multifunction Module is an external accessory which is connected in “daisy chain” fashion to the Audio Precision Interface Bus.

To bring the DCX panel to the screen:

Mouse: Click on the DCX-127 icon  on the toolbar or select Panels DCX via the menu bar

Keyboard: Type    (for Panels DCX-127).

The DCX-127 Panel can be viewed in both large and small versions. Clicking on the panel icon on the Panels Toolbar brings up the small version panel. Holding down the  key while clicking on the toolbar

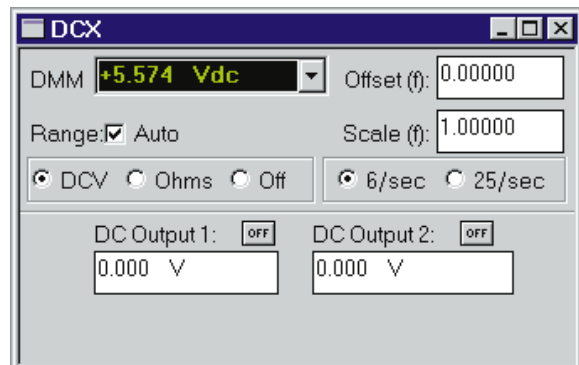


Figure 21-1 DCX-127 Panel, Small Version

icon brings up the large version.

## DMM Units

In DC Volts function, the reading may be displayed either in absolute Volts or as f(V), a function of Volts, after modification by Offset and Scale factors. Similarly, resistance readings may be displayed in Ohms or as f(O) with Offset and Scale. The “available units” list includes only units valid for the presently-selected function (DC or Ohms).

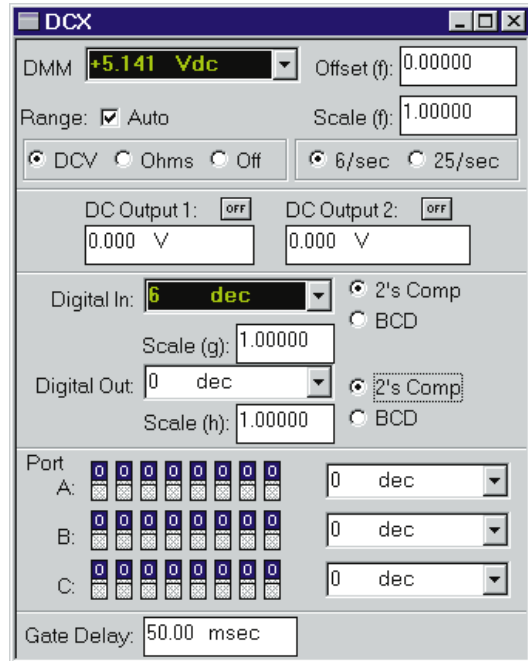


Figure 21-2 DCX-127 Panel, Large Version

## Auto/Fixed Range Checkbox

The Voltmeter/Ohmmeter may operate in autoranging mode or may be fixed on any one of its full-scale ranges.

## Fixed Range Selection Field

When the DMM Autoranging function is disabled (no check mark in Auto box), a Range field is displayed and active for manual range selection. New entries may be typed in or existing entries edited from the keyboard, and completed by pressing the **ENTER** key. When **ENTER** is pressed, APWIN software will select and display the most-sensitive range which can handle the entered value.

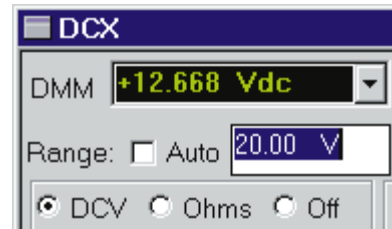


Figure 21-3 Autorange Disabled,




## Offset and Scale Values for f(V), f(O)

---

When f(V) (function of Volts) or f(O) (function of Ohms) units are selected for the DMM, APWIN software computes the value to display from the formula

$$\text{display} = (\text{measurement} + \text{Offset}) * \text{Scale}.$$

The “measurement” term is the value which would be displayed in Volts or Ohms units. The “Offset” and “Scale” values are the contents of the fields with those names, at the top right of the DCX panel. New entries may be typed into either field, or the existing entry edited, from the keyboard followed by pressing the  key.

## DCV/Ohms/Off Selection

---

The DMM (digital multi-meter) can measure DC Volts or Ohms, or be turned off, by use of the three buttons below the Auto Range check box.

## Reading Rate

---

The DMM can operate at 6 or 25 readings per second. Resolution is a full 4 1/2 digits at 6 readings per second. At 25 readings per second, the number of digits displayed is the same but the least significant digit is always 5 or 0.


## DC Output 1 and 2 On/Off

---

Either of the DC outputs (front panel double banana jacks) may be turned on and off individually.

## DC Output 1 and 2 Value

---

Either DC output may be independently set to any voltage in the range  $\pm 10.5$  Vdc. A new value may be entered or the existing value may be edited from the keyboard, followed by the  key.

## Digital Input Display and Units

---

The 21-bit digital input word connected to the DCX front panel DB25 Digital Input connector may be displayed in decimal,

hexadecimal, or octal units or as a scaled function  $g(x)$  of the decimal value at the digital input.

### Digital Input Data Format Selection

---


The binary values at the 21 active pints of the Digital Input connector may be interpreted as 2's complement or in 8-4-2-1 BCD (binary coded decimal) format.

### Scale Factor, Digital Input

---

When  $g(x)$  units are selected at the Digital In display, APWIN software computes the displayed value from the relationship

$$\text{display} = \text{measurement} * \text{Scale (g)}$$

where “measurement” is the decimal value of the binary data in the selected format and “Scale (g)” is the value entered in the Scale (g) field just below the Digital In display. A new Scale value may be entered or the present value edited from the keyboard, followed by pressing the  key.

### Digital Output and Units

---

The 21-bit digital output word presented to the DCX front panel DB25 Digital Output connector may be controlled in decimal, hexadecimal, or octal units or as a scaled function  $h(x)$  of the decimal value entered in the Digital Out field.

The value sent to the Digital Output connector, in the unit selected, is determined by the value entered into the Digital Out field but may also be swept from the Sweep Panel.

### Digital Output Data Format


---

The value and units entered in the Digital Out field will be presented to the front panel Digital Output connector in either 2's complement or 8-4-2-1 BCD (binary coded decimal) format.

## Scale Factor for Digital Output


When h(x) units are selected at the Digital Output control field, APWIN software computes the actual transmitted value from the relationship

$$\text{output value} = \text{entry value} * \text{Scale (h)}$$

where “entry value” is the decimal value entered into the Digital Out numeric field and “Scale (h)” is the value entered in the Scale (h) field just below the Digital Out control field. A new Scale value may be entered or the present value edited from the keyboard, followed by pressing the  key.

## Binary Control for Ports A, B, C

Ports A, B, and C are independent 8-bit digital control ports at separate DB9 connectors on the rear of the DCX-127. The settings of the 8 bits may be controlled from

the DCX software panel or they may be swept over a range via the Sweep Panel. Settings from the DCX software panel may be made as binary (mouse only), octal, decimal, or hexadecimal numbers. The binary control and display button-indicators are functionally in parallel with the octal-decimal-hexadecimal field. Any change made to the binary buttons is immediately reflected in the octal-decimal-hex field, and vice-versa. To change a value using octal, decimal, or hex units, the desired unit must first be selected (or the present unit accepted). A new value may be entered into the field from the keyboard or the existing value edited, followed by the  key.

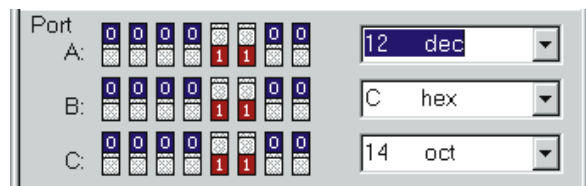


Figure 21-4 Output Ports, Showing Decimal, Hexadecimal, Octal, Binary Controls

## Program Control, Gate Delay

The front-panel Program Control Output DB9 connector provides several pulse or gate signals controlled from APWIN software. Pin 6 on this connector is a sweep gate which goes high at the beginning of

each APWIN sweep and drops low at the end of the sweep. Pin 1 is a delayed sweep gate, similar to pin 6 but with an additional user-settable delay time between sweep start and the leading edge of the gate signal. That delay time is controlled by the Gate Delay field at the bottom of the DCX panel.

### **Gate Delay Control for Delayed Trigger**

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The delayed sweep gate (pin 1) is similar to the sweep gate (pin 6) in that it goes low during a sweep and returns high at the completion of the sweep. However, the delayed sweep gate does not go low until the Gate Delay time has elapsed after the beginning of a sweep.

## **Hardware Discussion**

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The DCX-127 multi-function module is a separate unit packaged in a one-rack-unit high case which may be rack-mounted or used as a bench instrument. Control of the DCX-127 is via the same APIB (Audio Precision Interface Bus) which controls System One or System Two from the computer. The DCX-127 functions include:

- DC voltage measurement (from the millivolt range to 500 Volts)
- DC resistance measurement (from less than one Ohm to 2 Megohms)
- DC voltage outputs (two independent sources) anywhere in the  $\pm 10.5$  Volt range
- Digital input and display of static or low-speed parallel input, word width up to 21 bits
- Digital output of a parallel digital word of 21 bits
- Program control input of 8 lines which can trigger execution of software macros
- Program control output port which furnishes signals coincident with a number of different software activities such as sweeping,

selection of channel A versus channel B input, settling of data during a sweep, etc.

- Digital control outputs (four 8-bit ports) for general-purpose control of external devices.

All the DCX-127 output functions are controlled from APWIN software, and all of the inputs are displayed in APWIN software and may be displayed as bargraphs or plotted values during a sweep when appropriate.

## DC Voltage Measurements

The DCX-127 includes a 4 1/2 digit autoranging dc voltmeter-ohmmeter. In voltage mode, the meter will automatically select among its 200 mV, 2 V, 20 V, 200 V, or 500 V ranges for best resolution of the measured signal. It may also be fixed on any of its ranges. The meter has selectable reading rates of 6 or 25 readings per second. The 6/second selection provides full 4 1/2 digital resolution (20000 counts full scale). The 25/second selection still displays 4 1/2 digits, but the last digit will always be either zero or 5. Normal mode rejection and common mode rejection also degrade by approximately a factor of four when the 25/second reading rate is selected. Resolution versus range and reading rate are shown in the figure.

## Resistance Measurements

The resistance mode operates by forcing a calibrated value of current through the unknown resistance and measuring the consequent voltage drop. The ohmmeter is autoranging and will select either the 200 Ohm, 2 kOhm, 20 kOhm, 200 kOhm, or 2 megOhm range depending on the value of the unknown resistance being measured. It may also be fixed on any range. Reading rates of 6 readings per second or 25 readings per second may be selected. Software features also permit displaying a computed value which is a function of resistance, using user-entered values of offset and scale parameters. The DCX-127 input configuration permits making these resistance measurements on either a 2-wire or 4-wire basis.

## Two-wire connection

If only one pair of test leads is used, connected from the (+) and (-) terminals of the DCX-127 to the unknown resistance, a 2-wire measurement results. Current from the internal current source flows through an internal resistor, through the test leads and resistance being measured, and through another internal resistor back to the current source. The two internal resistors are indicated schematically on the front panel of the DCX-127. Voltage drop across these internal resistors, produced by this current, is internally measured at the (+) and (-) terminals. The resistance of the two test leads is thus included in the measurement. For moderate and high values of unknown resistance, the additional error introduced by resistance of the test leads is usually negligible.

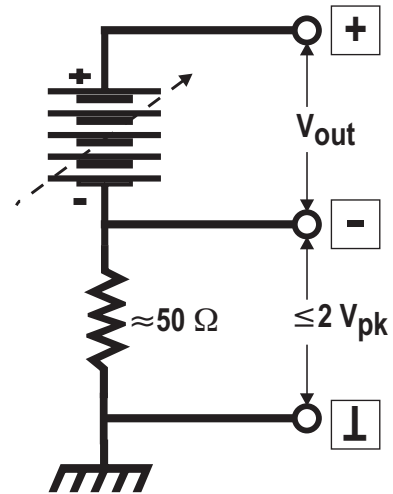


Figure 21-5 Equivalent Circuit, DC Outputs 1 or 2

## Four-wire (Kelvin) connection

For the highest accuracy when measuring low values of resistance, a 4-wire measurement (Kelvin lead connection) is recommended. For this measurement, one pair of test leads is connected from the two current source connectors (marked SOURCE) to the unknown resistance. A second pair of test leads is connected from the voltmeter input terminals (+ and - jacks) to the unknown. Current from the current source now does not flow through the voltmeter leads, so the resistance of the test leads is not included in the measurement.

The resolution of the reading varies with the range and with the reading rate as shown in the figure.

## DC Voltage Outputs

Two independent dc voltage outputs are available at the DCX-127 front panel. Each may be set with 20 microvolt resolution

(approximately 20 bits equivalent) to any voltage in the  $\pm 10.5000$  Volt range.

Example applications of the dc outputs include sweep control of the gain, offset, or both ports of voltage-controlled amplifiers (VCAs) while measuring their gain, distortion, or noise and plotting those values versus control voltage. The dc output can also be used to control the position of a dc-controlled turntable while measuring polar response patterns of a loudspeaker or microphone. Manually operated test equipment with a dc control port (such as the VCF input of a function generator) may be controlled by the dc outputs. Tape machines with dc-controllable bias oscillators may be connected, enabling MOL, SOL, sensitivity, and other data to be taken automatically.

The dc outputs may be floated up to 2 Volts away from ground. See Figure 23-5 for the equivalent output circuit of both dc outputs. The maximum current which may be drawn from them is 20 milliamperes. The maximum current which the DC outputs can sink is 10 milliamperes.

## Digital Input

A parallel digital word of up to 21 bits plus sign bit, at LSTTL-compatible levels, may be connected to the digital input connector on the front panel of the DCX-127. The value of this word may be displayed in decimal or hexadecimal representations. A user-entered scale factor may be entered and a function of actual input displayed, multiplied by the scale factor. The digital word presented to the connector may be in either two's complement (binary plus a sign bit) or 8-4-2-1 binary coded decimal (BCD) format. The word at the connector may either be asynchronously sampled at any of four different selectable reading rates or, in external mode, displayed each time a "data good" or "conversion completed" pulse is presented at pin 25. The maximum sampling rate depends largely on computer speed. A common application of the digital input is in static (DC) linearity testing of A/D converters. The digital input may also be used to interface readings from a BCD-display instrument such as a capacitance meter to the audio test system. The pin connections are shown in the figure.

## Digital Output

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An LSTTL-compatible parallel digital output word of up to 21 bits plus sign may be created at the digital output connector. The value of the word may be entered into the software in either decimal or hexadecimal representations or scaled decimal. The format of the word (bit-to-pin relationship) may be selected as two's complement or 8-4-2-1 binary coded decimal (BCD). The pin connections are shown in the figure. The strobe line (pin 25) will be pulsed low each time a new value is entered into the software panel or stepped to during a sweep.

A typical application of digital output is in static (dc) linearity testing of digital-to-analog converters. The digital output word drives the converter; the analog output of the converter is measured with the DCX-127 DVM. For high-resolution converters at output voltages more than  $\pm 200$  mV away from zero, it will be desirable to use one of the DCX-127 dc outputs in conjunction with the DVM input in differential voltmeter fashion so that the DVM can be set on its highest resolution range. Another application of the digital output is control of digitally-controllable turntables during polar response testing of microphones and loudspeakers. Still another application is in testing multiplying digital-to-analog converters (MDACs) used as variable resistors, attenuators, or in other audio applications. Each bit output has a five milliampere current drive capability and 390 Ohms output impedance.

## Program Control Outputs

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The Program Control Output connector provides six pulse or gate signals controlled by various actions of APWIN software. The pin assignments are shown in the figure. All lines are LSTTL compatible.

The Reset output, pin 2, pulses high when the DCX-127 power is turned on and whenever a Util Restore menu command is executed. The Reset output would typically be used to normalize the condition of external logic devices connected to a System One or System Two test station at power up. It may also be invoked in a procedure by inserting a Util Restore command into the procedure.



The Data Acquired output, pin 3, pulses high each time that a set of data satisfies the settling algorithm. See the Sweep Settling section for full details on the settling algorithm. This signal could be used, for example, to cause a Compact Disc player to automatically advance to the next track whenever a successful set of measurements is completed on the preceding track. This is useful during an external frequency or external amplitude test across a series of tracks of a test disc.


The Trigger output, pin 4, pulses high at the end of each delay portion of the settling process. Settling delay, in a generator-based sweep, is the time after the generator steps to a new value before the software starts examining data samples to see if they are settled. This signal could be viewed on a dual-trace oscilloscope along with the signal being measured, to verify that the delay has been optimally set to discard “left over” data from the previous generator step when measuring systems such as 3-head tape recorders or satellite paths which have significant time delays. The Trigger output of the DCX-127 could also be connected to the trigger/gate input of the generator when the BUR-GEN module is present. This will synchronize tone bursts in the sine trig mode with the system’s readiness to make a measurement. The Trigger output could also be used to strobe an external device being tested, such as an analog-to-digital converter. This will help insure that the source is stable before the conversion takes place.

The Sweep Gate output, pin 6, drips low at the beginning of a sweep test and goes high at the completion of a test. In a nested sweep, it will go high momentarily at the end of each Source-1 sweep and drop low again for the next Source-1 sweep after the Source-2 parameter is incremented. One example application of the Sweep Gate output is in testing of the attack characteristics of compressors, limiters, and similar audio processors. With the Sweep Gate connected to the trigger/gate input of the generator when the BUR-GEN module is present and the Burst-gated mode is selected, time zero on a time sweep graph and the start of the signal will be synchronized.

