2700 Series

Getting Started with Your 2700 Series Instrument

2722 • 2720 • 2712 • 2702

Audio precision ®

Getting Started with Your 2700 Series Instrument

An Introductory Guide to • SYS-2722 •

and • SYS-2720 • SYS-2712 • SYS-2702 •

APIB and GPIB Configurations

CE

This mark signifies that the product conforms to all applicable requirements of the European Community. A Declaration of Conformance is included with the user information that describes the specifications used to demonstrate conformity.

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Safety Information

Do NOT service or repair this product unless properly qualified. Servicing should be performed only by a qualified technician or an authorized Audio Precision distributor.

Do NOT defeat the safety ground connection. This product is designed to operate only from a 50/60 Hz AC power source (250 V rms maximum) with an approved three-conductor power cord and safety grounding. Loss of the protective grounding connection can result in electrical shock hazard from the accessible conductive surfaces of this product.

For continued fire hazard protection, fuses should be replaced ONLY with the exact value and type indicated on the rear panel of the instrument and discussed on page 12 of this manual. The AC voltage selector also must be set to the same voltage as the nominal power source voltage (100, 120, 230, or 240 V rms) with the appropriate fuses. Different fuses are required depending on the line voltage.

The International Electrotechnical Commission (IEC 1010-1) requires that measuring circuit terminals used for voltage or current measurement be marked to indicate their Installation Category. The Installation Category is defined by IEC 664 and is based on the amplitude of transient or impulse voltage that can be expected from the AC power distribution network. This product is classified as INSTALLATION CATEGORY II, abbreviated "CAT II" on the instrument front panel.

Do NOT substitute parts or make any modifications without the written approval of Audio Precision. Doing so may create safety hazards.

This product is for indoor use—pollution degree 2.

Safety Symbols

The following symbols may be marked on the panels or covers of equipment or modules, and are used in this manual:

WARNING!—This symbol alerts you to a potentially hazardous condition, such as the presence of dangerous voltage that could pose a risk of electrical shock. Refer to the accompanying Warning Label or Tag, and exercise extreme caution.

ATTENTION!—This symbol alerts you to important operating considerations or a potential operating condition that could damage equipment. If you see this marked on equipment, refer to the Operator's Manual or User's Manual for precautionary instructions.

FUNCTIONAL EARTH TERMINAL—A terminal marked with this symbol is electrically connected to a reference point of a measuring circuit or output and is intended to be earthed for any functional purpose other than safety.

PROTECTIVE EARTH TERMINAL—A terminal marked with this symbol is bonded to conductive parts of the instrument and is intended to be connected to an external protective earthing system.

Disclaimer

Audio Precision cautions against using their products in a manner not specified by the manufacturer. To do otherwise may void any warranties, damage equipment, or pose a safety risk to personnel.

Chapter 1

General Information

Figure 1. The Audio Precision 2700 series audio test and measurement instrument; SYS-2722 shown.

The 2700 Series: an Overview

Audio Precision 2700 series instruments are computer-controlled audio test sets offering broad, high-performance capabilities for testing analog, digital, and mixed-domain devices. The 2700 series instruments are based on models in the Audio Precision System Two Cascade and Cascade *Plus* lines. A 2700 series instrument consists of two key components:

■ The 2700 series instrument hardware, which provides the physical platform and measurement electronics necessary for proper signal interface and precise signal generation and analysis. The hardware is mounted in a sturdy aluminum and steel chassis that can be installed in a standard 19" wide equipment rack using optional mounting hardware.

■ The 2700 series control software, AP2700. AP2700 runs on a personal computer (PC) and provides the control, computation, display, report and automation functions for a 2700 series instrument. (The PC must be purchased separately). The AP2700 control software will run under Microsoft Windows 2000 and Windows XP operating systems.

With the exception of the mains power switch and the monitor speaker volume control, there are no knobs, dials, controls, readouts, meters or switches on 2700 series instrument hardware. All of the measurement settings and adjustments are made in software on the PC.

The control software on the PC communicates with the 2700 hardware via a proprietary bus interconnection called the Audio Precision Interface Bus (APIB), which requires a dedicated APIB cable and PC-mounted APIB interface card or adapter. APIB interfaces can be ordered as PCI or PCMCIA-compatible devices.

When ordered in the optional GPIB configuration (SYS-27xxG), a 2700 series instrument can also be controlled by the industry-standard IEEE 488.2 General Purpose Interface Bus (GPIB). See Chapter 7 for more information about the 2700 series GPIB configuration.

2700 Series Capabilities

The fully-equipped 2700 series instrument, model SYS-2722, is a *dual domain* instrument; it can perform analysis in both the analog domain and the digital domain simultaneously. The functional components implemented in SYS-2722 include an analog signal generator, a digital signal generator, an analog analyzer, a digital analyzer, and digital input and output and synchronization capabilities. "Cross-domain" testing (for example, providing an input in one domain to a converter such as an ADC or a DAC, and measuring the output in the second domain) is straightforward, and can be performed using optimal techniques and components for each domain.

The analog generator and analyzer have a number of hardware options that can be added to enhance the instrument's capability, including plug-in filters, noise and tone burst generator options and dedicated intermodulation and wow and flutter measurement modules.

Analog signals can be generated in analog circuitry, or can be generated digitally and converted into analog signals. In the same way, analog signals can be measured in the analog domain, or can be converted to the digital domain for digital analysis.

Digital signals are always generated and measured in the digital domain. Digital analysis is performed by one of seven DSP analysis tools, including a general-purpose audio analyzer, an FFT spectrum analyzer and a multitone analyzer. A dedicated 80 MHz ADC is connected across the digital input, and is available to digitize the serial interface bitstream for analysis.

These operations are all directed by AP2700, the 2700 series control software. Nearly every function of AP2700 can be automated using the Audio Precision programming language AP Basic, which is provided with the control software.

About This Manual

You're reading *Getting Started with Your 2700 Series Instrument*. This manual describes how to set up the 2700 series instrument hardware, and the AP2700 control software. It also contains a quick tutorial to familiarize you with the system. The manual is organized as follows:

- Chapter 1: Introduction.

- Chapter 2: Software installation and hardware setup information.

- Chapter 3: Description of the hardware and its connectors.

- Chapter 4: Description of the user interface.

- Chapter 5: Introductory tutorial.

- Chapter 6: Specifications.

- Chapter 7: GPIB Configuration

Other 2700 Series Documentation

Audio Precision publishes a variety of documents, many in electronic form, about hardware and software products, audio theory, and test and measurement techniques. The following list describes the documents that are included as Adobe Acrobat PDF files on the CD-ROM that is included with your instrument. They can also be downloaded from our Web site at audioprecision.com.

2700 Series User's Manual

The *2700 Series User's Manual* is the primary operation and reference manual for the system. Consult the *2700 Series User's Manual* for:

- Detailed descriptions of every 2700 series feature, software panel, control, and display.

- Reference information on audio test and measurement techniques, sample and utility file listings, and an audio glossary.

AP Basic Language Manual

You can create *tests* and *macros* to automate your measurements. Macros are written in the AP Basic programming language, which is a subset of Microsoft® Visual Basic®. The *AP Basic Language Manual* contains a list of the AP Basic commands, each with a full description and example of usage.