The Channel A/B output, pin 7, goes low whenever analog analyzer input channel A is selected and high whenever input channel B is selected. The level at this pin thus indicates which channel is being measured during a stereo sweep.

The Delay Gate output, pin 1, is similar to the Sweep Gate signal but with the addition of a user-settable delay time between the actual beginning of a sweep test and the leading edge of the gate output signal. This delay time is entered at the bottom of the DCX software panel, in the Gate Delay field.

An example application is power amplifier turn-on and turn-off transient (“thump”) testing. The amplifier power could be turned on by the sweep gate and off by the delayed gate. The output of the amplifier may be measured during a time sweep. To allow correct capture of transients, the Level meter should be used along with a fixed input range.

Another example application is to start a tape machine or turntable for run-up time testing. The basic measurement technique is to perform a time sweep while measuring frequency from a pre-recorded tape or disc. Use of the Delayed Sweep Gate to start the recorder or turntable permits the instrument to draw the graph when  is pressed, start the time sweep, and then start the machine at a predetermined time into the sweep when the software is ready to take continuous, rapid measurements.

## Digital Control Output Ports

Three rear-panel 9-pin connectors are provided on the DCX-127 for control of external devices. These output ports are intended for direct interface to LSTTL circuitry, or via LSTTL-compatible drivers to relays. Control of power, lights, annunciators, etc., are among the possible applications. The pin connections of these ports are shown below. The current drive capability of each bit line is five milliamperes; the output impedance is 390 Ohms.

CONNECTOR PIN	FUNCTION
1	bit 7 (msb)
2	bit 6
3	bit 5
4	bit 4
5	bit 3

6	bit 2
7	bit 1
8	bit 0 (lsb)
9	ground



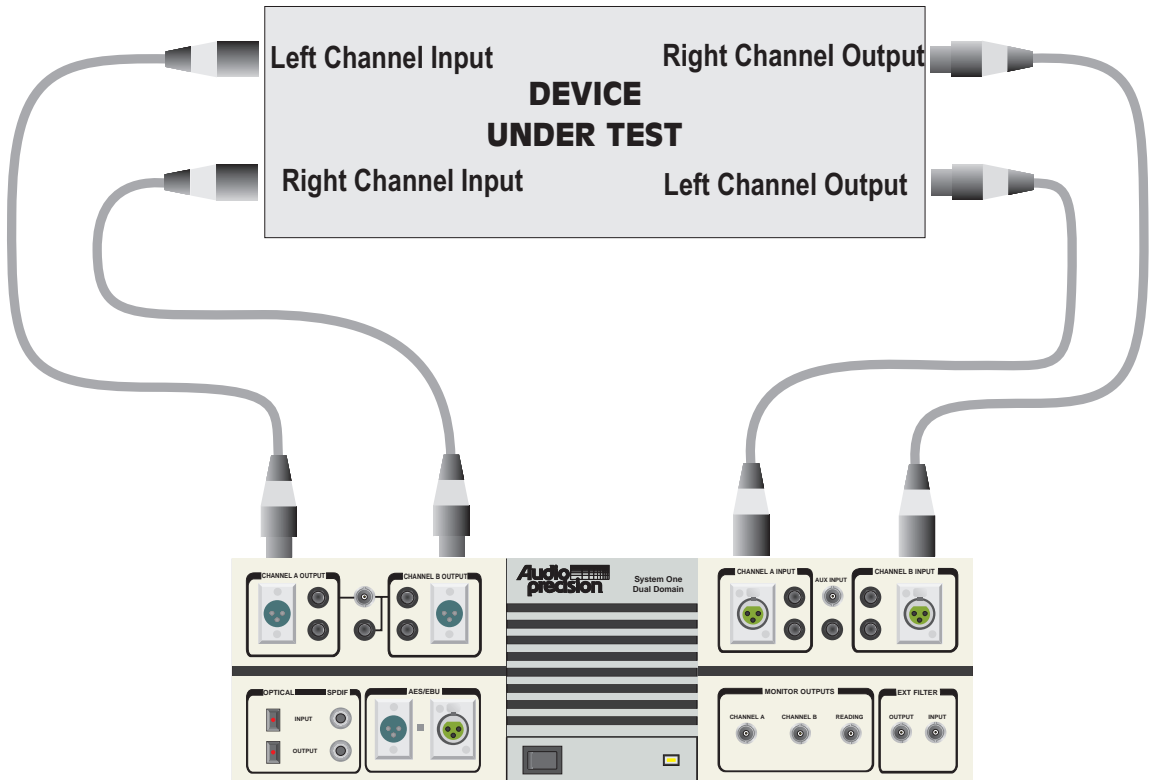
# Furnished Files

## Analog-to-Analog Tests

### General Information

A standard installation places System One analog-to-analog tests in the **C:\APWIN\S1\A-A** subdirectory.

Generator Output Configuration: All the tests described in this analog-to-analog section are set up using System One's balanced floating output configuration, which drives XLR and double-banana output connectors in parallel. If devices with unbalanced inputs are to be tested, the output configuration of each test used should be changed to unbalanced. Floating or grounded configurations may be



selected as required by the application. All tests are set up with the 50 Ohm source impedance at Z-Out; one of the other available impedances may be selected as required. All tests are set up with the generator amplitude at 1.000 Volt open circuit. This may be changed to a more appropriate value for any particular device under test in the generator Amplitude field.

**Analyzer Input Configuration:** All the tests in this analog-to-analog section expect signal from the device under test at the XLR or double banana input connectors. Do not change any GenMon selections.

## Frequency response

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### A-A\_RESP.AT1

**Purpose:** This test is set up to measure and graph frequency response on an analog input, analog output device. Mono or stereo devices may be tested, as described below.

**Stimulus:** The Analog Generator is at default conditions (sinewave, 50 ohms balanced output impedance, 1.000 Volt open circuit amplitude, “resting” frequency before and after sweep of 1 kHz).

**Measurement:** The 2-Ch. Ampl function of the Reading meter (no filters, RMS detector) makes the measurement.

**Sweep and Display:** Press the F4 function key to set the dBr reference value to the present measurement before pressing F9 or clicking Go to start the sweep. This will assure that the response curve passes through zero dBr at the generator resting frequency of 1 kHz. The sweep is 50 steps (51 values) logarithmically-spaced from 20 kHz to 20 Hz, sweeping from high to low. The graph Y axis covers the +/-15 dBV range. If values on the particular device under test plot off the top or bottom of the graph, use the Optimize function (click right mouse button on graph for a list of graph-related commands).

**Stereo:** Both generator channels are on but only channel A will be measured and plotted as the test is set up. To test a stereo device, click the Stereo Sweep checkbox at the bottom of the Sweep panel on Page 2. The Level B meter will then simultaneously plot at Data 3. Note that the Level meter is fully specified for accuracy only with signals of 10

mV and higher amplitude. If frequency response measurements are to be made at signal levels substantially below 10 mV, the Amplitude function (not 2-Ch. Ampl) of the Reading meter should be selected and stereo sweeps will require sweeping twice with a channel change before the second sweep. Both Data 1 and Data 3 share the left axis, which is calibrated in dBr units. The Level B trace will not pass through zero dBr at 1 kHz if there is a gain difference between the two channels at 1 kHz.

## Noise (single point)

### A-ANOISE.AT1

**Purpose:** This test is set up to measure noise, integrated across the audio spectrum, on a mono or stereo analog-to-analog audio device. The result is a single number (per channel), not a graph.

**Stimulus:** The Analog Generator output is off, so waveform and amplitude are irrelevant. However, the generator output impedance serves as a defined back termination for the device input if connected. The generator output configuration is at default conditions of 50 ohms impedance, balanced. Other impedances may be required by some noise-measurement specifications.

**Measurement:** The reading meter of the Analog Analyzer is in Amplitude function using the RMS detector. Other detectors may be selected instead; for example, the Q-Pk detector would be used for a CCIR 468 weighted noise measurement. The high and low pass filters are set to the 22 Hz to 22 kHz range. Thus, the measurement will be limited to noise in the 22 Hz-22 kHz band. Filter selections can be changed to other bandwidths if desired. An appropriate weighting filter (if installed) can be selected in the Filter selection field; for example, the CCIR weighting filter for a CCIR 468 weighted noise measurement.

**“Sweep” and Display:** A single-point measurement is made (no sweep) and tabular results are displayed in the Data Editor since a graph cannot be made of a single point. The reason for selecting Generator Amplitude A as the Source 1 parameter, with 1 nanovolt as the Start value, is merely to create a Column heading and value in column one of the Data Editor which implies that this is a noise measurement (no signal). Since the generator output is turned off on

the generator panel, the actual Source 1 parameter and setting is not important. The measurement is expressed in dBr units, which are decibels relative to the value in the Analog Analyzer dBr Reference field. This value could have been set to the measured level while the signal from the generator was turned on and the F4 key pressed, or a known standard reference level can be typed into the dBr Reference field.

**Stereo:** To test a stereo device, click the Stereo Sweep checkbox at the bottom of the Sweep panel on Page 2. Since the reading meter can only measure one channel at a time, the two channels will be automatically be measured in sequence with results displayed in columns two and three of the Data Editor.

### **THD + N vs. frequency**

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#### **A-ATHDFR.AT1**

**Purpose:** This test is set up to measure and graph THD+N (total harmonic distortion plus noise) as a function of stimulus frequency across the audio band on an analog-to-analog audio device. Mono or stereo devices may be tested.

**Stimulus:** The Analog Generator is at default conditions (sine waveform, 50 ohms balanced output impedance, 1.000 Volt open circuit amplitude, “resting” frequency before and after sweep of 1 kHz).

**Measurement:** The reading meter of the Analog Analyzer is in THD+N Relative function with the high and low pass filters set to the 22 Hz to 22 kHz range. Note that the 22 kHz low-pass filter will attenuate second harmonic products from fundamentals above 11 kHz, third harmonics of fundamentals above 7 kHz, etc. If the device under test is not already inherently band-limited in the 20 kHz area, it may be desirable to change the low-pass filter setting to 80 kHz in order to measure harmonics up through the fourth at the 20 kHz end of the sweep range.

**Sweep and Display:** The sweep is 20 steps (21 values) logarithmically-spaced from 20 kHz to 20 Hz, sweeping from high to low. The graph Y axis is logarithmic from zero dB (100%) at the top to -120 dB (0.0001%) at the bottom, covering virtually all possible



measurements. For better display of data after measurement of a particular device, use the Optimize function (click right mouse button on graph for a list of graph-related commands).

**Stereo:** Both generator Channels are on but only Channel A will be swept, measured, and plotted as the test is set up. To test a stereo device, click the Stereo Sweep checkbox at the bottom of the Sweep panel on Page 2. Since the reading meter can only measure one channel at a time, the stereo sweep will first sweep and plot Channel A, then switch analyzer channels automatically and sweep and plot Channel B onto the same graph.

## **THD + N vs. amplitude**

### **A-ATHDAM.AT1**

**Purpose:** This test is set up to measure and graph THD+N (total harmonic distortion plus noise) as a function of stimulus signal amplitude across a wide dynamic range on an analog-to-analog audio device. Mono or stereo devices may be tested.

**Stimulus:** The Analog Generator is at default conditions of sine waveform, 50 ohms balanced output impedance, frequency 1 kHz, and “resting” open circuit amplitude before and after sweep of 1.000 Volt. The frequency may be changed to any other desired audio frequency, but it should be recognized that the 22 kHz low pass filter will attenuate higher harmonics. If the frequency is changed to values below 80 Hz, note the Analyzer Reading Rate relationship described below.

**Measurement:** The reading meter of the Analog Analyzer is in THD+N Amplitude function with the high and low pass filters set to the 22 Hz to 22 kHz range. Note that the 22 kHz low-pass filter will attenuate second harmonic products from fundamentals above 11 kHz, third harmonics of fundamentals above 7 kHz, etc. If the device under test is not already inherently band-limited in the 20 kHz area and if the generator frequency is changed to a value above 7 kHz, it may be desirable to change the low-pass filter setting to 80 kHz in order to measure harmonics properly.

**Sweep and Display:** The sweep is 30 2-dB steps (31 values) from 0 dBV to -60 dBV, sweeping from high to low amplitude. Distortion is expressed in absolute units (dB relative to one volt) rather than relative to the present signal amplitude. The resulting graph will approximate a straight horizontal line below the clipping level of most typical devices. The graph Y axis is calibrated from 0 dBV to -120 dBV, covering the range of most measurements. For better display of data after measurement of a particular device, use the Optimize function (click right mouse button on graph for a list of graph-related commands).

**Stereo:** Both generator Channels are on but only Channel A will be swept, measured, and plotted as the test is set up. To test a stereo device, click the Stereo Sweep checkbox at the bottom of the Sweep panel on Page 2. Since the reading meter can only measure one channel at a time, the stereo sweep will first sweep and plot Channel A, then switch analyzer channels automatically and sweep and plot Channel B onto the same graph.

## **SMPTE IMD vs. amplitude**

---

### **A-ASMPTE.AT1**

The IMD option is required for this test.

**Purpose:** This test is set up to measure and graph SMPTE or DIN IMD (intermodulation distortion) as a function of stimulus signal amplitude on an analog-to-analog audio device. Mono or stereo devices may be tested.

**Stimulus:** The Analog Generator configuration is at default conditions (50 ohms balanced output impedance). The SMPTE 4:1 waveform is selected with frequency set to 7 kHz and IM Freq to 60 Hz. This waveform selection generates two sine waves, with the 60 Hz (IM Freq) sinewave amplitude being four times the amplitude of the high frequency sinewave. To test to the German DIN standard, change the main generator frequency from 7 kHz to 8 kHz and the IM Freq from 60 Hz to 250 Hz. The “resting” open circuit amplitude (before and after sweep) is 1.000 Volt sinewave equivalent peak; that is, the peak-to-peak amplitude of the two-tone signal is equal to the peak-to-peak amplitude of a sinewave of 1.000 V rms.

**Measurement:** The reading meter of the Analog Analyzer is in SMPTE function. This function measures the amplitude of demodulated sidebands around the high-frequency “carrier”.

**Sweep and Display:** The sweep is 30 2-dB steps (31 values) from zero dBV to -60 dBV, sweeping from high to low amplitude. The graph Y axis runs from 0 to -120. For better display of data after measurement of a particular device, use the Optimize function (click right mouse button on graph for a list of graph-related commands).

**Stereo:** Both generator Channels are on but only Channel A will be swept, measured, and plotted as the test is set up. To test a stereo device, click the Stereo Sweep checkbox at the bottom of the Sweep panel on Page 2. Since the reading meter can only measure one channel at a time, the stereo sweep will first sweep and plot Channel A, then switch analyzer channels automatically and sweep and plot Channel B onto the same graph.

## **CCIF IMD vs. amplitude**

---

### **A-A\_CCIF.AT1**

The IMD option is required for this test.

**Purpose:** This test is set up to measure and graph CCIF (also called twin-tone or difference-tone) IMD (intermodulation distortion) as a function of stimulus signal amplitude on an analog-to-analog audio device. Mono or stereo devices may be tested.

**Stimulus:** The Analog Generator configuration is at default conditions (50 ohms balanced output impedance). The CCIF waveform is selected with frequency set to 13.5 kHz (center frequency of the two-tone pair) and IM Freq (tone pair spacing) to 1.0 kHz. This waveform selection generates two equal-amplitude sine waves, spaced apart in frequency by the IM Freq value and centered at the Frequency value. Thus, the sinewaves will be located at 13.0 kHz and 14.0 kHz with these settings. The “resting” open circuit amplitude (before and after sweep) is 1.000 Volt sinewave equivalent peak; that is, the peak-to-peak amplitude of the two-tone signal is equal to the peak-to-peak amplitude of a sinewave of 1.000 V rms.

**Measurement:** The reading meter of the Analog Analyzer is in CCIF function. The System One CCIF IM analyzer measures only the amplitude of the difference product ( $f_2 - f_1$ ), which falls at one kHz in this case.

**Sweep and Display:** The sweep is 25 2-dB steps (26 values) from zero dBV to -50 dBV, sweeping from high to low amplitude. The graph Y axis extends from zero dB to -130 dB. For better display of data after measurement of a particular device, use the Optimize function (click right mouse button on graph for a list of graph-related commands).

**Stereo:** Both generator Channels are on but only Channel A will be swept, measured, and plotted as the test is set up. To test a stereo device, click the Stereo Sweep checkbox at the bottom of the Sweep panel on Page 2. Since the reading meter can only measure one channel at a time, the stereo sweep will first sweep and plot Channel A, then switch analyzer channels automatically and sweep and plot Channel B onto the same graph.

## **DIM IMD vs. amplitude**

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### **A-A\_DIM.AT1**

The IMD option is required for this test.

**Purpose:** This test is set up to measure and graph DIM (dynamic) or TIM (transient) intermodulation distortion as a function of stimulus signal amplitude on an analog-to-analog audio device. Mono or stereo devices may be tested.

**Stimulus:** The Analog Generator configuration is at default conditions (50 ohms balanced output impedance). The DIM 30 waveform is selected which generates a 3.15 kHz squarewave and a sinewave, with the peak-to-peak amplitude of the sinewave one-fourth that of the squarewave. The “30” connotes that the signal is passed through a 30 kHz low pass filter in the generator. The sinewave frequency is controlled by the Frequency field and is set to 15 kHz. The “resting” open circuit amplitude (before and after sweep) is 1.000 Volt sinewave equivalent peak; that is, the peak-to-peak amplitude of the complex signal is equal to the peak-to-peak amplitude of a sinewave of 1.000 V rms.

**Measurement:** The reading meter of the Analog Analyzer is in DIM function. The System One DIM IM analyzer measures the amplitude of the spectrum between 750 Hz and 2.4 kHz, which will include fifth and sixth order intermodulation products of the sinewave and squarewave.

**Sweep and Display:** The sweep is 25 2-dB steps (26 values) from zero dBV to -50 dBV, sweeping from high to low amplitude. The graph Y axis extends from 0 dB at the top to -130 dB at the bottom. For better display of data after measurement of a particular device, use the Optimize function (click right mouse button on graph for a list of graph-related commands).

**Stereo:** Both generator Channels are on but only Channel A will be swept, measured, and plotted as the test is set up. To test a stereo device, click the Stereo Sweep checkbox at the bottom of the Sweep panel on Page 2. Since the reading meter can only measure one channel at a time, the stereo sweep will first sweep and plot Channel A, then switch analyzer channels automatically and sweep and plot Channel B onto the same graph.

## **Interchannel phase vs. frequency**

### **A-APHASE.AT1**

**Purpose:** This test is set up to measure and graph interchannel phase difference of an analog-to-analog stereo or other two-channel audio device as a function of frequency.

**Stimulus:** The Analog Generator is at default conditions (sine waveform, 50 ohms balanced output impedance, 1.000 Volt open circuit amplitude, “resting” frequency before and after sweep of 1 kHz). Both Channels are on.

**Measurement:** The Phase meter measures phase difference between the signals at the two analyzer input connectors. It is fully-specified for accuracy only with signal amplitudes of 10 mV and above on both channels.

**Sweep and Display:** The sweep is 50 steps (51 values) logarithmically-spaced from 20 kHz to 20 Hz, sweeping from high to low. The graph Y axis covers the +/-45 degree range. If values on the

particular device under test plot off the top or bottom of the graph, use the Optimize function (click right mouse button on graph for a list of graph-related commands).

Stereo: Interchannel phase is intrinsically a measurement made only on stereo or two-channel devices.

## Input/Output phase

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### A-APHAIO.AT1

Purpose: This test is set up to measure and graph input-to-output phase shift of an analog-to-analog audio device.

Stimulus: The Analog Generator is at default conditions (sine waveform, 50 ohms balanced output impedance, 1.000 Volt open circuit amplitude, “resting” frequency before and after sweep of 1 kHz). Both Channels are on.

Measurement: Channel A of the Analog Analyzer is driven by the front-panel XLR input connector, but Channel B is set to “GenMon” which is an internal cable connection from the Channel B output of the Analog Generator. The Phase meter measures phase difference between the reference signal (generator Channel B output) and the Channel A XLR input, which is thus phase at the output of the device under test relative to the device input. The Phase meter is fully-specified for accuracy only with signal amplitudes of 10 mV and above on both channels.

Sweep and Display: The sweep is 50 steps (51 values) logarithmically-spaced from 20 kHz to 20 Hz, sweeping from high to low. The graph Y axis covers the  $\pm 45$  degree range. If values on the particular device under test plot off the top or bottom of the graph, use the Optimize function (click right mouse button on graph for a list of graph-related commands). If the device under test has a large amount of phase shift (time delay) between input and output, it may be necessary to increase the number of steps, to sweep from low to high, and to use a linear rather than logarithmic sweep in order to avoid phase curve discontinuities due to phase making one or more complete rotations in the frequency interval between two measurements.

Stereo: To make of graph of input-output phase of both channels of a stereo device requires first running this test as set up, then changing the Analog Analyzer Channel A input selection from Input to GenMon and Channel B from GenMon to Input, then clicking the Append checkbox and running the test again.

## Linearity

### A-A\_LIN.AT1

Purpose: This test is set up to measure and graph deviation from perfect input-output linearity of an analog-to-analog mono or stereo audio device.

Stimulus: The Analog Generator is at default conditions (sine waveform, 50 ohms balanced output impedance, frequency 1 kHz, and “resting” open circuit amplitude before and after sweep of 1.000 Volt).

Measurement: The reading meter of the Analog Analyzer is in Bandpass function with the Bandpass filter frequency slaved to the Analog Generator frequency via the AGen Track selection of the BP/BR Filter Frequency field. Bandpass function is used to produce a frequency-selective amplitude measurement that rejects noise across most of the spectrum in order to measure the generator signal amplitude accurately even at amplitudes below the wide-band noise level.

Sweep and Display: The sweep defined by Source 1 on the Sweep panel consists of 60 steps of two-dB size from +20 dBV to -100 dBV. The intent is to test a typical audio device from near or below its noise level to above its clipping level, and Start and Stop values may be changed as required by a particular device. The reading meter measurements are plotted at Data 1 (and Data 3 if stereo) in dBG units. This unit expresses the measured value in decibels relative to the present value of generator amplitude. A perfectly linear device will result in a flat horizontal trace across the entire range. A trace at zero dBG indicates a unity (x1) gain device. If the device has more than ten dB gain or loss, the trace will go off the graph top or bottom which are set for +/-10 dBG. Use of the right mouse button Optimize command

following the end of a sweep will result in displaying the full range of measurements.

**Stereo:** Both generator Channels are on but only Channel A will be swept, measured, and plotted as the test is set up. To test a stereo device, click the Stereo Sweep checkbox at the bottom of the Sweep panel on Page 2. Since the reading meter can only measure one channel at a time, the stereo sweep will first sweep and plot Channel A, then switch analyzer channels automatically and sweep and plot Channel B onto the same graph.

## Crosstalk vs. frequency

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### A-AXTALK.AT1

**Purpose:** This test measures and graphs crosstalk attenuation or stereo separation from one channel of an analog-to-analog audio device into another channel, as a function of frequency.

**Stimulus:** The Analog Generator is at default conditions (sine waveform, 50 ohms balanced output impedance, 1.000 Volt open circuit amplitude, “resting” frequency before and after sweep of 1 kHz), but only the Channel A output is on. If the device input is unbalanced, it is particularly critical for crosstalk measurements that the generator output configuration also be changed to unbalanced.

**Measurement:** The reading meter of the Analog Analyzer is in Crosstalk function, measuring the signal connected to Analyzer Channel B and expressing it in dB relative to the Level meter reading on Channel A. The analyzer thus measures the relative signal feed-through between channels. Crosstalk function uses the reading meter bandpass filter in order to reject wide-band noise and accurately measure low levels of crosstalk signal. The bandpass filter frequency is controlled by the Sweep Track selection in the BP/BR Filter Frequency field. This causes the filter to track the analog generator frequency since that is the Source 1 instrument and parameter.

**Sweep and Display:** The sweep is 30 steps (31 values) logarithmically-spaced from 20 kHz to 20 Hz, sweeping from high to low. The graph Y axis is logarithmic from -40 dB at the top to -140 dB at the bottom. For better display of data after measurement of a



particular device, use the Optimize function (click right mouse button on graph for a list of graph-related commands).

**Stereo:** This test is set up as a Stereo Sweep by the checkbox at the bottom of the Sweep panel, since crosstalk testing implies a two-channel device. Since the reading meter can only measure one channel at a time, the stereo sweep will first sweep while driving Channel A and plotting Channel B, then will switch generator and analyzer channels automatically and drive B while plotting A onto the same graph.

## FFT spectrum analysis

### A-A\_FFT.AT1

DSP versions required (SYS-222 or SYS-322)

**Purpose:** This test is set up to sample, acquire, and perform an FFT (Fast Fourier Transform) spectrum analysis on one or two channels of an analog output audio device. Stimulus signal may be supplied by System One's generator, but the test may also be used to acquire any audio signal including playback of recordings, broadcast signals, real-time speech or music, etc.

**Stimulus:** There is no intrinsic relation between stimulus and measurement in this test. Signal may be acquired and an FFT performed whether signal is coming from System One's generator or from any other source. If the generator signal is to be used, it is at default conditions (sine waveform, 50 ohms balanced output impedance, 1.000 Volt open circuit amplitude, and 1 kHz frequency) with both Channel A and B outputs on.

**Measurement:** The Spectrum Analyzer DSP program (FFTSLIDE.AZ1) is selected on the digital analyzer panel. Its analog-to-digital converters are fed from the two input channels of the Analog Analyzer, following the input ranging circuitry and balanced-to-unbalanced conversion. A Maximum sample acquisition and FFT is selected. This will cause acquisition of 30,720 samples with the FFT computed from 16k samples. The selected sample rate of 48 kHz produces a measurement bandwidth in excess of 20 kHz. If wider bandwidth, up to 80 kHz, is desired, change the Rate field of the

Digital I/O panel from 48kHz to 192 kHz and change the Source 1 Stop value to 80 kHz.“

Sweep” and Display: FFT channels 1 and 2, fed from analog input channels A and B, are set up as Data 1 and Data 2 selections. The +20 to -120 dBV graph top and bottom selected will cover the range of most signals and the Optimize command of the graph right mouse button dialog may be selected to provide better graphical resolution. Source 1 selects frequency domain (spectrum analysis) and plots the zero to 24 kHz range on a linear axis. Smaller portions of this range may be selected by zooming with the mouse or changing the Source 1 Start and Stop values. A logarithmic horizontal axis may be selected if both Start and Stop are positive values.

Stereo: As set up, the test measures both channels simultaneously. If signal is present on only one channel, the other trace may be turned off either by selecting “None” instead of the FFT channel at the Data 1 or Data 2 Browser, or by disabling the trace in the first column of the graph legend.

## Waveform display

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### **A-A\_WAVE.AT1**

DSP versions required (SYS-222 or SYS-322)

Purpose: This test is set up to sample, acquire, and display the waveform of one or two channels of an analog output audio device. Stimulus signal may be supplied by System One’s generator, but the test may also be used to acquire any audio signal including playback of recordings, broadcast signals, real-time speech or music, etc.

Stimulus: There is no intrinsic relation between stimulus and measurement in this test. Signal may be acquired and an FFT performed whether signal originates from System One’s generator or from any other source. If the generator signal is to be used, it is at default conditions (sine waveform, 50 ohms balanced output impedance, 1.000 Volt open circuit amplitude, and 1 kHz frequency) with both Channel A and B outputs on.

**Measurement:** The Spectrum Analyzer DSP program (FFTSLIDE .AZ1) is selected on the digital analyzer panel. Its analog-to-digital converters are fed from the two input channels of the Analog Analyzer, following the input ranging circuitry and balanced-to-unbalanced conversion. A Maximum (30,720 sample) acquisition is selected. The selected sample rate of 48 kHz produces a measurement bandwidth in excess of 20 kHz. If wider bandwidth, up to 80 kHz, is desired, change the Rate field of the Digital I/O panel from 48 kHz to 192 kHz.“

**Sweep” and Display:** FFT channels 1 and 2, fed from analog input channels A and B, are set up as Data 1 and Data 2 selections. Presuming a measured signal amplitude around one Volt RMS, the graph top and bottom are set to +6 and -2 Volts at Data 1 and +2 and -6 Volts at Data 2 to separate the two traces in dual-trace oscilloscope fashion. If it is desired to put the two channel traces on top of one another for comparison, Data 1 and Data 2 Top and Bottom values should be the same. The Optimize Together command of the graph right mouse button dialog will accomplish this automatically and will optimize the display for signals of significantly higher or lower amplitude than one Volt RMS. Source 1 selects time domain (oscilloscope mode) and plots the zero to five millisecond range on a linear axis. Smaller portions of this range may be selected by zooming with the mouse or changing the Source 1 Start and Stop values. The full record length is approximately 640 milliseconds (30,720 samples/48,000 samples per second) and the Source 1 Start and Stop values may be set anywhere in the 0-640 ms range to view the desired portion of the acquired record. The time span between Start and Stop values may be set as desired and is typically equal to several periods of the signal when viewing a sinewave.

**Stereo:** As set up, the test measures both channels simultaneously. If signal is present on only one channel, the other trace may be turned off either by selecting “None” instead of the FFT channel at the Data 1 or Data 2 Browser, or by disabling the trace in the first column of the graph legend.

## External frequency response

### A-AEXTRS .AT1

**Purpose:** This test is set up to measure and graph frequency response of an analog output device when the test signal is not under APWIN software control, such as during playback of a test tape or disc containing a series of tones. Mono or stereo devices may be tested.

**Stimulus:** This test is set up to measure frequency response from a frequency-stepped series of tones from an external, uncontrollable source such as playback of a tape or disk recording or reception of signal from a distant oscillator. All generator settings are thus irrelevant.

**Measurement:** The Amplitude function of the reading meter (no filters, RMS detector) is the measuring instrument.

**Sweep and Display:** For an external sweep, Source 1 on the Sweep panel is selected as a measurement (the Analog Analyzer Channel A Frequency counter in this case) rather than a setting as is common for most sweeps.

The Start value of 20 Hz and Stop value of 20 kHz indicate that the expected sweep direction will be from low to high; they must be reversed if the actual sweep proceeds from high to low frequency. The values also determine the graph extremes, but measurements will be taken even if the value is outside either extreme and the graph end values may be changed afterwards to display all the data.

The Spacing value of 5% means that a new measurement will be acquired and plotted each time the incoming frequency changes by at least 5% from its previous value. If the frequency steps of the external source are smaller than 5%, the Spacing value should be reduced in order to make measurements at every value.

The Min Lvl (minimum level) value of -30 dBV (about 30 millivolts) causes the plotting to pause if the measured signal level drops below that value, as it may between tracks of a test disk or tape. This value should be set higher than the noise level from the device under test when in-between tracks.

The End On value of 1 kHz will cause the test and plot to be automatically terminated if the external signal goes to a 1 kHz value following its series of frequency steps across the range. If automatic termination is desired, it should be set to the frequency value of the tone that follows the last step of the sweep.

Stereo: The Analog Analyzer Amplitude function of the Reading meter is plotted at Data 1. If a stereo device is to be tested, check the Stereo Sweep box at the bottom of the Sweep panel. The Amplitude reading will then be plotted at Data 3. At each new frequency step of the external signal, channel A will be measured and plotted, the input channels will be switched, channel B will be measured and plotted, and the channel will switch back to channel A to wait for the frequency to change.

### **External THD + N vs. frequency**

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#### **A-AEXTTH . AT1**

Purpose: This test is set up to measure and graph THD+N (total harmonic distortion plus noise) of an analog output device when the test signal is not under APWIN software control, such as during playback of a test tape or disc containing a series of tones. Mono or stereo devices may be tested.

Stimulus: This test is set up to measure from a frequency-stepped series of tones from an external, uncontrollable source such as playback of a tape or disk recording or reception of signal from a distant oscillator. All generator settings are thus irrelevant.

Measurement: The reading meter of the Analog Analyzer is in THD+N Relative function with the high and low pass filters set to the 22 Hz to 22 kHz range. Note that the 22 kHz low-pass filter will attenuate second harmonic products from fundamentals above 11 kHz, third harmonics of fundamentals above 7 kHz, etc. If the device under test is not already inherently band-limited in the 20 kHz area, it may be desirable to change the low-pass filter setting to 80 kHz in order to measure up through the fourth harmonic at the 20 kHz end of the sweep range.

**Sweep and Display:** For an external sweep, Source 1 on the Sweep panel is selected as a measurement (the Analog Analyzer Channel A Frequency counter in this case) rather than a setting.

The Start value of 20 Hz and Stop value of 20 kHz indicate that the expected sweep direction will be from low to high; they must be reversed if the actual sweep proceeds from high to low frequency. The values also determine the graph extremes, but measurements will be taken even if the value is outside either extreme and the graph end values may be changed afterwards to display all the data.

The Spacing value of 5% means that a new measurement will be acquired and plotted each time the incoming frequency changes by at least 5% from its previous value. If the frequency steps of the external source are smaller than 5%, the Spacing value should be reduced in order to make measurements at every value.

The Min Lvl (minimum level) value of -30 dBV (about 30 millivolts) causes plotting to pause if the signal level drops below that value. This value should be set higher than the noise level from the device under test when in-between tracks.

The End On value of 1 kHz will cause the test and plot to be automatically terminated if the external signal goes to a 1 kHz value following its series of frequency steps across the range. If automatic termination is desired, “End On” should be set to the frequency value of the tone that follows the last step of the sweep.

**Stereo:** The Analog Analyzer in THD+N relative function is plotted at Data 1. If a stereo device is to be tested, check the Stereo Sweep box. Channel B will then be plotted at Data 3. Since the reading meter can only measure one channel at a time, at each new frequency step of the external signal channel A will be measured, the input channels switched, channel B measured, and the input switched back to channel A. This pre-supposes that the dwell time of the external signal at each frequency is long enough to permit settled distortion measurements on each channel in sequence.

## Wow & flutter vs. time

### A-A\_W&F.AT1

The Wow & Flutter option is required for this test.

**Purpose:** This test is set up to measure wow and flutter of the signal from a tape or disk player. The test measures wow and flutter for a period of time and graphs it in chart recorder style. At the end of the measurements, the test also computes the “2 sigma” value of all the measurements in order to provide a single number that may be compared to a specification.

**Stimulus:** Wow and flutter measurements should be made while a tape or disk player is reproducing a previously-recorded reference signal, normally either 3.15 kHz or 3.00 kHz. This is usually a purchased wow and flutter reference tape or disk which was recorded on a high-quality machine maintained and adjusted to have as little wow and flutter as possible. Some people with access to a high-quality recorder prefer to record their own wow and flutter test tapes. For such a case, this test is set up with the Analog Generator in high-accuracy mode, generating a 3.15 kHz sinewave. Actual measurements should not be made while recording, but only after rewinding and while playing the recording. Thus, the generator settings are irrelevant when measuring and the generator outputs are typically not connected to the device under test when measuring wow and flutter.

**Measurement:** The Analog Analyzer reading (main) meter in Wow & Flutter function makes the measurement. This test is set up with the NAB detector selection and Weighted measurement selected, but depending upon the standard being followed the IEC/DIN or JIS detectors may be selected, weighted or unweighted.