AP Basic Extensions Reference for the 2700 Series

The *AP Basic Extensions Reference for 2700 Series Instruments* contains a list of ActiveX Automation (OLE) commands that control the specific features of the 2700 series instrument hardware and the AP2700 control software.

Online Help

Much of the information contained in the *2700 Series User's Manual* is also available in the online help system included in the 2700 series control software. You can access the help system in two ways:

- **Context-sensitive Help**.
  Almost every control and display on each software panel has a help topic associated with it. To view the topic, click on the field or control, then press the **F1** function key.

- **The Help menu.**
  On the Menu bar, click **Help**. The Help menu offers commands to search the **2700 series Help** files or select a specific help library such as **AP Basic Language Help**.

Filter Design Package Manual

The 2700 series can access and use *downloadable filter files*. This feature allows you to install custom software high-pass, low-pass and weighting filters for use with the 2700 series analyzer.

The 2700 series CD-ROM includes a version of Momentum Data Systems' QEDLite filter design package. This special version creates filter files in the format used by the 2700 series. The *Filter Design Package* manual describes the operation of this software.

Other Publications

Audio Precision publishes application notes, technical papers, TECH NOTES, and other material. These publications cover all aspects of audio test and measurement. Information about out library of technical papers and announcements of new papers is available on the Audio Precision Web site at audioprecision.com.

GPIB Documentation for the 2700 series

When ordered in the optional GPIB configuration, a 2700 series instrument can also be controlled by the industry-standard General Purpose Interface Bus (GPIB).

Basic installation with GPIB is covered in Chapter 7 of this manual. For GPIB development and programming, order the *2700 Series GPIB Programmer's Reference Manual*, which includes a CD-ROM with GPIB sample program files.

Chapter 2
Installation and Setup

2700 Series Components

The following items are included with a standard, APIB-configured 2700 series instrument. For a GPIB-configured instrument, refer to Chapter 7.

- The 2700 series instrument chassis.

- A mains power cord.

- This manual, *Getting Started with Your 2700 Series Instrument*.

- The *2700 Series User's Manual.*

- The *AP Basic Extensions Reference for 2700 Series Instruments.*

- The *AP Basic Language Manual.*

- A packet of miscellaneous printed materials.

- A box containing the 2700 WIN-KIT interface kit.
  The following items are contained within the 2700 WIN-KIT:

  - An Audio Precision Interface Bus (APIB) interface card, as ordered[1].

  - An APIB interface cable.

  - A CD-ROM containing AP2700, the control software for the 2700 series, and documentation files.

We recommend that you keep the shipping box and packing materials to protect your instrument if you need to ship it in the future.

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[1] *The APIB interface card is available for PCI and PCMCIA slots. Refer to Page 13 for detailed installation information.*

Getting Up and Running

There are three main tasks to setting up your 2700 series instrument:

- Installing the 2700 series control software (AP2700) on the PC;

- Configuring your 2700 series instrument for the local mains power supply; and

- Installing the APIB card in the PC and connecting it to the 2700 series instrument.

These tasks are described below.

Installing AP2700 on a PC

PC System Requirements

Your 2700 series instrument must be connected to an IBM-compatible PC to operate. The PC must have the following minimum features and capabilities:

- Windows 2000 or Windows XP as the operating system.

  > *NOTE: You must have local administrator rights to install or operate AP2700 software. Go to User Accounts in the Windows Control Panel, or check with your network administrator.*

- At least the minimum processor type and memory required by Microsoft for the installed operating system.

- At least 300 MB of free hard disk space.

- A CD-ROM drive.

- An open PCI or PCMCIA slot to mount the APIB interface card.

- A color monitor and a video card with at least VGA capabilities. Video resolution of 1024×768 is recommended.

You will also need signal interface cables to connect your 2700 series instrument to your device under test (DUT). Since there are many possible connection configurations, signal cables are not provided with the instrument. Audio Precision offers cable kits with common connectors and adapters for purchase. Contact a sales representative or visit the Audio Precision Web site at audioprecision.com for more information.

Installing the Software

To install the 2700 series control software, use the following procedure:

- Close all Windows programs.

- Insert the AP2700 CD-ROM into the CD-ROM drive. The Windows Autorun feature should start the installation program automatically. If it does not, click on the Start menu and choose **Run**. Click **Browse** and choose the file Setup.exe on the AP2700 CD-ROM. Click **OK** in the Run dialog box.

- On the installation menu screen, click **Install AP2700 Software**. Follow the on-screen instructions. The installation program prompts you to enter the required information.

The installer may prompt you to reboot the machine. Although you do not need to reboot immediately, you must reboot before you can operate your 2700 series instrument.

Setting Up the Hardware

Connecting your 2700 Series Instrument to the Electrical Mains Supply

The 2700 series instrument must be connected to a 50–60 Hz alternating current (ac) electrical mains supply, maximum voltage 250 Vrms.

The instrument has been configured at the factory for the expected voltage at its intended destination, as ordered. The voltage setting and fusing arrangement will normally be correct unless the instrument has been transported into another area. The power entry module has a strip of indicator tape showing its mains voltage setting. This tape must be removed before use.

> You **MUST** be sure that the 2700 series instrument mains power configuration is correct for the electrical mains power supplied in your area. If you are not sure, do **not** plug the instrument into the mains power. Follow the instructions below to check or change the instrument mains supply voltage selection.

The mains power supply is applied to your 2700 series instrument through the power entry module located on the rear panel. Before connecting the power cord, confirm that the input voltage selection and fusing arrangement in the power entry module are correct for your mains power supply.
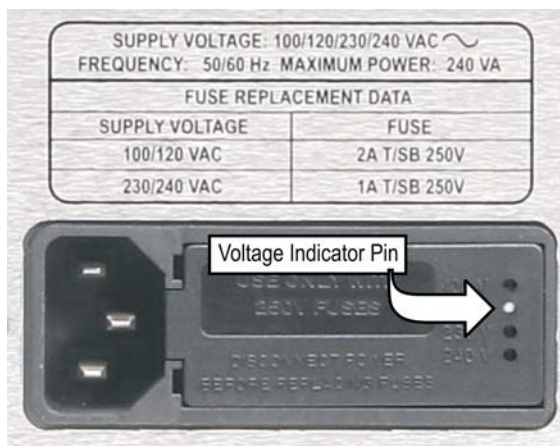
Figure 2. Detail, power entry module on 2700 series instrument rear panel.

Checking the Mains Supply Voltage Configuration

The voltage indicator pin protrudes through one of the four labeled holes in the module cover to indicate the selected input voltage, as shown in Figure 2. Check that the indicated voltage matches your mains supply voltage.

Opening the Power Entry Module

Unplug the power cord from the instrument before changing fuses or performing any other operations described in this section.