**Sweep and Display:** The test is set up to display wow and flutter versus time, as a chart recorder would. A ten second period is plotted. The graph starts at 5 seconds and continues to 15 seconds. Transients occurring during the first 5 seconds are thus ignored; the wow and flutter detectors have such long time constants that it takes several seconds for the measured value to settle out following the application of a new signal. The stream of measurements from the W&F meter is

plotted at both Data 1 and Data 2 in order to use the Compute 2-Sigma function on Data 2 at the end of the test without losing the original measurements at Data 1. At the end of 15 seconds, the Compute 2-Sigma command (Compute Menu) is automatically invoked for Data 2 and the Data 2 measurements are replaced with a single value (horizontal straight line) which is the value which was exceeded by 5% of the readings.

Stereo: Wow and flutter is normally only measured on one channel even if the tape or disk player is stereo, since identical measurements would be expected on either channel.

### **Locating -3 dB Point**

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#### **A-A\_3dB.AT1**

Purpose: This test is set up to automatically locate the upper or lower minus three dB response point of an analog input-analog output monaural or stereo device, using the Regulation function of APWIN.

Stimulus: The Analog Generator is at default conditions (sinewave, 50 Ohms balanced output impedance, 1.000 Volt open circuit amplitude, “resting” frequency of 1 kHz.

Measurement: Press the F4 function key to set the dBr reference value to the present measured value at 1 kHz. As stored, the Regulation function has the “-Normal” Operation mode selected. This mode assumes An inverse relationship between the regulated parameter (Analog Analyzer Level A meter amplitude) and the “By Varying” parameter (Analog Generator frequency). An inverse relationship exists at the upper band limit of a device. Press F9 or click on the green light icon on the toolbar and the Regulation function will adjust the generator frequency to the value that produces a -3.0 dBr measurement, if such a point exists between 1 kHz and 204 kHz. To find the minus three dB point at the lower band limit of a device, change the “Operation” selection to “+Normal” and press F9.“

Sweep” and Display: This test is set up for a single-point “sweep” and therefore produces a tabular display rather than a graph. Column 2 of the Data Editor shows the final amplitude measurement, which



should be  $-3.00 \text{ dB} \pm 0.05 \text{ dB}$ . Column 3 shows the measured frequency at that amplitude.

Stereo: As stored, the test will determine the minus 3 dB frequency of the device connected to channel A. Testing a stereo device requires two consecutive tests since the Regulation function does not support stereo testing with System One.

## Measuring Power Bandwidth

### A-APWRBD.AT1

Purpose: This test uses the Regulation function to determine how much power an amplifier can furnish at various frequencies across the audio spectrum, at an acceptable amount of distortion.

Stimulus: The Analog Generator is set to resting values of 1.000 volt amplitude and 1 kHz in frequency, but both the values will be changed during the course of the test. The generator output configuration is set to Unbalanced Floating 25 ohms impedance since the test was set up with a power amplifier with unbalanced input, but can be changed to Balanced if testing a balanced input device.

Measurement: The Analog Analyzer measures Level (in Watts) and THD+N at the output of the device under test. The Regulation function is set up to vary the generator amplitude as required to achieve a measured distortion value of 0.5% at each step of the sweep. This is accomplished with the “+Normal” selection in the “Operation” field of the Regulation panel. The +Normal selection tells the Regulation function to expect distortion to increase as the generator amplitude increases, which is normal behavior near the clipping point of a device.

Sweep and Display: Source 1 sweeps the generator frequency from 200 kHz down to 20 Hz; these values may be changed if they are not appropriate for the device under test. Amplifier output power in watts, measured with the Level meter, is plotted as Data 1. Note that the watts unit calibration must “know” the actual value of load impedance on the device being tested. The Reference Watts field at the bottom of the Analog Analyzer panel is set to 4.00 ohms as this test was stored and must be changed if another load impedance value is used.

Distortion (THD+N) is not plotted (but could be) since it will be a horizontal straight line at 0.5% due to the action of the Regulation function.

**Stereo:** This test is set up for a monaural device connected to Channel A. A stereo device must be tested one channel at a time (although the alternate channel may be driven at any desired fixed level to obtain desired loading of shared power supplies) since the Regulation function does not support stereo testing with System One.

## **Equalized Phono Response**

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### **A-A\_RIAA.AT1**

**Purpose:** This test is set up to measure the deviation from perfect RIAA equalization of a phonograph preamplifier.

**Stimulus:** The Analog Generator is set to EQ Sine (equalized sinewave) waveform. The selected equalization file is RIAA-PRE.ADQ (located in the C:\APWIN\EQ directory). Resting frequency is 1 kHz and output amplitude is set to 10 mV, a typical value for phono preamplifier input sensitivity. The generator output configuration is set to Unbalanced Float since most phono preamps have unbalanced inputs. Generator output impedance is 25 ohms and both channels are turned on.

**Measurement:** The Analyzer Amplitude (2-Ch) meter, measuring Channel A, is plotted at Data 1 in dBr units. The Analyzer Level B meter is plotted at Data 3 and thus uses the same unit. Press the F4 key before starting the sweep, to set the dBr reference to the measured Channel A amplitude value so the Channel A curve will go through zero dBr at 1 kHz.

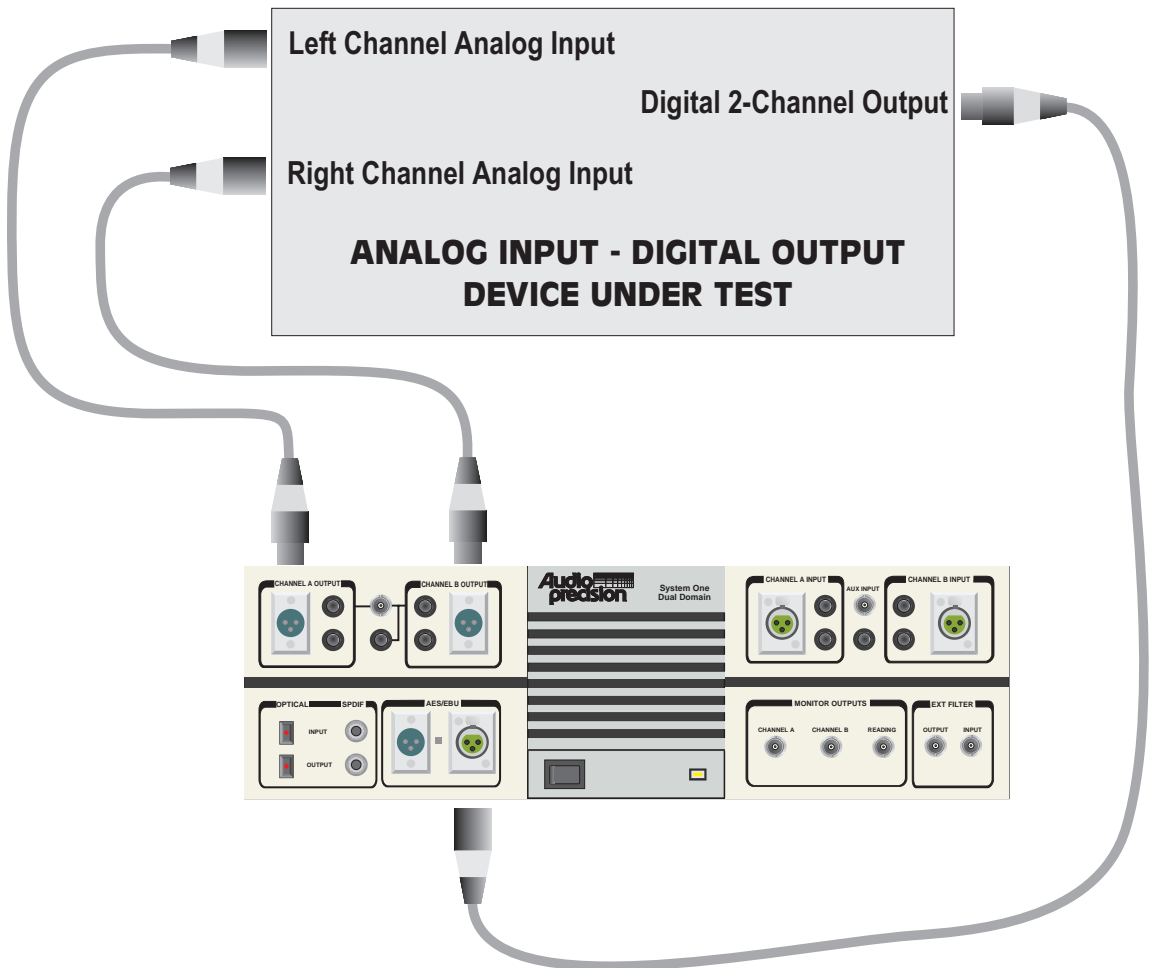
**Sweep and Display:** At each step of the sweep, the EQ Sine function and the attached equalization file will change the actual generator output amplitude to follow the shape of the standard RIAA phono preamp preemphasis curve. If the phono preamp under test has perfect equalization (deemphasis) to the RIAA curve, the result will be a perfectly flat trace. Deviations from flatness of the trace are errors in the preamplifiers' equalization curve.

Stereo: The test is set up as a stereo test since virtually all phono preamplifiers are stereo devices. If a monaural device is to be tested, uncheck the Stereo Sweep checkbox at the bottom of the Sweep panel.

## Analog-to-Digital Tests

An SYS-322 (Dual Domain) unit is required for all tests in this section. A standard installation places System One analog-to-digital tests in the **C:\APWIN\S1\A-D** subdirectory.

Generator Output Configuration: All the tests described in this analog-to-digital section are set up using System One's balanced



floating output configuration, which drives XLR and double-banana output connectors in parallel. If devices with unbalanced inputs are to be tested, the output configuration of each test used should be changed to unbalanced. In unbalanced modes, the XLR or double banana connectors, may be used. Floating or grounded configurations may be selected as required by the application. All tests are set up with the 50 ohm source impedance at Z-Out; one of the other available impedances may be selected as required.

**Digital Analyzer Input Configuration:** All the tests described in this analog-to-digital section are set up expecting an AES/EBU or SPDIF format digital input signal connected to either the XLR, phono, or optical digital input connector. Switching among these three input connectors is performed automatically when signal is sensed.

**Audio Frequency Range vs. Sample Rate, Digital Input:** All the tests described in this analog-to-digital section are set up with the presumption that the device under test has a sample rate of at least 48 kHz. System One automatically locks to the actual digital sample rate, but for correct display of the imbedded audio signal frequency, the Rate field on the DIO panel should correspond to the actual input sample rate (48 kHz, 44.1 kHz, or 32 kHz). If the device has a lower rate and no changes have been made to the test, many of the tests as provided will fail with an error message of the form “Conflict with maximum DSP setting value while setting XXX (Source 1 parameter) to nnn kHz.” To adapt a test to any specific sample rate, change the upper frequency value at Source 1 on the Sweep Panel from its stored value (typically either 20 kHz or 24 kHz) to a value not greater than  $\frac{1}{2}$  the sample rate.

## Frequency response

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### **A-D\_RESP.AT1**

**Purpose:** This test is set up to measure and graph frequency response on an analog input, digital output device such as an A/D converter. Mono or stereo devices may be tested, as described below

**Stimulus:** The Analog Generator is at default conditions of sinewave, 50 ohms balanced output impedance, and “resting” frequency before and after sweep of 1 kHz. The generator Amplitude is

set at 0 dBr with the dBr Reference at 10 Volts which was the full-scale value of a particular A/D converter being used; it should be changed as necessary to an appropriate value for the device under test.

**Measurement:** The digital domain level measurement of the channel A digital signal is made by the main (reading) meter of the Digital Domain Audio Tester DSP program (GENANLR . AZ1). If stereo mode is selected, the channel B digital signal is simultaneously measured by the Level meter of the same program.

**Sweep and Display:** The sweep is 50 steps (51 values) logarithmically-spaced from 24 kHz to 10 Hz, sweeping from high to low. The graph Y axis covers the +2 to -10 dBFS range (dB relative to digital full scale). If values on the particular device under test plot off the top or bottom of the graph, use the Optimize function (click right mouse button on graph for a list of graph-related commands).

**Stereo:** Both generator channels are on but only channel A will be measured and plotted as the test is set up. To test a stereo device, click the Stereo Sweep checkbox at the bottom of the Sweep panel on Page 2. The Level meter will then simultaneously plot at Data 3.

## Noise (single point)

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### A-DNOISE . AT1

**Purpose:** This test is set up to measure noise, integrated across the audio spectrum, on a mono or stereo analog-to-digital audio device such as an A/D converter. The result is a single number (per channel), not a graph.

**Stimulus:** The Analog Generator output is off, so waveform and amplitude are irrelevant. However, the generator output impedance serves as a defined back termination for the device input if connected. The generator output configuration is at default conditions of 50 ohms impedance, balanced. Other impedances may be required by some noise-measurement specifications.

**Measurement:** The main (reading) meter of the Digital Domain Audio Tester DSP program (GENANLR . AZ1) is in Noise Unweighted function using the RMS detector. The Noise A-Weighted or Noise

CCIR-Weighted functions may be selected instead to provide weighted measurements. The CCIR quasi-peak (Q-Pk) detector may be used with the Noise CCIR-Weighted function for a CCIR 468 weighted noise measurement, by selecting “4/sec QPK” in the Reading Rate field of the digital analyzer panel.

**Sweep and Display:** A single-point measurement is made (no sweep) and tabular results are displayed in the Data Editor since a graph cannot be made of a single point. The reason for selecting Generator Amplitude A as the Source 1 parameter, with -999 dBV as the Start value, is merely to create a Column heading and value in column one of the Data Editor which helps indicate that this is a noise measurement (no signal). Since the generator output is turned off on the generator panel, the actual Source 1 parameter and setting is not important.

**Stereo:** To test a stereo device, click the Stereo Sweep checkbox at the bottom of the Sweep panel on Page 2. Since the reading meter can only measure one channel at a time, the two channels will be automatically be measured in sequence with results displayed in columns two and three of the Data Editor.

## **THD + N vs. frequency**

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### **A-DTHDFR.AT1**

**Purpose:** This test is set up to measure and graph THD+N (total harmonic distortion plus noise) as a function of stimulus frequency across the audio band on an analog-to-digital audio device such as an A/D converter. Mono or stereo devices may be tested.

**Stimulus:** The Analog Generator is at default conditions (sine waveform, 50 ohms balanced output impedance, “resting” frequency before and after sweep of 1 kHz). The generator amplitude is set to zero dBr with the dBr Reference at 10 Volts which was the full scale value of the converter used when the test was set up. This value should be changed if necessary to an appropriate value for the particular device under test.

**Measurement:** The main (reading) meter of the Digital Domain Audio Tester DSP program (GENANLR.AZ1) is in THD+N Relative.

**Sweep and Display:** The sweep is 20 steps (21 values) logarithmically-spaced from 20 kHz to 50 Hz, sweeping from high to low. THD+N functions of GENANLR .AZ1 are only specified at 50 Hz and above. The graph Y axis is calibrated in dB below signal level, from zero to -110 dB. For better display of data after measurement of a particular device, use the Optimize function (click right mouse button on graph for a list of graph-related commands).

**Stereo:** Both generator Channels are on but only Channel A will be swept, measured, and plotted as the test is set up. To test a stereo device, click the Stereo Sweep checkbox at the bottom of the Sweep panel on Page 2. Since the reading meter can only measure one channel at a time, the stereo sweep will first sweep and plot Channel A, then switch analyzer channels automatically and sweep and plot Channel B onto the same graph.

## **THD + N vs. amplitude**

### **A-DTHDAM .AT1**

**Purpose:** This test is set up to measure and graph THD+N (total harmonic distortion plus noise) as a function of stimulus signal amplitude across a wide dynamic range on an analog-to-digital audio device such as an A/D converter. Mono or stereo devices may be tested.

**Stimulus:** The Analog Generator is at default conditions of sine waveform, 50 ohms balanced output impedance, and frequency 1 kHz. The “resting” open circuit amplitude before and after the sweep is set to zero dBr with the dBr Reference set to 10.0 Volts, since an A/D converter requiring 10 Volts for full scale output was used in setting up the test. The frequency may be changed to any other desired audio frequency above 50 Hz but results will not be meaningful at stimulus frequencies above 6-7 kHz since third and higher order harmonics will be attenuated by the intrinsic band limiting at  $\frac{1}{2}$  sample rate of a digital system.

**Measurement:** The main (reading) meter of the Digital Domain Audio Tester DSP program (GENANLR .AZ1) is in THD+N Absolute function. Absolute THD+N states distortion and noise in absolute terms (% of full scale, dBFS, etc.) rather than relative to the present signal amplitude.

**Sweep and Display:** The sweep is 40 2-dB steps (41 values) from 0 dBr to -80 dBr, sweeping from high to low amplitude. The graph Y axis is calibrated in dBFS (decibels below digital full scale) from zero to -100 dBFS. An ideal A/D converter will be limited only by the intrinsic quantization distortion and noise due to its resolution (word width), independently of signal level. Therefore, an absolute distortion vs. amplitude curve can be expected to be approximately a horizontal straight line across the graph. For better display of data after measurement of a particular device, use the Optimize function (click right mouse button on graph for a list of graph-related commands).

**Stereo:** Both generator Channels are on but only Channel A will be swept, measured, and plotted as the test is set up. To test a stereo device, click the Stereo Sweep checkbox at the bottom of the Sweep panel on Page 2. Since the reading meter can only measure one channel at a time, the stereo sweep will first sweep and plot Channel A, then switch analyzer channels automatically and sweep and plot Channel B onto the same graph.

## **SMPTE IMD (FFT)**

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### **A-DSMPTE.AT1**

The IMD generator option is required for this test.

**Purpose:** This test is set up to make an FFT (Fast Fourier Transform) spectrum analysis graph with a SMPTE or DIN IMD (intermodulation distortion) test signal on an analog-to-digital audio device such as an A/D converter. Mono or stereo devices may be tested.

**Stimulus:** The Analog Generator configuration is at default conditions (50 ohms balanced output impedance). The SMPTE 4:1 waveform is selected with frequency set to 7 kHz and IM Freq to 60 Hz. This waveform selection generates two sine waves, with the 60 Hz (IM Freq) sinewave amplitude being four times the amplitude of the high frequency sinewave. To test to the German DIN standard, change the main generator frequency from 7 kHz to 8 kHz and the IM Freq from 60 Hz to 250 Hz. The open circuit amplitude is zero dBr with the dBr Reference set as 10 Volts since an A/D converter with an input requirement of 10 Volts for full scale digital output was used in setting up the test. Signal amplitude is calibrated in sinewave equivalent peak;



that is, the peak-to-peak amplitude of the two-tone signal is equal to the peak-to-peak amplitude of a sinewave of the stated amplitude.

Measurement: The Spectrum Analyzer DSP program (FFTSLIDE .AZ1) is selected on the digital analyzer panel. Digital input is selected, with digital channel A selected as Channel 1 and channel B as Channel 2. A 16k sample FFT is selected. The digital input will lock to a wide range of sample rates.“

Sweep” and Display: The horizontal axis is set by Source 1 Start and Stop to the audio spectrum between 6 and 8 kHz. This spectrum includes the 7 kHz “carrier frequency” and intermodulation products (sidebands) up to  $\pm 1$  kHz from the carrier. One graph cursor can be placed on the “carrier” and the second cursor placed on any sideband. The “dy” (difference in y coordinate) display at the graph top will then show the amplitude of each intermodulation distortion product. The 2<sup>nd</sup> order IMD products will be seen at  $\pm 60$  Hz from the 7 kHz signal, 3<sup>rd</sup> order IMD products at  $\pm 120$  Hz, etc. If the generator frequency is changed to the German DIN standard 8 kHz/250 Hz frequencies, the Sweep Source 1 Start and Stop frequencies should be changed to display the spectrum between 7 kHz and 9 kHz. In this case, 2<sup>nd</sup> order sidebands will be at  $\pm 250$  Hz and 3<sup>rd</sup> order sidebands at  $\pm 500$  Hz from the 8 kHz carrier.

Stereo: As set up, the test measures both channels simultaneously. If signal is present on only one channel, the other trace may be turned off either by selecting “None” instead of the FFT channel at the Data 1 or Data 2 Browser, or by disabling the trace in the first column of the graph legend.

## CCIF IMD (FFT)

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### A-D\_CCIF.AT1

Purpose: This test is set up to make an FFT (Fast Fourier Transform) spectrum analysis graph with a CCIF (also sometimes called twin-tone or difference tone) IMD (intermodulation distortion) test signal on an analog-to-digital audio device such as an A/D converter. Mono or stereo devices may be tested.

**Stimulus:** The Analog Generator output configuration is at default conditions (50 ohms balanced output impedance). The CCIF waveform is selected with frequency set to 13.5 kHz (center frequency of the two-tone pair) and IM Freq (tone pair spacing) to 1.0 kHz. This waveform selection generates two equal-amplitude sine waves, spaced apart in frequency by the IM Freq value and centered at the Frequency value. Thus, the sinewaves will be located at 13.0 kHz and 14.0 kHz with these settings. The open circuit amplitude is zero dBr with the dBr Reference value set to 10 Volts since the device used in setting up the test was an A/D converter requiring 10 Volts for full-scale digital output. Amplitude calibration of the two-tone signal is in terms of sinewave equivalent peak; that is, the peak-to-peak amplitude of the two-tone signal is equal to the peak-to-peak amplitude of a sinewave of 10 V rms

**Measurement:** The Spectrum Analyzer DSP program (FFTSLIDE .AZ1) is selected on the digital analyzer panel. Digital input is selected, with digital channel A selected as Channel 1 and channel B as Channel 2. A 16k sample FFT is selected. The digital input will lock to a wide range of sample rates. “

**Sweep” and Display:** The horizontal axis is set by Source 1 Start and Stop to the full audio spectrum between zero and 24 kHz. This pre-supposes a 48 kHz or higher sample rate from the A/D unit under test. If the sample rate is lower than 48 kHz, the Source 1 Stop frequency must be changed to a value no higher than  $\frac{1}{2}$  the sample rate. The spectrum displayed will include the test signal tones at 13 kHz and 14 kHz and all intermodulation distortion products falling within the range. The second order IMD product (difference tone) will fall at  $f_2 - f_1$ , or 1 kHz. Third order IMD products fall at  $2f_2 - f_1$  ( $28 - 13 = 15$  kHz) and  $2f_1 - f_2$  ( $26 - 14 = 12$  kHz). The signals visible at 12.5 kHz, 13.5 kHz, and 14.5 kHz are artifacts of the balanced modulator technique used in the IMD generator and not IMD products generated in the device under test. One graph cursor can be placed on either of the test tones and the second cursor placed on any IMD product. The “dy” (difference in y coordinate) display at the graph top will then show the amplitude of each intermodulation distortion product.

**Stereo:** As set up, the test measures both channels simultaneously. If signal is present on only one channel, the other trace may be turned off either by selecting “None” instead of the FFT channel at the Data 1

or Data 2 Browser, or by disabling the trace in the first column of the graph legend.

## **DIM IMD (FFT)**

### **A-D\_DIM.AT1**

The IMD option is required for this test.

**Purpose:** This test is set up to make an FFT (Fast Fourier Transform) spectrum analysis graph with a DIM (dynamic intermodulation) test signal on an analog-to-digital audio device such as an A/D converter. Mono or stereo devices may be tested.

**Stimulus:** The Analog Generator configuration is at default conditions (50 ohms balanced output impedance). The DIM 30 waveform is selected which generates a 3.15 kHz squarewave and a sinewave with the peak-to-peak amplitude of the squarewave four times that of the sinewave. The “30” connotes that the signal is passed through a 30 kHz low pass filter in the generator. The sinewave frequency is controlled by the Frequency field and is set to 15 kHz. The open circuit amplitude is zero dBr with the dBr Reference set to 10 Volts, corresponding to the full-scale input value of the particular A/D converter used in setting up the test. Amplitude calibration of this complex test signal is calibrated in terms of sinewave equivalent peak; that is, the peak-to-peak amplitude of the complex signal is equal to the peak-to-peak amplitude of a sinewave of 10 V rms.

**Measurement:** The Spectrum Analyzer DSP program (FFTSLIDE . AZ ) is selected on the digital analyzer panel. Digital input is selected, with digital channel A selected as Channel 1 and channel B as Channel 2. A 16k sample FFT is selected. The digital input will lock to a wide range of sample rates.

**Sweep” and Display:** The horizontal axis is set by Source 1 Start and Stop to the full audio spectrum between zero and 24 kHz. This pre-supposes a 48 kHz or higher sample rate from the A/D unit under test. If the sample rate is lower than 48 kHz, the Source 1 Stop frequency must be changed to a value no higher than  $\frac{1}{2}$  the sample rate. The spectrum displayed will include the test signal sinewave tone at 15 kHz plus the squarewave fundamental frequency at 3.15 kHz,

squarewave third harmonic at 9.45 kHz, squarewave fifth harmonic at 15.75 kHz, and squarewave seventh harmonic at 22.05 kHz. IMD products may occur at the sum and difference frequencies of all combinations of squarewave fundamental and harmonics with the 15 kHz tone. The particular IMD products which would be measured on an analog-output device by System One's analog IMD analyzer are the fifth order product falling at 2.4 kHz and the sixth order product falling at 750 Hz. One graph cursor can be placed on the 15 kHz "probe" tone and the second cursor placed on the IMD product of interest. The "dy" (difference in y coordinate) display at the graph top will then show the amplitude of that intermodulation distortion product.

Stereo: As set up, the test measures both channels simultaneously. If signal is present on only one channel, the other trace may be turned off either by selecting "None" instead of the FFT channel at the Data 1 or Data 2 Browser, or by disabling the trace in the first column of the graph legend.

### Interchannel phase vs. frequency (multitone)

#### **A-DPHASE.AT1**

Purpose: This test is set up to measure the interchannel phase versus frequency characteristic of an analog input digital output device such as an A/D converter. The test uses multitone technology.

Stimulus: The Analog Generator output configuration is at default conditions (50 ohms balanced output impedance). The DGen waveform selection on the Analog Generator routes the output of the digital generator through a D/A converter to the analog output stage. In the Digital Generator section at the bottom of the Digital Analyzer panel with the Triggered Multitone Tester DSP program (FASTTRIG.AZ1255) loaded, the selected Waveform is ISO31.AGM. This waveform file defines a multitone test signal consisting of 31 equal-amplitude sinewaves, one at approximately each ISO standard 1/3 octave frequency from 20 Hz to 20 kHz. The open circuit amplitude of the analog generator is set to zero dBr with the dBr Reference value set to 10 Volts since the device used in setting up the test was an A/D converter requiring 10 Volts for full-scale digital output. Amplitude calibration of the 31-tone signal is in terms of

sinewave equivalent peak with 1.0 dB headroom; that is, the peak-to-peak amplitude of the complex signal is 1.0 dB less than the peak-to-peak amplitude of a sinewave of 10 V rms.

Measurement: The Triggered Multitone Tester DSP program (FASTTRIG.AZ1) is selected on the digital analyzer panel. Digital input is selected, with digital channel A selected as Channel 1 and channel B as Channel 2. A Maximum (16k sample) acquisition and FFT is selected. Spectrum measurement function, Window None (for synchronous processing), and Free Run Triggering are selected on the digital analyzer panel. The digital input will lock to a wide range of sample rates. “

Sweep” and Display: FASTTRIG FFT Frequency is selected at Source 1 with Start and Stop frequencies of 10 Hz and 24 kHz and a log horizontal display. The 24 kHz value pre-supposes a 48 kHz or higher sample rate from the A/D unit under test. If the sample rate is lower than 48 kHz, the Source 1 Stop frequency must be changed to a value no higher than  $\frac{1}{2}$  the sample rate. Log display is used since the 31 test tones are logarithmically spaced across the audio spectrum. The sweep table FASTTEST.ADS is attached at Source 1, causing phase measurements to be made only at the exact fundamental frequencies of the multitone signal. FASTTRIG Channel 1-2 Phase is selected at Data 1. If the measured phase difference exceeds the  $\pm 2$  degree range at Data 1 Graph Top and Bottom, the Optimize command of the graph right mouse button menu may be used to fit the graph top and bottom to the actual measured data.

Stereo: Interchannel phase is intrinsically a stereo test.

## Linearity

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### A-D\_LIN.AT1

Purpose: This test is set up to measure and graph deviation from perfect input-output linearity of an analog-to-digital mono or stereo audio device such as an A/D converter.

Stimulus: The Analog Generator is at default conditions of sine waveform and 50 ohms balanced output impedance. The frequency is set to 200 Hz. The “resting” open circuit amplitude before and after the

sweep is zero dBr, with the dBr Reference set to 10 Volts since a 10 Volt full scale A/D converter was used when setting up the test.

**Measurement:** The Digital Domain Audio Tester DSP program (GENANLR .AZ1) is selected on the Digital Analyzer panel. Digital Channel A is connected to the reading meter of the Digital Domain Analyzer. The analyzer is in Bandpass function with the Bandpass filter frequency slaved to the Analog Generator frequency via the Anlg Gen selection in the BP/BR Tuning field. Bandpass function is used to produce a frequency-selective amplitude measurement, which rejects noise across most of the spectrum in order to measure the generator signal amplitude accurately even at amplitudes below the wide-band noise level.

**Sweep and Display:** The sweep defined by Source 1 on the Sweep panel consists of 60 steps (61 measurement points) of two-dB size from 0 dBr to -120 dBr. The intent is to test a typical A/D converter from its maximum input level to below the wideband noise level. Start and Stop values or the Analog Generator dBr Reference value may be changed as required by a particular device. The reading meter measurements are plotted at Data 1 (and Data 3 if stereo) in dBFS (decibels relative to digital full scale) units. A perfectly linear device will result in a straight diagonal trace across the entire range with the horizontal and vertical coordinates of each point exactly equal. To make it easier to see small deviations from perfect linearity, the Compute Linearity function is set up to automatically execute following the last point of the sweep. Compute Linearity operates by fitting a best straight line to the data over a certain range (-20 dBr to -40 dBr in this case), then subtracts every measurement point from the perfect straight line. The result will be an approximately straight horizontal line near zero dBFS. Use the right mouse button and select the Optimize Left Only command following the Compute Linearity operation to automatically select a new graph top and bottom to best displaying the deviation from linearity graph. Most good A/D converters can be expected to be very linear from a few dB below full scale down to approximately the noise level. At some point well below the wideband noise level, the curve can be expected to swing upwards since the analyzer measurement is still influenced by noise falling within the bandpass filter range.

Stereo: Both generator Channels are on but only Channel A will be swept, measured, and plotted as the test is set up. To test a stereo device, click the Stereo Sweep checkbox at the bottom of the Sweep panel on Page 2. Since the reading meter can only measure one channel at a time, the stereo sweep will first sweep and plot Channel A as Data 1, then switch analyzer channels automatically and sweep and plot Channel B as Data 3 onto the same graph. The Compute Linearity function is enabled for both Data 1 and Data 3, so the deviation from perfect linearity result for both traces will be produced and both will be Optimized by the graph right button command.

## Crosstalk vs. frequency

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### A-DXTALK . AT1

Purpose: This test measures and graphs crosstalk attenuation or stereo separation from one channel of an analog-to-digital audio device (such as an A/D converter) into another channel, as a function of frequency.

Stimulus: The Analog Generator is at default conditions of sine waveform, 50 ohms balanced output impedance, and “resting” frequency before and after the sweep of 1 kHz. If the device input is unbalanced, it is particularly critical for crosstalk measurements that the generator output configuration also be changed to unbalanced. The Amplitude is set to zero dBr with the dBr Reference at 10 Volts, which was the full-scale value of the A/D converter in use when this test was set up. Only the Channel A output is on.

Measurement: The Digital Domain Audio Tester DSP program (GENANLR . AZ1) is selected on the Digital Analyzer panel. Digital Channel B is connected to the reading meter of the Digital Domain Analyzer, which is in Crosstalk function. Crosstalk function measures the signal connected to Digital Analyzer Channel B (the non-driven channel) and expresses it in dB relative to the Digital Analyzer Level meter reading on Channel A (the driven channel). The analyzer thus measures the relative signal feed-through between channels. Crosstalk function uses the bandpass filter in order to reject wide-band noise and accurately measure low levels of crosstalk signal. The bandpass filter



frequency is slaved to the Analog Generator frequency by the Anlg Gen selection in the BP/BR Tuning field.

**Sweep and Display:** The sweep is 30 steps (31 values) logarithmically-spaced from 20 kHz to 50 Hz, sweeping from high to low. The graph Y axis is logarithmic from zero dB at the top to -120 dB at the bottom. For better display of data after measurement of a particular device, use the Optimize function (click right mouse button on graph for a list of graph-related commands).

**Stereo:** This test is set up as a Stereo Sweep by the checkbox at the bottom of the Sweep panel, since crosstalk testing implies a two-channel device. Since the reading meter can only measure one channel at a time, the stereo sweep will first sweep while driving Channel A and plotting Channel B, then will switch generator and analyzer channels automatically and drive B while plotting A onto the same graph.

## **FFT spectrum analysis**

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### **A-D\_FFT.AT1**

**Purpose:** This test is set up to sample, acquire, and perform an FFT (Fast Fourier Transform) spectrum analysis on one or two channels of a digital output audio device. Stimulus signal may be supplied to analog in-digital out devices such as an A/D converter by System One's generator, but the test may also be used to acquire any digital domain audio signal including playback of recordings, broadcast signals, real-time speech or music, etc.

**Stimulus:** There is no intrinsic relation between stimulus and measurement in this test. Signal may be acquired and an FFT performed whether signal is coming from System One's analog generator or from any other source. If the analog generator signal is to be used, it is at default conditions of sine waveform, 50 ohms balanced output impedance, and 1 kHz frequency. Open circuit amplitude is set to zero dB with the dB Reference value 10 Volts which was the full-scale value of the A/D converter used when this test was set up. Both Channel A and B outputs are on.



**Measurement:** The Spectrum Analyzer DSP program (FFTSLIDE .AZ1) is selected on the digital analyzer panel. Digital input is selected with Digital Channel A connected to FFT Channel 1 and Digital Channel B connected to Channel 2. A 16k sample FFT is selected. The digital analyzer will lock to digital input sample rates over a wide range, but the Rate field on the DIO panel should be set to correspond to the known input sample rate if the displayed frequencies of the imbedded audio signal are to be correct. “

Sweep” and Display: FFT channels 1 and 2, fed from digital input channels A and B, are set up as Data 1 and Data 2 selections. The zero to -140 dBFS (dB relative to digital full scale) graph top and bottom selected will cover the range of most signals. The Optimize command of the graph right mouse button dialog may be selected to provide better graphical resolution following an acquisition. The Source 1 FFT Freq selection chooses frequency domain (spectrum analysis) mode. The zero and 24 kHz Source 1 Start and Stop values are based on the assumption of a 48 kHz or higher sample rate from the device under test. If the sample rate of the device is lower, the Source 1 Stop value must be changed to a number no greater than  $\frac{1}{2}$  the sample rate. A linear axis is chosen. Smaller portions of this range may be selected by zooming with the mouse or changing the Source 1 Start and Stop values following an acquisition. A logarithmic horizontal axis may be selected if both Start and Stop are positive values.