To open the Power Entry Module, refer to Figure 3 and proceed as follows:
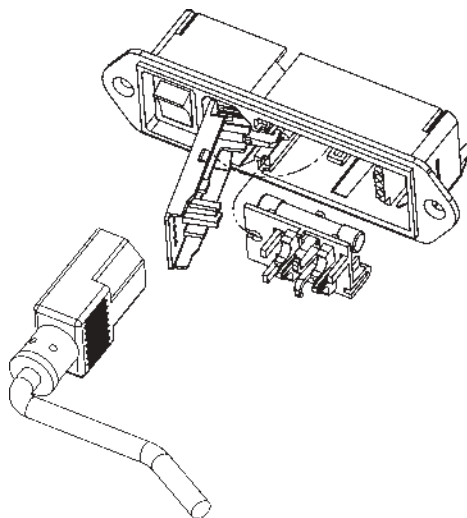
Figure 3. Power entry module door and fuse block.

- Remove the mains power supply cord from the power cord connector.

- Locate the slot in the module cover door hinge. The hinge is a the left side of the cover door, and the slot in the hinge is visible in the power cord connector cavity. Insert a small screwdriver or similar tool in the slot and pry the cover door hinge outward. The cover door will snap out, and then can be pivoted on its hinge for access to the fuse block assembly and voltage selector card.

Changing the Mains Supply Voltage Configuration

- Open the Power Entry Module as described above.

- The voltage selector card is a small circuit board fitted with a white plastic indicator pin, installed in a housing on the right side of the Power Entry Module as shown in Figure 4. Pull the voltage selector card straight out of the housing, using narrow pliers to grab the card. Do not use the indicator pin as a handle.
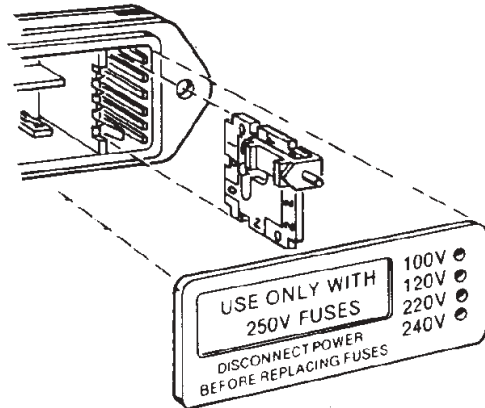
Figure 4. Changing the mains power supply voltage.

- Orient the selector card so that the desired input voltage is readable at the bottom, shown in Figure 5. Then move the indicator pin to point UP, opposite the indicated voltage. Seat the pin assembly in the notch on the board edge.

- Insert the voltage selector card into the housing with the printed side of the card facing toward the mains power connector. The card edge indicating the desired voltage should enter the housing first.

- Confirm that the correct fuse or fuse combination is installed for the intended input voltage (refer to the fuse ratings marked on the instrument rear panel). If necessary, change the fuse type as described in the following section.

100V   (90°)   120V   (90°)   230V   (90°)   240V
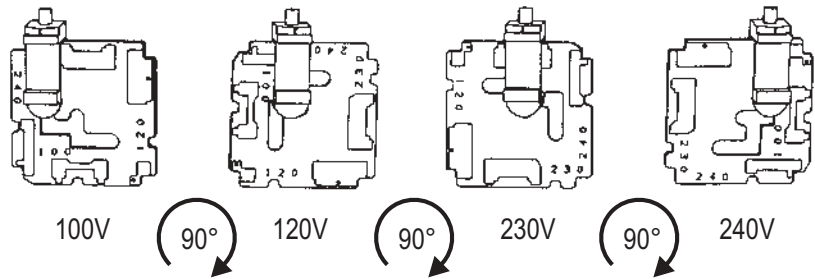
Figure 5. Voltage card selector orientation.

■ Close the module the cover door and verify that the indicator pin shows the desired voltage.

Once you have verified that the line voltage selection is correct, connect the power cord from a mains power outlet to the power cord connector on the instrument rear panel.

Fuse Information

The power entry module accommodates two fusing arrangements, as illustrated in Figure .

■ The North American fusing arrangement uses a single type 3AG (0.25" × 1.25") SB (slow blow) fuse.

■ The European fusing arrangement uses two 5×20 mm IEC-approved type T fuses.

Refer to the label on the rear panel for fuse current ratings.

Changing the Fusing Arrangement
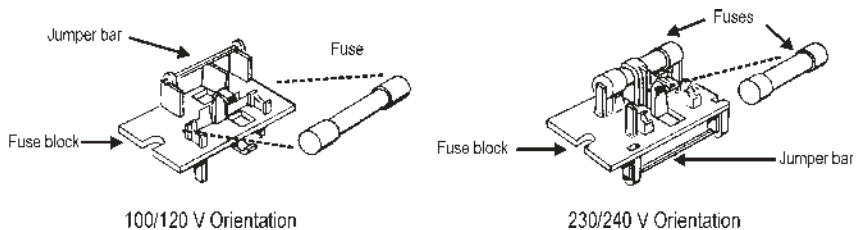
Figure 6. Fuse Block Orientation for North American (100/120 V) and European (230/240 V) operation..

To replace a fuse or change the fusing arrangement, proceed as follows:

■ Remove the mains power cord from the power cord connector and open the Power Entry Module as described above.

- Using narrow pliers, pull the fuse assembly out of the housing.

- Change or add the correct fuses as necessary, referring to Figure 6. Refer to the instrument rear panel for the correct fuse electrical current rating.

- Insert the fuse assembly in the housing, with the side of the assembly that carries the fuse(s) for your desired fusing arrangement facing into the housing. Press the fuse assembly firmly into the housing.

- Confirm that the line voltage selection is correct for your mains voltage and your fusing arrangement.

Once you have verified that the line voltage selection is correct, connect the power cord from a mains power outlet to the power cord connector on the instrument rear panel.

Connecting the Instrument to Your PC

Before connecting the 2700 series instrument to your PC, install the AP2700 control software. See page 9 for details.

The control software communicates with the 2700 series instrument through the Audio Precision Interface Bus (APIB). We offer two options for the APIB interface:

- The PCI card. Most newer PCs use the PCI bus exclusively; many PCs have a mix of both ISA and PCI bus slots available.

    Be sure that you have installed the AP2700 control software on your PC before you install the PCI card in your PC. If the card is installed before AP2700, Windows will not find the proper device driver.

- The PCMCIA card. This card allows you to use an existing PCMCIA interface found in many notebook computers.

Your system was shipped with an APIB interface cable and with the interface option you ordered. Refer to the appropriate section for your interface below.

    The APSI interface is only used with for the OPT-2711 Dolby* Digital Generator option, and is only functional with the installation of that option and the licensed activation of the software components. Instructions for use of APSI are included with the OPT-2711 Dolby Digital Generator option.

* Dolby and the double-D symbol are trademarks of Dolby Laboratories.

Installing a PCI APIB Card

Note: You must install the AP2700 software on your PC (see page 9) BEFORE you install the APIB card, so that Windows can locate the correct driver software.

To install the PCI card, use the following procedure:
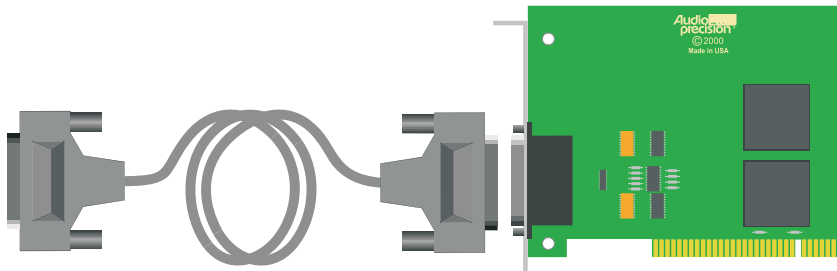
Figure 7. PCI card and APIB cable.

- Turn off your computer, remove the cover, and install the PCI APIB interface card in an open PCI bus slot. Remember to handle the card properly to minimize static electricity discharges while installing it. Be sure the card is well seated in the slot; it should snap down firmly into the bus connector.

- Secure the card bracket with the mounting screw and replace the computer cover.

- Connect the APIB cable between the 2700 series instrument and the PCI card.

- Turn on the computer. Windows should automatically find and identify the APIB interface and install the correct driver.

Using a PCM-WIN PCMCIA Interface Card
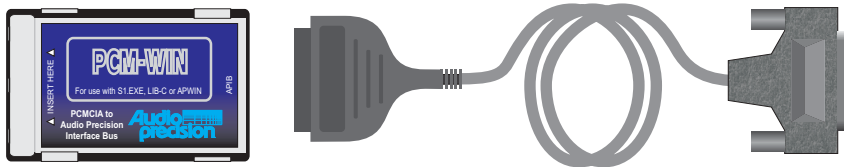
Figure 8. PCMCIA card and APIB cable.

> *Note: You must install the AP2700 software on your PC (see page 9) BEFORE you install the APIB card, so that Windows can locate the correct driver software.*

The Audio Precision APIB PCMCIA interface is most commonly used with laptop computers, but can be used in other computers with the appropriate adapter.

To use the PCMCIA card, you must have a PCMCIA interface already installed in your computer.

To install the PCMCIA card, use the following procedure:

- Insert the PCMCIA card into the PCMCIA interface.

- Connect the APIB cable between the 2700 series instrument and the PCMCIA card.

Chapter 3
Hardware Overview

The 2700 series instrument hardware includes input and output connectors, ranging and signal conditioning modules, digital signal processors, memory, and other circuitry needed to perform audio measurements. It is connected to the controller PC by the Audio Precision Interface Bus (APIB). This chapter describes the connectors, controls and indicators on the instrument front and rear panels. For a full list of hardware specifications, see Chapter 6.

When ordered in the optional GPIB configuration (SYS-27xxG), a 2700 series instrument can also be controlled by the industry-standard IEEE 488.2 General Purpose Interface Bus (GPIB). GPIB operation is not covered in this chapter. See Chapter 7 for more information about the GPIB configuration.

All signal generation and measurement is performed in the 2700 series instrument hardware, including storage of signals acquired for DSP waveform display or FFT analysis.

The SYS-2722 Front Panel

The SYS-2722 instrument front panel, shown in Figure 9, includes the digital and analog inputs and outputs, sync and monitor connectors, a headphone jack and volume control, and the mains power switch and power indicator.

Figure 9. SYS-2722 front panel.

- DIGITAL OUTPUT
  In single-connector mode, the 2700 series instrument transmits stereo digital audio simultaneously on the XLR "I" connector, the BNC connector, and the optical connector. In dual-connector mode, channels A and B are transmitted on the XLR and BNC "I" and "II" connectors, respectively.

- DIGITAL INPUT
  In single-connector mode, the instrument receives stereo digital audio from the XLR "I" connector, the BNC connector, or the optical connector. In dual-connector mode, channels A and B are received on the XLR or BNC "I" and "II" connectors, respectively.

- ANALOG OUTPUTS
  The instrument analog generator outputs (both "A" and "B" channels) are available as balanced signals on the XLR and dual banana connectors, and as unbalanced signals on the BNC connectors.

- ANALOG INPUTS
  The XLR and dual banana (balanced) and the BNC (unbalanced) connectors (both "A" and "B" channels) are connected to the instrument analyzer inputs.

- SIGNAL MONITOR PORTS
  The instrument provides seven signal monitor outputs suitable for audible monitoring, oscilloscope viewing or other monitoring purposes. Three of the signals originate within the Analog Analyzer. Four signals originate in the Digital Analyzer and are converted to the analog domain for monitoring.

- SYNC and TRIGGER PORTS
  Two Analog Generator monitor outputs are available, along with an Analog Generator Sync Output and an Analog Generator Trigger Input.

- GROUND CONNECTORS
  Four banana-jack / bare-wire-terminal ground connectors are provided, each connected to chassis ground.

- HEADPHONE JACK
  A 1/4" stereo phone jack connected to the audible monitor output. Inserting a plug into this jack interrupts the signal to the internal speaker.

- VOLUME CONTROL
  The volume control adjusts the audio volume of both the internal speaker and the headphone jack.

- POWER SWITCH and INDICATOR
  The power switch turns the mains power supply to the instrument hardware ON (I) or OFF (O).

See Chapter 4 of the *2700 Series User's Manual* for more information about the instrument inputs and outputs and the panels associated with them.

The 2700 Series Rear Panel

The SYS-2722 instrument rear panel, shown in Figure 10, includes the power entry module, sync and reference inputs and outputs, parallel digital input and output, APIB interface connectors and the instrument configuration and serial number labels.
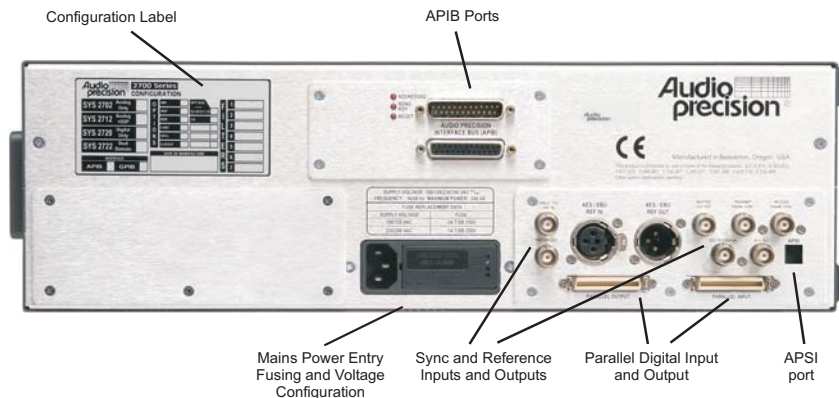
Figure 10. SYS-2722 rear panel.

- POWER ENTRY MODULE—this module includes:

  - POWER CORD CONNECTOR—This is a standard grounded connector for the mains power supply cord.

  - FUSE HOLDER/MAINS SUPPLY VOLTAGE JUMPER—Contains the mains power fuse and the mains supply voltage configuration jumper card. See Chapter 2.

  - MAINS SUPPLY VOLTAGE INDICATOR—The white tip of the plastic indicator on the voltage configuration jumper card appears in one of four small holes to show the mains voltage selection. See Chapter 2.

- SYNC and REFERENCE OUTPUTS

  - AES/EBU Reference

  - Master Clock Out

  - Transmitter Frame Sync Out

  - Receiver Frame Sync Out

- Aux Out

- Trigger Out

■ SYNC and REFERENCE INPUTS

- AES/EBU Reference

- Video/TTL Reference

- External Trigger In

■ PARALLEL PORTS
  Digital Audio Input and Output in parallel format, compatible with Audio Precision PSIA-2722 Programmable Serial Interface Adapter input and output.