**Stereo:** As set up, the test measures both channels simultaneously. If signal is present on only one channel, the other trace may be turned off either by selecting “None” instead of the FFT channel at the Data 1 or Data 2 Browser, or by disabling the trace in the first column of the graph legend.

## Waveform display

### A-D\_WAVE.AT1

**Purpose:** This test is set up to sample, acquire, and display the waveform of one or two channels of a digital output audio device. Stimulus signal to an analog in-digital out device such as an A/D converter may be supplied by System One’s generator, but the test

may also be used to acquire any digital audio signal including playback of recordings, broadcast signals, real-time speech or music, etc.

**Stimulus:** There is no intrinsic relation between stimulus and measurement in this test. Signal may be acquired and an FFT performed whether signal is coming from System One's analog generator or from any other source. If the analog generator signal is to be used, it is at default conditions of sine waveform, 50 ohms balanced output impedance, and 1 kHz frequency. Open circuit amplitude is set to zero dBr with the dBr Reference value of 10 Volts which was the full-scale value of the A/D converter used when this test was set up. Both Channel A and B outputs are on.

**Measurement:** The Spectrum Analyzer DSP program (FFTSLSLIDE . AZ1) is selected on the digital analyzer panel. Digital input is selected, with digital analyzer Channel 1 fed from Channel A of the digital signal and Channel 2 fed from Channel B. A Maximum (30,720 samples) acquisition is selected. The digital analyzer will lock to digital input sample rates over a wide range, but the Rate field on the DIO panel should be set to correspond to the known input sample rate if the displayed frequencies of the imbedded audio signal are to be correct. “

Sweep” and Display: FFT channels 1 and 2, fed from digital input channels A and B, are set up as Data 1 and Data 2 selections. Presuming a measured signal amplitude of approximately digital full scale, the graph top and bottom are set to display signals  $\pm 1.5$  FFS (fraction of digital full scale). Note the definition of digital full scale is the rms value of a sinewave whose peaks are just touching the most positive and most negative allowable digital codes. A sinewave has a crest factor (peak-to-rms value) of 1.414. Thus, the peaks of a zero dBFS sinewave signal level will touch  $\pm 1.414$  FFS and  $\pm 1.5$  FFS was used to preserve round numbers. The actual Graph Top and Bottom values of +4.5 FFS and -1.5 FFS at Data 1 and +1.5 and -4.5 FFS at Data 2 separate the two traces in dual-trace oscilloscope fashion. If it is desired to put the two channel traces on top of one another for comparison, Data 1 and Data 2 Top and Bottom values should be the same. The Optimize Together command of the graph right mouse button dialog will accomplish this automatically and will optimize the display for signals of significantly lower amplitude than one FFS.

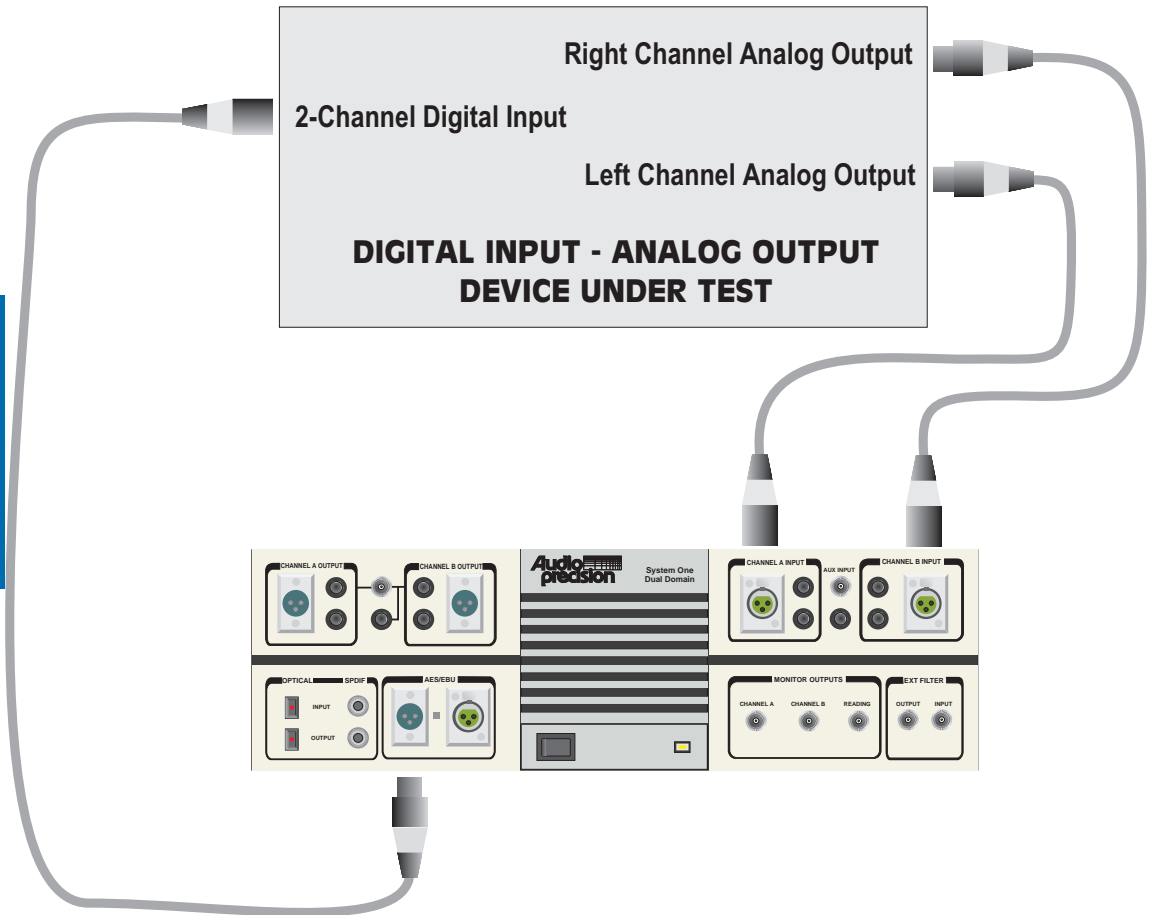
Source 1 selects FFT Time to provide time domain (oscilloscope mode) display rather than spectrum analysis. The zero to five millisecond range is selected at Source 1 Start and Stop to display five cycles if the signal is 1.00 kHz. Smaller portions of this range may be selected by zooming with the mouse or changing the Source 1 Start and Stop values. The full record length is approximately 640 milliseconds (30,720 samples/48,000 samples per second) and the Source 1 Start and Stop values may be set anywhere in the 0-640 ms range to view the desired portion of the acquired record. The time span between Start and Stop values may be set as desired and is typically equal to several periods of the signal when viewing a sinewave.

Stereo: As set up, the test measures both channels simultaneously. If signal is present on only one channel, the other trace may be turned off either by selecting “None” instead of the FFT channel at the Data 1 or Data 2 Browser, or by disabling the trace in the first column of the graph legend.

## Digital-to-Analog Tests

An SYS-322 (Dual Domain) unit is required for all tests in this section. A standard installation places System One digital-to-analog tests in the **C : \APWIN\S1\D-A** subdirectory.

Generator: All the tests described in this digital-to-analog section use System One’s digital generator, which is a function of both GENANLR.AZ1 and FFTGEN.AZ1 (plus several other) Digital Analyzer programs. This generator drives the XLR Bal, RCA phono Unbal, and Optical digital output connectors simultaneously. Most tests in this section are set up with the Resolution field of the DIO Panel set to 24 bits. The Resolution field determines at which bit level (word width) the internally-generated 24-bit signal is rounded, with dither (if enabled) added at the appropriate amplitude for a signal of that resolution. It is generally desirable to change the Resolution value to the known input resolution of the D/A device under test; this is particularly critical when tests are made of the low-amplitude performance of the device such as linearity and noise. All tests are set up with Triangular dither enabled, but Rectangular or Shaped dither or no dither may be selected via the DIO panel.



**Sample Rate, Digital Output:** All the tests described in this digital-to-analog section are set up with the presumption that the device under test is to be tested at a 48 kHz sample rate controlled from System One's internal crystal. To test at 44.1 kHz or 32 kHz, select the desired rate at the Rate field on the DIO panel. If a rate lower than 48 kHz is chosen and no changes have been made to the tests in this section as provided, many of them will fail with an error message of the form "Conflict with maximum DSP setting value while setting XXX (Source 1 parameter) to nnn kHz." To adapt a test to any specific sample rate, change the upper frequency value at Source 1 on the Sweep Panel from its stored value (typically either 20 kHz or 24 kHz) to a value not greater than  $\frac{1}{2}$  the actual sample rate.

Status Bits: All tests in this section are set up to transmit status bits in the Professional format with sample rate and emphasis not indicated on both channels (hex code of 01 00 00 00 for bytes 0-3). Many digital input devices, particularly digital recorders, will not accept a digital input unless the status bits on the digital signal correspond to what the digital product designer thought was appropriate. If the device under test requires other status bits in order to function as desired, the Status Bits panel should be used to select the desired output. This panel may be displayed by clicking on the Status Bits button of the DIO panel, by clicking on the STAT BIT icon on the toolbar, or via the Panels Status Bits menu command. The Professional/Consumer selection is almost always important, and it may also be important that the transmitted Sample Frequency information in the status bits matches the actual Sample Rate being generated by System One. The Sample Frequency information in the status bits is not automatically set when System One sample rate is changed. When Consumer format is selected, most digital recorders will not accept digital input signal unless Non-Copyright is selected.

Analog Analyzer Input Configuration: All the tests described in this digital-to-analog section are set up with the XLR balanced Channel A and B input connectors. The double banana input connectors are hard-wired in parallel with the XLR connectors and may be used instead. All tests are set up with a high (100 kilohms) input impedance, but either 600 or 150 Ohms terminating impedance may be selected individually for Channels A and B by fields at the top of the full-size version of the Analog Analyzer panel.

## Frequency response

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### D-A\_RESP.AT1

Purpose: This test is set up to measure and graph frequency response on an digital input, analog output device such as a D/A converter. Mono or stereo devices may be tested, as described below.

Stimulus: The Digital Generator function of the Digital Domain Audio Tester DSP program (GENANLR.AZ1) is set to an Amplitude of zero dBFS with both channels turned on. The “resting” frequency before and after the sweep is 1 kHz.

**Measurement:** The Analog Analyzer Reading meter in 2-Channel Amplitude function (no filters, RMS detector) makes the measurement.

**Sweep and Display:** The sweep is 50 steps (51 values) logarithmically-spaced from 24 kHz to 10 Hz, sweeping from high to low. The graph Y axis covers the  $\pm 10$  dBr range. The dBr unit is a relative unit and the dBr Reference value will be automatically set to the presently-measured Channel A Level value when the F4 function key is pressed. Since the “resting” frequency of the digital generator is 1 kHz, pressing F4 before pressing F9 or clicking Go will assure that the response curve goes through zero dBr at 1 kHz. If values on the particular device under test plot off the top or bottom of the graph, use the Optimize function (click right mouse button on graph for a list of graph-related commands).

**Stereo:** Both generator channels are on but only channel A will be measured and plotted as the test is set up. To test a stereo device, click the Stereo Sweep checkbox at the bottom of the Sweep panel on Page 2. The Level B meter will then simultaneously plot at Data 3. Note that the Level meter is fully specified only with signal amplitudes of 10 mV or greater. Response measurements at levels significantly below 10 mV should be made using the Amplitude function of the Reading meter, which will require two sweeps with channel switching when the Stereo Sweep checkbox is checked. Since both Data 1 and Data 3 share the left axis which is calibrated in dBr units, the Level B trace will not pass through zero dBr at 1 kHz if there is a gain difference between the two channels at 1 kHz.

## Noise (single point)

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### D-ANOISE.AT1

**Purpose:** This test is set up to measure noise, integrated across the audio spectrum, on a mono or stereo digital-to-analog audio device such as a D/A converter. The result is a single number (per channel), not a graph.

**Stimulus:** The Digital Generator of the Digital Domain Audio Tester DSP program (GENANLR .AZ1) output is turned off. If it is desired to establish a reference level before the noise measurement in order to express the result as signal-to-noise ratio, the digital generator output

should be turned on (control button at the bottom of the Digital Analyzer panel) and the F4 key pressed to set the Analog Analyzer dBr Reference field to the measured level. In this case, the generator must then be turned off again before pressing F9 or clicking Go to obtain the noise measurement.

**Measurement:** : The reading meter of the Analog Analyzer is in Amplitude function using the RMS detector. Other detectors may be selected instead; for example, the Q-Pk detector would be used for a CCIR 468 weighted noise measurement. The high and low pass filters are set to the 22 Hz to 22 kHz range. Thus, the measurement will be limited to noise in the 22 Hz-22 kHz band. Filter selections can be changed to other bandwidths if desired. An appropriate weighting filter (if installed) can be selected in the Filter selection field; for example, the CCIR weighting filter for a CCIR 468 weighted noise measurement. “

**Sweep” and Display:** A single-point measurement is made (no sweep) and tabular results are displayed in the Data Editor since a graph cannot be made of a single point. The reason for selecting Genanlr DGen Amplitude A as the Source 1 parameter, with -999 dBFS as the Start value, is merely to create a Column heading and value in column one of the Data Editor which implies that this is a noise measurement (no signal). Since the generator output is turned off on the generator panel, the actual Source 1 parameter and setting is not important. The measurement is expressed in dBr units, which are decibels relative to the value in the Analog Analyzer dBr Reference field. This value could have been set to the measured level with signal if the generator was turned on and the F4 key pressed as described above, or a known standard reference level can be typed into the dBr Reference field.

**Stereo:** To test a stereo device, click the Stereo Sweep checkbox at the bottom of the Sweep panel on Page 2. Since the reading meter can only measure one channel at a time, the two channels will be automatically be measured in sequence with results displayed in columns two and three of the Data Editor.

## THD + N vs. frequency

### D-ATHDFR.AT1

**Purpose:** This test is set up to measure and graph THD+N (total harmonic distortion plus noise) as a function of stimulus frequency across the audio band on a digital-to-analog audio device such as a D/A converter. Mono or stereo devices may be tested.

**Stimulus:** The Digital Generator function of the Digital Domain Audio Tester DSP program (GENANLR .AZ1) is set to an Amplitude of zero dBFS with both channels turned on. The “resting” frequency before and after the sweep is 1 kHz.

**Measurement:** The reading meter of the Analog Analyzer is in THD+N Relative function with the high and low pass filters set to the 22 Hz to 22 kHz range. Note that the 22 kHz low-pass filter will attenuate second harmonic products from fundamentals above 11 kHz, third harmonics of fundamentals above 7 kHz, etc. The intrinsic band-limiting of the digital device at  $\frac{1}{2}$  the sample rate also eliminates higher-frequency harmonics.

**Sweep and Display:** The sweep is 20 steps (21 values) logarithmically-spaced from 20 kHz to 20 Hz, sweeping from high to low. The graph Y axis covers from zero to -120 dB. For better display of data after measurement of a particular device, use the Optimize function (click right mouse button on graph for a list of graph-related commands).

**Stereo:** Both generator Channels are on but only Channel A will be swept, measured, and plotted as the test is set up. To test a stereo device, click the Stereo Sweep checkbox at the bottom of the Sweep panel on Page 2. Since the reading meter can only measure one channel at a time, the stereo sweep will first sweep and plot Channel A, then switch analyzer channels automatically and sweep and plot Channel B onto the same graph.



## THD + N vs. amplitude

### D-ATHDAM.AT1

**Purpose:** This test is set up to measure and graph THD+N (total harmonic distortion plus noise) as a function of stimulus signal amplitude across a wide dynamic range on a digital-to-analog audio device such as a D/A converter. Mono or stereo devices may be tested.

**Stimulus:** The Digital Generator function of the Digital Domain Audio Tester DSP program (GENANLR .AZ1) is set to a “resting” Amplitude zero dBFS before and after the sweep. Both channels A and B are turned on. The frequency is 1 kHz. The frequency may be changed to any other desired value, but it should be recognized that the band-limited nature of the digital device plus the analyzer 22 kHz low-pass filter will attenuate high-frequency harmonics.

**Measurement:** The reading meter of the Analog Analyzer is in THD+N Amplitude function. THD+N Amplitude expresses the integrated distortion and noise measurement in absolute units rather than relative to the present value of the fundamental signal. The high and low pass filters are set to the 22 Hz to 22 kHz range. Note that the 22 kHz low-pass filter will attenuate second harmonic products from fundamentals above 11 kHz, third harmonics of fundamentals above 7 kHz, etc..

**Sweep and Display:** The sweep is 40 2-dB steps (41 values) from 0 dBFS to -80 dBFS, sweeping from low to high amplitude. The graph Y axis is calibrated from zero to -120 dBr. The F4 key should be pressed while the generator output is at 0 dBFS, before pressing F9 or clicking Go for the sweep. The F4 key causes the present Level A reading (full-scale output level of the D/A converter under test) to be written into the dBr Reference field. Distortion and noise measurements are thus expressed in dB below full scale. An ideal D/A converter would show a flat horizontal line across the graph, with its quantization distortion and noise level independent of signal level. For better display of data after measurement of a particular device, use the Optimize function (click right mouse button on graph for a list of graph-related commands).

Stereo: Both generator Channels are on but only Channel A will be swept, measured, and plotted as the test is set up. To test a stereo device, click the Stereo Sweep checkbox at the bottom of the Sweep panel on Page 2. Since the reading meter can only measure one channel at a time, the stereo sweep will first sweep and plot Channel A, then switch Analyzer channels automatically and sweep and plot Channel B onto the same graph.