■ APSI INTERFACE—This connector provides the two-way Audio Precision Streaming Interface. APSI is only used with for the OPT-2711 Dolby Digital Generator option, and is only functional with the installation of that option and the licensed activation of the software components.

■ APIB INTERFACE—These connectors provide the two-way Audio Precision Interface Bus connection between the instrument and the PC running the control software. For convenience, both a male and a female connector are provided; both connectors carry the same two-way data.

See Chapters 4 and 24 of the *2700 Series User's Manual* for more information about the instrument rear panel parallel digital and sync and reference inputs and outputs.

Chapter 4
2700 Series Control Software

Overview

Most of your interaction with your 2700 series instrument will be through the control software, AP2700. Once the system is plugged in, turned on and connected to your device under test (or *DUT*), you will be making all your adjustments and taking all your readings from your computer screen.

The 2700 series control software is a powerful and complex program, and most of the *2700 Series User's Manual* is dedicated to exploring and explaining its many capabilities. This chapter provides an introduction to the software features. Chapter 5 provides a brief tutorial on the basics of operating the system.

When ordered in the optional GPIB configuration (SYS-27xxG), a 2700 series instrument can also be controlled by the industry-standard IEEE 488.2 General Purpose Interface Bus (GPIB). GPIB operation is not covered in this chapter. See Chapter 7 for more information about the GPIB configuration.

The User Interface

Like many Windows applications, the 2700 series control software has a menu bar, a status bar, toolbars, and a workspace. With these tools you can save test and data files, export and print measurement results, configure both the hardware and software to your needs, access online help, design and launch automated processes and open and operate the measurement control panels.

The Workspace

The 2700 series workspace has five pages that you can select by menu commands (**View > Page #**) or by clicking the numbered tabs on the Status Bar.

The workspace is like a multi-layered desktop where you can display and organize the control panels you are using for a particular set of tests.
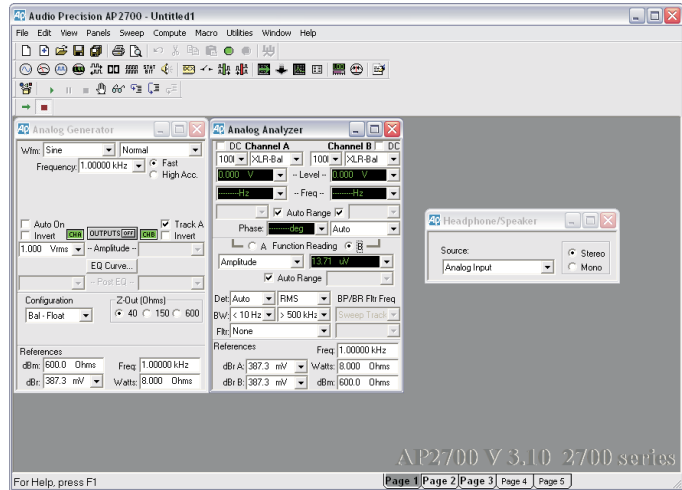
Figure 11. The 2700 series workspace.

Panels can be placed on any workspace page, and can be duplicated on more than one page. An control panel that appears on more than one page is simply a redundant display, for your convenience. The settings and results shown on redundant panels are identical, and a change made on one page is reflected on the others.

The 2700 series panels

The interaction between the control software and the 2700 series instrument is through control and measurement windows called *panels*. These can be thought of like the physical control panels often found on conventional test and measurement equipment, with range, function and trigger controls, ON and OFF switches, linear adjustment pots and sliders, numerical meter readouts, bar meters, X-Y waveform displays, graphs and so on.

The 2700 series panels are available through menu commands, toolbar icons and keyboard shortcuts.

You can control your 2700 series instrument from just one or two software panels, if that's all your measurement requires. A more typical test might involve four or five panels; a complicated test, perhaps six or seven.

Figure 12. A 2700 series panel.
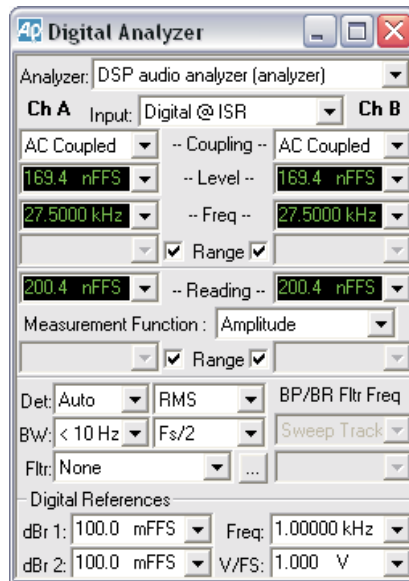
Although the 2700 series panels can all be placed on one workspace page, most users prefer to organize their workspace by placing panels on several different pages.

You can close a panel without canceling its function or effect; for example, you could set a signal's output voltage and frequency on the Analog Generator panel, turn the generator **ON**, then close the panel window. The signal will continue at the settings you've made until you open the panel again and make changes to the settings.

Many of the 2700 series panels come in two sizes, the smaller having fewer commands and readouts visible, and the larger showing every option. Double-click on the panel title bar to expand or contract it.

Panel Settings

Settings for tests and measurements are made using several different types of controls on the panels:

- ■ *Setting fields*, which come in several versions:

  - • The text box, in which you can directly enter a value from the keyboard.

  - • The drop-down list box, in which you can make a selection from a set of choices.
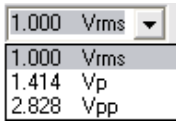
• The combo box, a combination of a text box with an added drop-down list from which you can choose the units to express your value.

Setting fields display dark (black or blue) text or numbers on a white field. 2700 series text boxes and combo boxes, where appropriate, will accept certain letters as shorthand for standard SI multipliers: "n" for nano- (10^{-9}), "u" for µ or micro- (10^{-6}), "m" for milli- (10^{-3}), "k" for kilo- (10^{3}) and "M" for mega- (10^{6}). For example, entering either "1001" and "1.001k" in a frequency field will set the frequency to 1.001 kHz.

■ *Check boxes*, which allow you to set non-exclusive conditions.

■ *Option buttons*, which provide for exclusive conditions.

■ *Browser buttons*, which open dialog boxes called *browsers*, which allow you to navigate lists of options, search for a file by browsing through a hierarchy of folders, and so on. Browser buttons are identified by the ellipses mark (…).

■ *Custom buttons* that control special functions, such as **GO**, **ON**, **OFF** and so on.
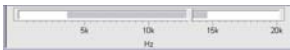
■ *Sliders*, with which you can continuously adjust a value. Sliders are found on Bar Graph panels. Sliders can be operated by either the mouse pointer or by using the arrow keys.

Panel Readings

Results of tests or measurements are called *readings*, which the 2700 series control software can display in several ways:
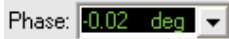
■ *Reading fields*, sometimes called *measurement fields* or *meters*. Reading fields come in two variations, with and without a drop-down list to select units.

Reading fields always display green text or numbers on a black field.

■ *Bar graphs,* which display values as a continuous colored bar, like a conventional analog meter.
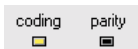
■ *Status indicators*, which show the presence or absence of a condition by a small colored rectangle.