## **Interchannel phase vs. frequency**

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### **D-APHASE . AT1**

Purpose: This test is set up to measure and graph interchannel phase difference of a digital-to-analog stereo or other two-channel audio device such as a D/A converter, as a function of frequency.

Stimulus: The Digital Generator function of the Digital Domain Audio Tester DSP program (GENANLR . AZ1) is set to an Amplitude of zero dBFS with both channels turned on. The “resting” frequency before and after the sweep is 1 kHz.

Measurement: The Phase meter measures phase difference between the signals at the two analyzer input connectors. It is fully-specified for accuracy only with signal amplitudes of 10 mV and above on both channels.

Sweep and Display: The sweep is 50 steps (51 values) logarithmically-spaced from 20 Hz to 20 kHz, sweeping from low to high. The graph Y axis covers the  $\pm 10$  degree range. If values on the particular device under test plot off the top or bottom of the graph, use the Optimize function (click right mouse button on graph for a list of graph-related commands).

Stereo: Interchannel phase is intrinsically a measurement made only on stereo or two-channel devices.

## Linearity

### D-A\_LIN.AT1

**Purpose:** This test is set up to measure and graph deviation from perfect input-output linearity of a digital-to-analog mono or stereo audio device such as a D/A converter.

**Stimulus:** The Digital Generator function of the Digital Domain Audio Tester DSP program (GENANLR .AZ1) is set to a “resting” zero dBFS before and after the sweep. Both channels A and B are turned on. The waveform is a sinewave with a frequency of 200 Hz.

**Measurement:** The reading meter of the Analog Analyzer is in Bandpass function with the Bandpass filter frequency fixed at 200 Hz. Bandpass function is used to produce a frequency-selective amplitude measurement which rejects noise across most of the spectrum in order to measure the generator signal amplitudes accurately even at amplitudes below the wide-band noise level.

**Sweep and Display:** The sweep defined by Source 1 on the Sweep panel consists of 60 steps of two-dB size from zero dBFS to -120 dBFS. The intent is to test a typical D/A converter from full scale to near or below its noise level, and Start and Stop values may be changed as required by a particular device. The reading meter measurements are plotted at Data 1 (and Data 3 if stereo) in dBr units. A perfectly linear device will result in a diagonal straight line trace across the entire range with the horizontal and vertical coordinates of each point exactly equal. To make it easier to see small deviations from perfect linearity, the Compute Linearity function is set up to automatically execute following the last point of the sweep. Compute Linearity operates by fitting a best straight line to the data over a certain range (-20 dBFS to -40 dBFS in this case), then subtracts every measurement point from the perfect straight line. The result will be an approximately straight horizontal line near zero dBr. Use the right mouse button and select the Optimize Left Only command following the Compute Linearity operation to automatically select a new graph top and bottom to best displaying the deviation from linearity graph. Most good D/A converters can be expected to be very linear from a few dB below full scale down to approximately the noise level. At some point well below

the wideband noise level, the curve can be expected to swing upwards since the analyzer measurement is still influenced by noise falling within the bandpass filter range.

**Stereo:** Both generator Channels are on but only Channel A will be swept, measured, and plotted as the test is set up. To test a stereo device, click the Stereo Sweep checkbox at the bottom of the Sweep panel on Page 2. Since the reading meter can only measure one channel at a time, the stereo sweep will first sweep and plot Channel A, then switch analyzer channels automatically and sweep and plot Channel B onto the same graph.

## **Crosstalk vs. frequency**

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### **D-AXTALK . AT1**

**Purpose:** This test measures and graphs crosstalk attenuation or stereo separation from one channel of a digital-to-analog audio device such as a D/A converter into another channel, as a function of frequency.

**Stimulus:** The Digital Generator function of the Digital Domain Audio Tester DSP program (GENANLR . AZ1) is set to an Amplitude of zero dBFS. The “resting” frequency before and after the sweep is 1 kHz. Only the Channel A output is turned on, so that crosstalk/separation into Channel B may be measured.

**Measurement:** The reading meter of the Analog Analyzer is in Crosstalk function, measuring the signal connected to Analyzer Channel B and expressing it in dB relative to the Level meter reading on Channel A. The analyzer thus measures the relative signal feed-through between channels. Crosstalk function uses the reading meter bandpass filter in order to reject wide-band noise and accurately measure low levels of crosstalk signal. The bandpass filter frequency is controlled by the Sweep Track selection in the BP/BR Filter Frequency field. This causes the filter to track the frequency of the device selected at Source 1 on the Sweep panel, which is the digital generator section of GENANLR . AZ1.

**Sweep and Display:** The sweep is 50 steps (51 values) logarithmically-spaced from 20 kHz to 20 Hz, sweeping from high to

low. The graph Y axis is set for the -40 to -130 dB range. For better display of data after measurement of a particular device, use the Optimize function (click right mouse button on graph for a list of graph-related commands).

Stereo: This test is set up as a Stereo Sweep by the checkbox at the bottom of the Sweep panel, since crosstalk testing implies a two-channel device. Since the reading meter can only measure one channel at a time, the stereo sweep will first sweep while driving Channel A and plotting Channel B, then will switch generator and analyzer channels automatically and drive B while plotting A onto the same graph.

## FFT spectrum analysis

### D-A\_FFT.AT1

Purpose: This test is set up to sample, acquire, and perform an FFT (Fast Fourier Transform) spectrum analysis on one or two channels of an analog output audio device. Stimulus signal to a D/A converter may be supplied by System One's digital generator, but the test may also be used to acquire any audio signal including playback of recordings, broadcast signals, real-time speech or music, etc.

Stimulus: There is no intrinsic relation between stimulus and measurement in this test. Signal may be acquired and an FFT performed whether signal is coming from System One's digital generator or from any other source. If the digital generator signal is to be used, the function is provided by the generator section of the Spectrum Analyzer/Generator DSP program (FFTGEN.AZ1) which is set for a 1 kHz sinewave at an Amplitude of 1.000 FFS (fraction of full scale) with both Channel A and B outputs on.

Measurement: The Spectrum Analyzer/Generator DSP program (FFTGEN.AZ1) is selected on the digital analyzer panel. Its analog-to-digital converters are fed from the two input channels of the Analog Analyzer, following the input ranging circuitry and balanced-to-unbalanced conversion. A Maximum (16k sample) acquisition and FFT is selected. The sample rate selected for the digital output is also used by the A/D converters. If the rate is 48 kHz a measurement bandwidth in excess of 20 kHz results."

Sweep” and Display: FFT channels 1 and 2, fed from analog input channels A and B, are set up as Data 1 and Data 2 selections. The zero to -130 dBr graph top and bottom selected will cover the range of most signals. The F4 function key should be pressed before F9 or Go is clicked, which will write the presently-measured Level reading into the dBr Reference field. The Optimize command of the graph right mouse button dialog may be selected to provide better graphical resolution. Source 1 selects frequency domain (spectrum analysis) and plots the zero to 24 kHz range on a linear axis. Smaller portions of this range may be selected by zooming with the mouse or changing the Source 1 Start and Stop values. A logarithmic horizontal axis may be selected if both Start and Stop are positive values.

Stereo: As set up, the test measures both channels simultaneously. If signal is present on only one channel, the other trace may be turned off either by selecting “None” instead of the FFT channel at the Data 1 or Data 2 Browser, or by disabling the trace in the first column of the graph legend.

## Waveform display

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### D-A\_WAVE.AT1

Purpose: This test is set up to sample, acquire, and display the waveform of one or two channels of an analog output audio device. Stimulus signal may be supplied to a D/A device by System One’s digital generator, but the test may also be used to acquire any audio signal including playback of recordings, broadcast signals, real-time speech or music, etc.

Stimulus: There is no intrinsic relation between stimulus and measurement in this test. Signal may be acquired and displayed whether signal is coming from System One’s digital generator or from any other source. If the digital generator signal is to be used, it is part of the Spectrum Analyzer/Generator DSP program (FFTGEN.AZ1) and is set for a 1 kHz sinewave at an Amplitude of 1.000 FFS (fraction of full scale) with both Channel A and B outputs on.

Measurement: The FFT Spectrum Analyzer DSP program (FFTGEN.AZ1) is selected on the digital analyzer panel. Its analog-to-digital converters are fed from the two input channels of the

Analog Analyzer, following the input ranging circuitry and balanced-to-unbalanced conversion. A Maximum (16k sample) acquisition and FFT is selected. The A/D converters will operate at the same sample rate as the digital generator. If a sample rate of 48 kHz is used, it produces a measurement bandwidth in excess of 20 kHz.“

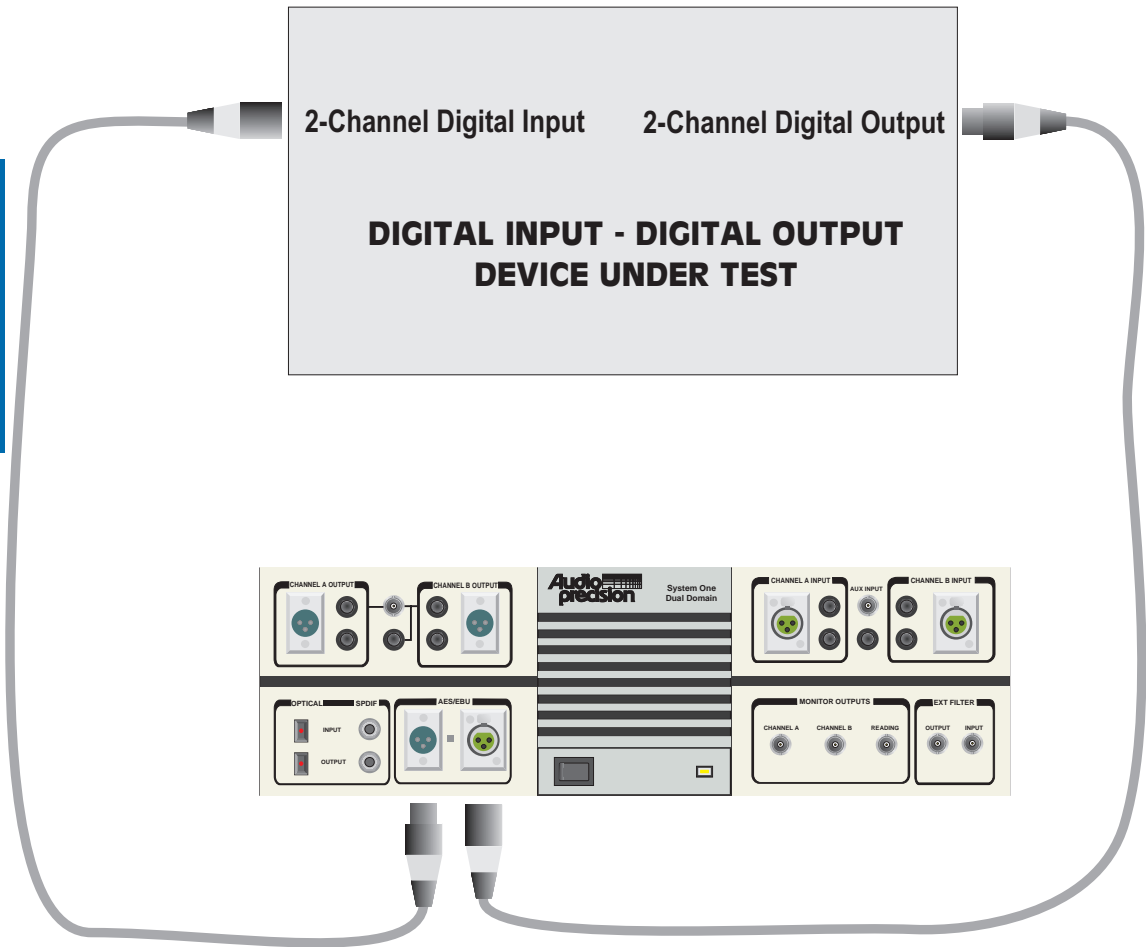
Sweep” and Display: FFT channels 1 and 2, fed from analog input channels A and B, are set up as Data 1 and Data 2 selections. Presuming a measured signal amplitude around ten volts RMS, the graph top and bottom are set to +60 and -20 Volts at Data 1 and +20 and -60 Volts at Data 2 to separate the two traces in dual-trace oscilloscope fashion. If it is desired to put the two channel traces on top of one another for comparison, Data 1 and Data 2 Top and Bottom values should be the same. The Optimize Together command of the graph right mouse button dialog will accomplish this automatically and will optimize the display for signals of significantly higher or lower amplitude than ten volts RMS. Source 1 selects time domain (oscilloscope mode) and plots the zero to five millisecond range on a linear axis. Smaller portions of this range may be selected by zooming with the mouse or changing the Source 1 Start and Stop values. The full record length is approximately 341 milliseconds (16,384 samples/48,000 samples per second) and the Source 1 Start and Stop values may be set anywhere in the 0-341 ms range to view the desired portion of the acquired record. The time span between Start and Stop values may be set as desired and is typically equal to several periods of the signal when viewing a sinewave.

Stereo: As set up, the test measures both channels simultaneously. If signal is present on only one channel, the other trace may be turned off either by selecting “None” instead of the FFT channel at the Data 1 or Data 2 Browser, or by disabling the trace in the first column of the graph legend.

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## Digital-to-Digital Tests

An SYS-322 (Dual Domain) unit is required for all tests in this section. A standard installation places System One digital-to-digital tests in the **C:\APWIN\S1\D-D** subdirectory.



Digital Generator: All the tests described in this section use System One's digital generator. This generator drives the XLR, RCA phono, and Optical digital output connectors simultaneously. Most tests in this section are set up with the Resolution field of the DIO Panel set to 24 bits. The Resolution field determines at which bit level (word width) the internally-generated 24-bit signal is rounded, with dither (if enabled)



added at the appropriate amplitude for a signal of that resolution. It is generally desirable to change the Resolution value to the known input resolution of the device under test; this is particularly critical when tests are made of the low-amplitude performance of the device such as linearity and noise. All tests are set up with Triangular dither enabled, but Rectangular or Shaped dither or no dither may be selected via the DIO panel.

**Sample Rate, Digital Output:** All the tests described in this section are set up with the presumption that the device under test is to be tested at a 48 kHz sample rate controlled from System One's internal crystal. To test at 44.1 kHz or 32 kHz, select the desired rate at the Rate field on the DIO panel. If a rate lower than 48 kHz is chosen and no changes have been made to the tests in this section as provided, many of them will fail with an error message of the form "Conflict with maximum DSP setting value while setting XXX (Source 1 parameter) to nnn kHz." To adapt a test to any specific sample rate, change the upper frequency value at Source 1 on the Sweep Panel from its stored value (typically either 20 kHz or 24 kHz) to a value not greater than  $\frac{1}{2}$  the actual sample rate.

**Status Bits:** All tests in this section are set up to transmit status bits in the Professional format with sample rate and emphasis not indicated on both channels (hex code of 01 00 00 00 for bytes 0-3). Many digital input devices, particularly digital recorders, will not accept a digital input unless the status bits on the digital signal correspond to what the digital product designer thought was appropriate. If the device under test requires other status bits in order to function as desired, the Status Bits panel should be used to select the desired output. This panel may be displayed by clicking on the Status Bits button on the DIO panel, by clicking on the STAT BIT icon on the toolbar, or via the Panels Status Bits menu command. The Professional/Consumer selection is almost always important, and it may also be important that the transmitted Sample Frequency information in the status bits matches the actual Sample Rate being generated by System One. The Sample Frequency information in the status bits is not automatically set when System One sample rate is changed. When Consumer format is selected, most digital recorders will not accept digital input signal unless Non-Copyright is selected.

Digital Analyzer Input Configuration: All the tests described in this section are set up expecting an AES/EBU or SPDIF format digital input signal connected to either the XLR, phono, or optical digital input connector. Presence of a signal at one or another of these three connectors is automatically sensed.

Audio Frequency Range vs. Sample Rate, Digital Input: The digital analyzer always locks to the sample rate of the digital input signal and is independent from the digital generator output sample rate. However, the measured frequency of the incoming digital audio signals will be correct only when the received sample rate is the same as the Rate selection on the DIO panel (which controls the output sample rate).

## Frequency response

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### D-D\_RESP.AT1

Purpose: This test is set up to measure and graph frequency response on a digital input, digital output device. Mono or stereo devices may be tested, as described below.

Stimulus: The Digital Generator function of the Digital Domain Audio Tester DSP program (GENANLR .AZ1) is set to an Amplitude of -1.0 dBFS. A level below full scale was selected so that both upward and downward response variations, if they exist, can be seen without clipping. Both channels are turned on. The “resting” frequency before and after the sweep is 1 kHz.

Measurement: The digital domain level measurement of the channel A digital signal is made by the main (reading) meter of the Digital Domain Audio Tester DSP program (GENANLR .AZ1) in 2 Channel function. The channel B signal is simultaneously measured by the Level B meter of the same program.

Sweep and Display: The sweep is 50 steps (51 values) logarithmically-spaced from 20 kHz to 20 Hz, sweeping from high to low. The graph Y axis covers the zero to -2.0 dBFS range, presuming unity gain through the device. If values on the particular device under test plot off the graph, use the Optimize function (click right mouse button on graph for a list of graph-related commands).

**Stereo:** The test is stored as a stereo test since most digital domain devices are stereo. Both Channel A and B will be plotted in a single sweep. For single channel devices, uncheck the Stereo Sweep checkbox at the bottom of the Sweep panel and only Channel A will be tested.

## Noise (single point)

### D-DNOISE.AT1

**Purpose:** This test is set up to measure noise, integrated across the audio spectrum, on a mono or stereo digital-to-digital audio device. The result is a single number (per channel), not a graph.