The 2700 series menus

File   Edit   View   Panels   Sweep   Compute   Macro   Utilities   Window   Help

The 2700 series Menu bar extends across the top of the window and is always visible. In this section we will take a brief tour of the 2700 series menus. Details of menu options, controls and submenus will be explored in the user's manual.

The File Menu

The File menu offers commands for opening, saving, appending, importing, exporting and printing 2700 series files, and access to the Quick Launch menu.

File	
New	▶
Open	▶
Save	▶
Save As	▶
Save All	
Set Working Directory...	
Quick Launch	▶
Append	
Import...	▶
Export...	▶
Print...	▶
Print Preview	
Print Setup...	
Page Setup...	
1 C:\AP2700\last.at27	
2 C:\APWIN\last.at2c	
Exit	
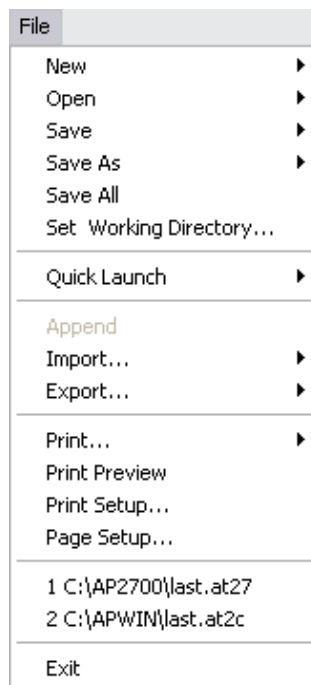
Since there are a number of different files types used by the 2700 series, submenus are available (at the small black arrows ▶ accompanying some menu choices) to choose, for example, whether to **Save As** a **Test**, **Data** or **Macro** file.

The listings at the bottom of the File menu show the last tests opened by the 2700 series, and provide a convenient way to quickly launch a previously used test. Simply click on the name of the test you would like to open.

The Edit Menu

Edit	
Undo	Ctrl+Z
Cut	Ctrl+X
Copy	Ctrl+C
Paste	Ctrl+V
Copy Panel to Clipboard	
Set Analyzer dBr Ref	F4
Set Analyzer Freq Ref	Ctrl+F4
Set Generator dBr Ref	F3
Set Generator Freq Ref	Ctrl+F3
Reset Bar-graph Max/Min	F2
Insert Row Before	
Insert Row After	
Add Row to the End	
Delete Row	

The Edit Menu provides several groups of commands:

- **Undo** and **Cut, Copy,** and **Paste** provide editing functions for 2700 series text entries.

- **Copy Panel to Clipboard** places a bitmap image of the selected 2700 series panel on the Windows clipboard.

- The **Set Analyzer** and **Set Generator** commands provide an easy way to set reference levels and frequencies for the Analog Analyzer and Analog Generator panels for relative measurements. These are discussed in detail in the user's manual.

- **Reset Bar-graph Max/Min** clears the gray minimum/maximum excursion bar from a selected Bar Graph display.

- The four last commands are only active when the Data Editor is open and selected, and provide the means to insert, add and delete data rows. See the user's manual for more information about using the Data Editor.

The View Menu

View
✔ Standard Toolbar
✔ Panel Toolbar
Macro Toolbar
✔ Learn Mode Toolbar
Quick Launch Bar
✔ Status Bar
Page 1   Ctrl+1
Page 2   Ctrl+2
Page 3   Ctrl+3
Page 4   Ctrl+4
Page 5   Ctrl+5
Attached Files
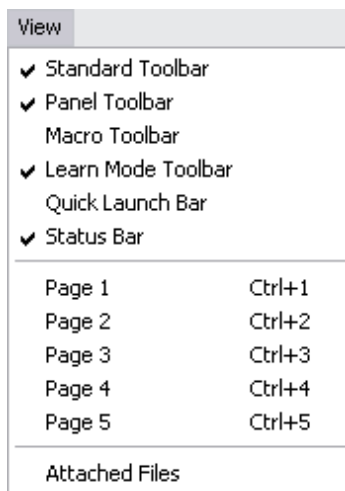
The View menu enables you to choose which workspace page to view, and whether or not the Status bar and any combination of Toolbars are displayed. A checkmark next to a View menu listing indicates that the selection is visible.

You can also open the Attached Files viewer from the View menu. Go the user's manual for more information about the Attached Files viewer.

The Panels Menu

Panels	
Analog Generator	Ctrl+G
Digital Generator	Ctrl+D
Analog Analyzer	Ctrl+A
Digital Analyzer	Ctrl+Y
Digital I/O	Ctrl+I
IEC 61937	▶
Sync/Ref Input/Output	
Status Bits	Ctrl+B
Headphone/Speaker	Ctrl+H
DCX	
Switcher	
PSIA Receiver	Ctrl+R
PSIA Transmitter	Ctrl+T
Sweep	Ctrl+S
Settling	
Regulation	
External Sweep	
Graph	
Data Editor	
Bar Graphs	▶
Macro Editor	
Diagnostic	

The Panels menu shows all the control panels available in the control software. Click on a Panel name (or on its corresponding Toolbar button) to open a panel on the current workspace page. Panels can be duplicated on more than one page for convenience.

> *Closing a panel does not stop its activity. If a generator panel has been set to output a sine wave, or an analyzer panel to measure a voltage, these activities continue whether or not the panel is visible on any page.*

The panel functions are discussed in detail in the *2700 Series User's Manual*.

The Status Bar

The Status bar runs across the bottom of the workspace. On the right, the Status bar displays the workspace page tabs; on the right are status notes updated as the control software operates.

The Toolbars

The 2700 series control software has five toolbars for quick access to panels and common functions. The toolbars can "float" at any location you place them on the workspace, or they can be "docked" or attached to the Menu bar. Whether docked or floating, the toolbars appear on every workspace page.

The toolbars display command buttons for one-click access to many often-used functions. See the *2700 Series User's Manual* for detailed information.

The Standard Toolbar

The Standard toolbar gives you quick access to 2700 series file, printer and editing operations, as well as Sweep Stop/Start and FFT Time/Frequency domain switching. See the *2700 Series User's Manual* for detailed information.

The Panels Toolbar
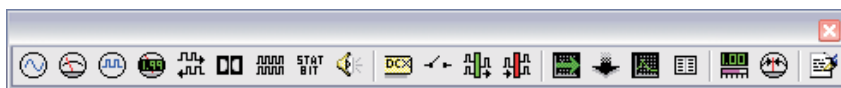
The Panels toolbar gives you quick access to 2700 series measurement and control panels: See the *2700 Series User's Manual* for detailed information.

The Macro Toolbar

The Macro toolbar offers one-click operations for running and editing an AP Basic macro. See the *2700 Series User's Manual* for detailed information.

The Learn Mode Toolbar

Learn mode logs all your 2700 series operations as you perform them, building a macro (script) for automation purposes. The Learn Mode toolbar gives you the ability to turn the macro recording function ON and OFF with a click of the mouse. See the *2700 Series User's Manual* for detailed information.

The Quick Launch Toolbar

Quick Launch allows you to create and use custom toolbar buttons to open 2700 series tests, run AP Basic macros and even launch other Windows programs.

From the Main menu, choose **File > Quick Launch > Customize Quick Launch** to open the Quick Launch Configuration panel, where you can view or edit the existing Quick Launch commands or to add new commands of your own.

Figure 13.
The Quick Launch
Configuration panel.
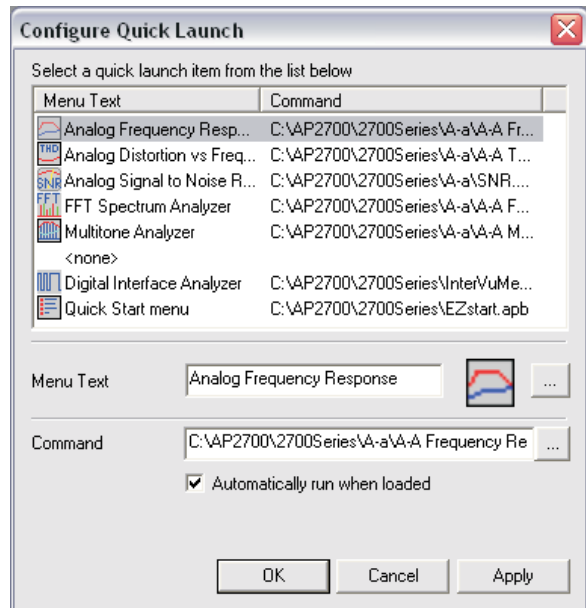
For each toolbar command, enter (or browse to) a command line with a complete path in the **Command** field. Name the command in the **Menu Text** field, and choose a toolbar button icon using the icon browser to the right. If you would like a macro from the Quick Launch toolbar to run immediately, check **Automatically run when loaded**.

To open the test, macro or program, click the button on the Quick Launch toolbar. In a default installation, the Quick Launch sample files are installed under …\Program Files\Audio Precision\AP2700\Quick Launch\.

See the *2700 Series User's Manual* for detailed information.

Chapter 5

Quick Guides

Introduction
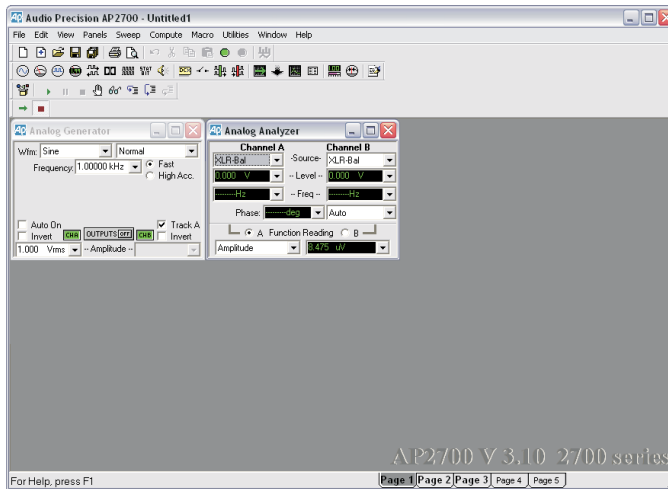
Figure 14. 2700 series workspace, New Test defaults.

The Quick Guides in this chapter provide a way to get familiar with the 2700 series user interface, and to get some immediate results from your instrument. As you become more comfortable with the system, please refer to the *2700 series User's Manual* for further information.

The Quick Guides begin with the assumption that your 2700 series instrument is turned on and connected to your PC, and that the AP2700 software has been launched and is in communication with and properly controlling your instrument. No audio cables or external devices under test (DUTs) are needed for the Quick Guides.

The Analog Signal Path

When the 2700 series control software is launched, the default Page 1 workspace is loaded with two panels, the Analog Generator and the Analog Analyzer, as shown in Figure 14. To return to this configuration at any time, click the **New Test** button. Previous settings and data will be discarded.

Analog Input Selection
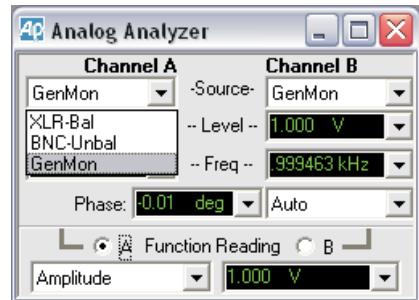
Figure 15. Analog input selection via Analog Analyzer Source drop-down lists.

The 2700 series has two channels of analog input that are accessed from the Analog Analyzer panel. Each channel can receive a signal from its front panel balanced or unbalanced connector, or via an internal path from the Analog Generator. The Source fields on the Analog Analyzer panel present these choices in a drop-down list. Go to the Analog Analyzer by selecting **Panels > Analog Analyzer** or by clicking the **Analog Analyzer** button. Choose **GenMon** for each channel to select the internal path.

Analog Generator Panel
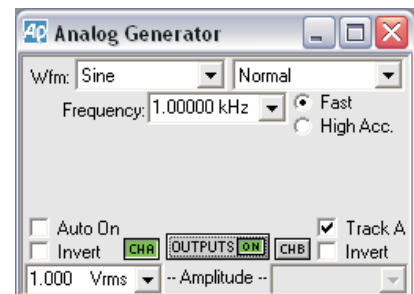
Figure 16. Analog Generator, OUTPUTS ON, CHB OFF.

In the default configuration, the Analog Generator panel is set for a **1 Vrms**, **1 kHz sine** wave, but the analog outputs are disabled. To turn on the outputs, click the OUTPUTS **OFF** button near the center of the panel. The button will turn green, and you will hear relays click in the instrument hardware. The Level meters on the Analog Analyzer panel will show approximately 1.000 V. The Level meters show the rms level of the signals at the analog inputs.

The stereo Analog Generator outputs can be individually disabled with the **CHA** and **CHB** buttons next to the OUTPUTS field. Click on the **CHB** button. The button will turn gray, relays will click in the instrument, and the Channel B Level meter on the Analog Analyzer panel will indicate a level of 0.000 V.

Now that we've successfully routed a signal from the Analog Generator to the Analog Analyzer, we will perform some basic measurements.

Using the Analog Analyzer

The 2700 series has two independent general-purpose analyzers that can perform measurements on analog signals. We will be using the Analog Analyzer for the Quick Guide; the alternative DSP Audio Analyzer is an analysis tool available on the Digital Analyzer panel that offers similar analysis capabilities. You can read about the both analyzers in Chapters 8 and 10 of the *2700 Series User's Manual*.

The Analog Analyzer is an analog-domain real-time audio analyzer, implemented in instrument hardware. It provides continuous *readings* of signal level and frequency for both input channels. It also allows you to select a channel and apply filtering to it, and to measure parameters such as total harmonic distortion, phase, and so on.

Figure 17. Analog Generator routed to Analog Analyzer inputs; generator Channel B is set OFF in this illustration.

At this point, you should have the setup shown in Figure 17:

■ Analog Generator Channel A producing a 1 V, 1 kHz sine wave.

- Analog Generator Channel B **OFF**.

- Analog Analyzer Channel A and B input Sources set to the internal generator path, **GenMon**.