**Stimulus:** The Digital Generator function of the Digital Domain Audio Tester DSP program (GENANLR .AZ1) is turned on to achieve proper dithering, but the amplitude is set to -999 dBFS. Triangular dither is selected on the DIO panel and the Resolution (word width) is set to 16 bits. The Resolution value should be changed to correspond to the digital input resolution of the device under test in order to establish dither amplitude properly.

**Measurement:** The main (reading) meter of the Digital Domain Audio Tester DSP program (GENANLR .AZ1) is in Noise Unweighted function using the RMS detector. The CCIR quasi-peak (Q-Pk) detector and Noise CCIR-Weighted function may be used instead for a CCIR 468 weighted noise measurement, by selecting “4/sec QPK” in the Reading Rate field of the digital analyzer panel. A Noise A-weighted function is also selectable.

**Sweep and Display:** A single-point measurement is made (no sweep) and tabular results are displayed in the Data Editor since a graph cannot be made of a single point.

**Stereo:** The test is set up as a stereo sweep since most digital devices are stereo. Two sequential single-point measurements will be made, with the Channel A first appearing in column 2 of the Data Editor, followed by Channel B in column 3. If a monaural digital channel is to be tested, uncheck the Stereo Sweep checkbox at the bottom of the Sweep panel and only Channel A will be tested.

## THD + N vs. frequency

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### D-DTHDFR.AT1

**Purpose:** This test is set up to measure and graph THD+N (total harmonic distortion plus noise) as a function of stimulus frequency across the audio band on a digital-to-digital audio device. Mono or stereo devices may be tested.

**Stimulus:** The Digital Generator function of the Digital Domain Audio Tester DSP program (GENANLR . AZ1) is set to an Amplitude of zero dBFS. Both channels are turned on. The “resting” frequency before and after the sweep is 1 kHz.

**Measurement:** The main (reading) meter of the Digital Domain Audio Tester DSP program (GENANLR . AZ1) is in THD+N Relative function.

**Sweep and Display:** The sweep is 20 steps (21 values) logarithmically-spaced from 20 kHz to 50 Hz, sweeping from high to low. GENANLR . AZ1 THD+N function is not specified below 50 Hz. The graph Y axis is calibrated in dB below signal level, from zero to -120 dB. For better display of data after measurement of a particular device, use the Optimize function (click right mouse button on graph for a list of graph-related commands).

**Stereo:** Since most digital devices are stereo, the test is set up in Stereo Sweep mode. Two sequential sweeps will be made, with Channel A first plotting as Data 1 followed by Channel B plotting as Data 3. To test a single channel device, uncheck the Stereo Sweep checkbox and only Channel A will be tested.

## THD + N vs. amplitude

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### D-DTHDAM.AT1

**Purpose:** This test is set up to measure and graph THD+N (total harmonic distortion plus noise) as a function of stimulus signal amplitude across a wide dynamic range on a digital-to-digital audio device. Mono or stereo devices may be tested.

**Stimulus:** The Digital Generator function of the Digital Domain Audio Tester DSP program (GENANLR . AZ1) is set to a “resting” Amplitude of 1.000 FFS (fraction of full scale) before and after the sweep. Both channels A and B are turned on. The frequency is 1 kHz. The frequency may be changed to any other desired value above 50 Hz, but it should be recognized that the band-limited nature of the digital device will attenuate harmonics above one-half the sample rate.

**Measurement:** The main (reading) meter of the Digital Domain Audio Tester DSP program (GENANLR . AZ1) is in THD+N Absolute function. Absolute THD+N states distortion and noise in absolute terms (% of full scale, dBFS, etc.) rather than relative to the present signal amplitude.

**Sweep and Display:** The sweep is 40 2-dB steps (41 values) from zero to -80 dBFS, sweeping from high to low amplitude. The graph Y axis is calibrated in dBFS (decibels below digital full scale) from -60 to -130 dBFS. An ideal digital device would be limited only by the intrinsic quantization distortion and noise due to its resolution (word width), independent of signal level. Therefore, an absolute distortion vs. amplitude curve can be expected to be approximately a horizontal straight line across the graph. For better display of data after measurement of a particular device, use the Optimize function (click right mouse button on graph for a list of graph-related commands).

**Stereo:** Since most digital devices are stereo, this test is set up with the Stereo Sweep checkbox checked. Two sequential sweeps will be made, with Channel A first plotting as Data 1 followed by Channel B plotting as Data 3. If a mono device is to be tested, uncheck the Stereo Sweep checkbox and only Channel A will be measured.

## **Interchannel phase vs. frequency**

### **D-DPHASE . AT1**

**Purpose:** This test is set up to measure the interchannel phase versus frequency characteristic of a digital-to-digital device. The test uses multitone technology.

**Stimulus:** The Triggered Multitone Tester DSP program (FASTTRIG . AZ1) provides both stimulus and measurement. At the

bottom of the Digital Analyzer panel, a test signal waveform file name ISO31.AGM is selected. This waveform file defines a multitone test signal consisting of 31 equal-amplitude sinewaves, one at approximately each ISO standard 1/3 octave frequency from 20 Hz to 20 kHz. Amplitude calibration of the 31-tone signal is in terms of sinewave equivalent peak with 1.0 dB headroom; that is, the peak-to-peak amplitude of the complex signal is 1.0 dB less than the peak-to-peak amplitude of a sinewave of 1.000 FFS.

Measurement: The Triggered Multitone Tester DSP program (FASTTRIG.AZ1) is selected on the digital analyzer panel. Digital input is selected, with digital channel A selected as Channel 1 and channel B as Channel 2. A Maximum (16k sample) acquisition and FFT is automatically selected since the generator file is 8k. Spectrum measurement function, Window None (for synchronous processing), and Free Run Triggering are selected on the digital analyzer panel. “

Sweep” and Display: FASTTRIG FFT Frequency is selected at Source 1 with Start and Stop frequencies of 10 Hz and 20 kHz and a log horizontal display. If the sample rate is lower than 44.1 kHz, the Source 1 Stop frequency must be changed to a value no higher than  $\frac{1}{2}$  the sample rate. Log display is used since the 31 test tones are logarithmically spaced across the audio spectrum. The sweep table FASTTEST.ADS is attached at Source 1, causing phase measurements to be made only at the exact fundamental frequencies of the multitone signal. FASTTRIG Channel 1-2 Phase is selected at Data 1. If the measured phase difference exceeds the  $\pm 5$  degree range at Data 1 Graph Top and Bottom, the Optimize command of the graph right mouse button menu may be used to fit the graph top and bottom to the actual measured data.

Stereo: Interchannel phase is intrinsically a stereo test.

## Linearity

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### D-D\_LIN.AT1

Purpose: This test is set up to measure and graph deviation from perfect input-output linearity of an digital-to-digital mono or stereo audio.

**Stimulus:** The Digital Generator function of the Digital Domain Audio Tester DSP program (GENANLR . AZ1) is set to a “resting” Amplitude of zero dBFS before and after the sweep. Both channels A and B are turned on. The waveform is a sinewave with a frequency of one kHz. The Resolution on the DIO panel is set to 16 bits; it must be changed if necessary to match the input resolution of the device under test.

**Measurement:** The Digital Domain Audio Tester DSP program (GENANLR . AZ1) is selected on the Digital Analyzer panel. Digital Channel A is connected to the reading meter of the Digital Domain Analyzer. The analyzer is in Bandpass function with the Bandpass filter frequency slaved to the Digital Generator frequency via the DGen selection of the BP/BR Tuning field. Bandpass function is used to produce a frequency-selective amplitude measurement, which rejects noise across most of the spectrum in order to measure the generator signal amplitude accurately even at amplitudes below the wide-band noise level.

**Sweep and Display:** The sweep defined by Source 1 on the Sweep panel consists of 60 steps (61 measurement points) of two-dB size from 0 dBFS to -120 dBFS. The intent is to test a device from digital full scale to below the wideband noise level. Start and Stop values may be changed as required by a particular device. The Channel A measurements are plotted at Data 1 and Data 2 in dBFS. Channel B measurements are plotted at Data 3 and Data 4. The purpose of plotting each channel twice is so that the Compute Linearity function may be used on one trace while still preserving the original data. A perfectly linear device will result in a straight diagonal trace across the entire range with the horizontal and vertical coordinates of each point exactly equal. To make it easier to see small deviations from perfect linearity, the Compute Linearity function is set up to automatically execute on the Data 2 (second instance of Channel A data) and Data 4 (second instance of Channel B data) following the last point of the sweep. Compute Linearity operates by fitting a best straight line to the data over a certain range (-20 dBFS to -40 dBFS in this case), then subtracts every measurement point from the perfect straight line. The result will be an approximately straight horizontal line near zero dBFS for the Data 1 and 3 traces. Use the right mouse button and select the



Optimize Individually command following the Compute Linearity operation to automatically select a new graph top and bottom for the Data 1/3 (left hand) vertical axis to best display the deviation from linearity graph. The original diagonal traces will still be plotted, calibrated against the right-hand vertical axis. Most devices can be expected to be very linear from a few dB below full scale down to approximately the noise level. At some point well below the wideband noise level, the deviation from linearity curve can be expected to swing upwards since the analyzer measurement is still influenced by noise falling within the bandpass filter range.

Stereo: Most digital devices are stereo, so the test is set up with the Stereo Sweep checkbox checked. Since the reading meter can only measure one channel at a time, the stereo sweep will first sweep and plot Channel A as Data 1 and 2, then switch analyzer channels automatically and sweep and plot Channel B as Data 3 and 4 onto the same graph. The Compute Linearity function is enabled for both Data 1 and Data 3, so the deviation from perfect linearity result for both traces will be produced (left-hand vertical axis calibration) and both will be individually optimized by the graph right button Optimize Individually command.

## FFT spectrum analysis

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### D-D\_FFT.AT1

Purpose: This test is set up to sample, acquire, and perform an FFT (Fast Fourier Transform) spectrum analysis on one or two channels of a digital output audio device. Stimulus signal may be supplied by System One's digital generator, but the test may also be used to acquire any digital domain audio signal including playback of recordings, broadcast signals, real-time speech or music, etc.

Stimulus: There is no intrinsic relation between stimulus and measurement in this test. Signal may be acquired and an FFT performed whether signal is coming from System One's digital generator or from any other source. If the digital generator signal is to be used, it is part of the Spectrum Analyzer/Generator DSP program (FFTGEN.AZ1) and is set for a 1 kHz sinewave at an Amplitude of zero dBFS with both Channel A and B outputs on.



Measurement: The Spectrum Analyzer/Generator DSP program (FFTGEN .AZ1) is selected on the digital analyzer panel. Digital input is selected with Digital Channel A connected to FFT Channel 1 and Digital Channel B connected to Channel 2. A Maximum (16k sample) acquisition and FFT is selected. The digital analyzer will lock to digital input sample rates over a wide range. “

Sweep” and Display: FFT channels 1 and 2, fed from digital input channels A and B, are set up as Data 1 and Data 2 selections. The zero to -140 dBFS (dB relative to digital full scale) graph top and bottom selected will cover the range of most signals. The Optimize command of the graph right mouse button dialog may be selected to provide better graphical resolution following an acquisition. The Source 1 FFT Freq selection chooses frequency domain (spectrum analysis) mode. The zero and 24 kHz Source 1 Start and Stop values are based on the assumption of a 48 kHz sample rate. If the sample rate of the device is 44.1 kHz or 32 kHz, the Source 1 Stop value must be changed to a number no greater than  $\frac{1}{2}$  the sample rate. A linear axis is chosen. Smaller portions of this range may be selected by zooming with the mouse or changing the Source 1 Start and Stop values following an acquisition. A logarithmic horizontal axis may be selected if both Start and Stop are positive values.

Stereo: As set up, the test measures both channels simultaneously. If signal is present on only one channel, the other trace may be turned off either by selecting “None” instead of the FFT channel at the Data 1 or Data 2 Browser, or by disabling the trace in the first column of the graph legend.

## Waveform display

---

### D-D\_WAVE .AT1

Purpose: This test is set up to sample, acquire, and display the waveform of one or two channels of a digital output audio device. Stimulus may be supplied by System One’s digital generator, but the test may also be used to acquire any digital audio signal including playback of recordings, broadcast signals, real-time speech or music, etc.

Stimulus: There is no intrinsic relation between stimulus and measurement in this test. Signal may be acquired and displayed whether signal originated from System One's digital generator or from any other source. If the digital generator signal is to be used, it is part of the Spectrum Analyzer/Generator DSP program (FFTGEN.AZ1) and is set for a 1 kHz sinewave at an Amplitude of zero dBFS with both Channel A and B outputs on.

Measurement: The Spectrum Analyzer/Generator DSP program (FFTGEN.AZ1) is selected on the digital analyzer panel. Digital input is selected, with digital analyzer Channel 1 fed from Channel A of the digital signal and Channel 2 fed from Channel B. A 16k sample acquisition is selected. “

Sweep” and Display: FFT channels 1 and 2, fed from digital input channels A and B, are set up as Data 1 and Data 2 selections. Presuming a measured signal amplitude of approximately digital full scale, the graph top and bottom are set to display signals  $\pm 1.5$  FFS (fraction of digital full scale). Note that the definition of digital full scale is the rms value of a sinewave whose peaks are just touching the most positive and most negative allowable digital codes. A sinewave has a crest factor (peak-to-rms value) of 1.414. Thus, the peaks of a zero dBFS (rms) sinewave signal level will touch  $\pm 1.414$  FFS and  $\pm 1.5$  FFS was used to preserve round numbers. The actual Graph Top and Bottom values of +4.5 FFS and -1.5 FFS at Data 1 and +1.5 and -4.5 FFS at Data 2 separate the two traces in dual-trace oscilloscope fashion. If it is desired to put the two channel traces on top of one another for comparison, Data 1 and Data 2 Top and Bottom values should be the same. The Optimize Together command of the graph right mouse button dialog will accomplish this automatically and will optimize the display for signals of significantly lower amplitude than one FFS. Source 1 selects FFT Time to provide time domain (oscilloscope mode) display rather than spectrum analysis. The zero to five millisecond range is selected at Source 1 Start and Stop to display five cycles of the signal is 1.00 kHz. Smaller portions of this range may be selected by zooming with the mouse or changing the Source 1 Start and Stop values. The full record length is approximately 341 milliseconds (16,384 samples/48,000 samples per second) and the Source 1 Start and Stop values may be set anywhere in the 0-341 ms

range to view the desired portion of the acquired record. The time span between Start and Stop values may be set as desired and is typically equal to several periods of the signal when viewing a sinewave.

Stereo: As set up, the test measures both channels simultaneously. If signal is present on only one channel, the other trace may be turned off either by selecting “None” instead of the FFT channel at the Data 1 or Data 2 Browser, or by disabling the trace in the first column of the graph legend.

## Summary

### Analog-to-Analog

**C : \APWIN\S1\A-A**

Frequency response: A-A\_RESP .AT1

Noise (single point): A-ANOISE .AT1

THD+N vs. frequency: A-ATHDFR .AT1

THD+N vs. amplitude: A-ATHDAM .AT1

SMPTE IMD vs. amplitude: A-ASMPTE .AT1

CCIF IMD vs. amplitude: A-A\_CCIF .AT1

DIM IMD vs. amplitude: A-A\_DIM .AT1

Interchannel phase vs. frequency: A-APHASE .AT1

Input/Output phase: A-APHAIO .AT1

Linearity: A-A\_LIN .AT1

Crosstalk vs. frequency: A-AXTALK .AT1

FFT spectrum analysis: A-A\_FFT .AT1

Waveform display: A-A\_WAVE .AT1

External frequency response: A-AEXTRS . AT1

External THD+N vs. frequency: A-AEXTTH . AT1

Wow & flutter vs. time: A-A\_W&F . AT1

Locating -3 dB Point: A-A\_3dB . AT1

Power Bandwidth: A-APWRBD . AT1

Equalized Phonograph Preamplifier: A-A\_RIAA . AT1

## **Analog-to-Digital**

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**C:\APWIN\S1\A-D**

Frequency response: A-D\_RESP . AT1

Noise (single point): A-DNOISE . AT1

THD+N vs. frequency: A-DTHDFR . AT1

THD+N vs. amplitude: A-DTHDAM . AT1

SMPTE IMD (FFT): A-DSMPTE . AT1

CCIF IMD (FFT): A-D\_CCIF . AT1

DIM IMD (FFT): A-D\_DIM . AT1

Interchannel phase vs. frequency (multitone): A-DPHASE . AT1

Linearity: A-D\_LIN . AT1

Crosstalk vs. frequency: A-DXTALK . AT1

FFT spectrum analysis: A-D\_FFT . AT1

Waveform display: A-D\_WAVE . AT1

## Digital-to-Analog

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**C: \APWIN\S1\D-A**

Frequency response: D-A\_RESP .AT1

Noise (single point): D-ANOISE .AT1

THD+N vs. frequency: D-ATHDFR .AT1

THD+N vs. amplitude: D-ATHDAM .AT1

Interchannel phase vs. frequency: D-APHASE .AT1

Linearity: D-A\_LIN .AT1

Crosstalk vs. frequency: D-AXTALK .AT1

FFT spectrum analysis: D-A\_FFT .AT1

Waveform display: D-A\_WAVE .AT1

## Digital-to-Digital

---

**C: \APWIN\S1\D-D**

Frequency response: D-D\_RESP .AT1

Noise (single point): D-DNOISE .AT1

THD+N vs. frequency: D-DTHDFR .AT1

THD+N vs. amplitude: D-DTHDAM .AT1

Interchannel phase vs. frequency (multitone): D-DPHASE .AT1

Linearity: D-D\_LIN .AT1

FFT spectrum analysis: D-D\_FFT .AT1

Waveform display: D-D\_WAVE .AT1

**22** files

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