The topmost meters in the Analog Analyzer panel (just below the Source selection fields) show the audio level in volts. You should see approximately 1 V in the left-hand meter (Channel A), and approximately 0 V in the rig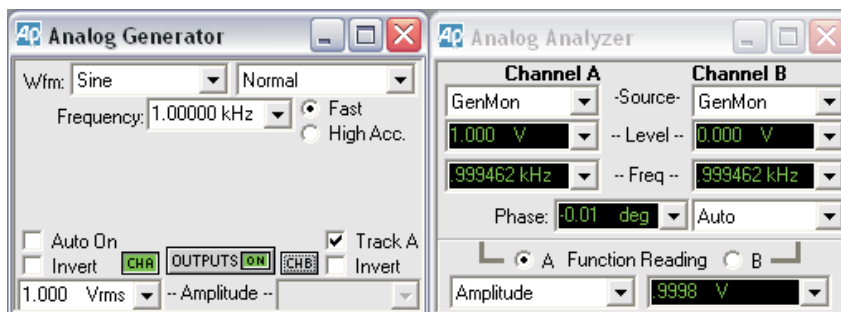ht-hand meter (Channel B). Click the **CHB** button on the Analog Generator panel to re-enable the Channel B generator signal. You now see approximately 1 V in the right-hand level meter on the Analog Analyzer panel, since the Channel B generator is now active and its signal is being routed to the analyzer via the **GenMon** internal path.

The meters in the next row of the Analog Analyzer panel show the frequency of the signal in each channel. Both of these meters should be reading very close to 1 kHz, the frequency of the analog generator.

Audible Signal Monitor
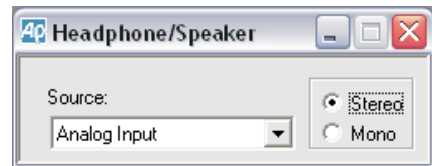
Figure 18. Headphone/Speaker panel set for Analog Input Source.

2700 series instruments have an internal mono speaker to monitor both analog and digital input signals. In addition, the signal at the speaker is available at the headphone jack on the instrument front panel. The volume control for signals applied to both the speaker and the headphone jack is located on the front panel next to the jack. When a headphone plug is inserted into the jack, the internal speaker is disconnected.

The signal applied to the speaker and headphone jack is selected on the Headphone/Speaker panel. Click the **Headphone/Speaker** button on the Panel toolbar, or go to **Panels > Headphone/Speaker** on the Main menu. On the Headphone/Speaker panel, drop the Source list and select **Analog Input**, and turn up the front panel volume control. You will hear the 1 kHz tone from the generator become louder. In the default New Test setup, the Headphone/Speaker mode is stereo, and the mono speaker in the instrument sums both channels of the signal. Click the CHA button on the Analog Generator to disable the channel A signal. The speaker level will drop by about half, since it is now reproducing only the channel B signal. Click CHA **ON** again, and the sound will get louder.

Controlling the Analog Generator
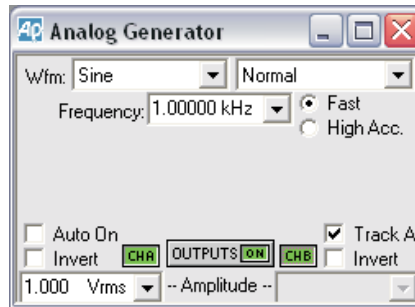
Figure 19. The Analog Generator.

The topmost two fields in the Analog Generator panel control the generator waveform. The Analog Generator is capable of producing sine waves, square waves, noise and other waveforms. At this point the waveform should be set to **Sine: Normal** (**Sine** in the left field, that is, and **Normal** in the right). Drop down the left-hand list box and choose **Square** to change the waveform to a square wave. If you are monitoring the signal with the internal speaker, you will hear the timbre of the signal change. You will also see the signal level change in the Analog Analyzer panel. (A square wave has a higher rms level than a sine wave for a given amplitude setting.)

The square waveform has one parameter or *setting* available: the **frequency**. This is set in the field below the waveform selection fields. We will now change the generator frequency to 2 kHz, as follows:

- Click in the Frequency field in the Analog Generator panel. The text **1.00000 kHz** becomes highlighted.

- Use the keyboard to type **2k**, the new generator frequency. As you can hear in the monitor speaker, the frequency does not change.

- Now press the **Enter** key. The text changes to **2.00000 kHz**, the pitch increases from the monitor speaker, and the frequency reading in the Analog Analyzer panel changes to about 2 kHz.

Now switch the waveform type back to sine. The current square wave frequency is used for the sine wave.

> *You can enter standard SI prefixes for multipliers as you enter the value in the setting field. For example, '2000', '2k', '0.002M', and '2000000m' all produce the same frequency.*

All waveforms can be generated at any *amplitude* from 0 V to the maximum output voltage of the instrument. The amplitude for each channel is controlled by fields below the OUTPUTS field. If the Track A checkbox is checked, the
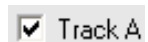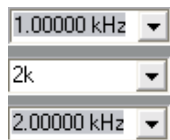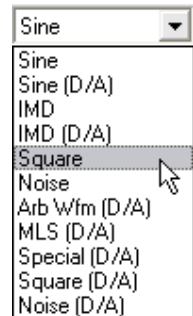
amplitude is identical in both channels and is set by the Channel A Amplitude field at the left of the panel. If Track A is unchecked, a Channel B Amplitude field appears at the right of the panel, and the amplitudes of the two channels can be varied independently.

Change the Channel A Amplitude to 0.5 V by clicking in the Amplitude field on the Analog Generator panel, typing **0.5**, and pressing **Enter**. The level meter reading on the Analog Analyzer panel will change. (You will also hear relays click if the system needs to change the analog range.) Now uncheck Track A and set the channel B amplitude to 10 V. The level meters on the Analyzer panel will show the different levels. Both frequency meters, however, will show the same frequency. This is because the **Sine: Normal** waveform generates identical frequencies in the two channels.

The **Sine (D/A): Stereo** waveform, which is obtained by choosing **Sine (D/A)** from the left-hand waveform field and **Stereo** from the right-hand waveform field, can generate independent frequencies in the two channels. You can verify the frequency in set each channel by referring to the frequency meters on the Analog Analyzer panel.

Units

Settings and readings consist of a *value* and a *unit*. The 2700 series can convert the units of a setting or reading to a related unit. For instance, the analog generator amplitude can be set in Vp (volts peak), dBV (decibels relative to 1 V), and several other units. Similarly, readings can be displayed on a meter in any of the related units.

The list of units for a setting or reading can be viewed by clicking the downward-pointing arrow next to the setting or reading field. For instance, the list of units for the analog generator amplitude can be displayed by clicking on the arrow next to the Amplitude field on the Analog Generator panel. The control software displays the value currently in the field, converted to each available unit. By clicking on the list, you can choose a setting unit.
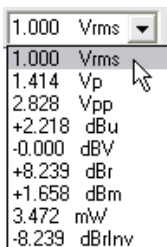
As an example, if the amplitude is set to 1 Vrms, the drop-down list includes conversions such as 1.414 Vp and 2.220 dBu. If you click on the line displaying the amplitude in Vp, the setting changes to 1.414 Vp. The generator amplitude has not changed; the setting is displayed in a different unit. You can now set the generator amplitude in Vp.

Some units, such as dBr, depend on a *reference* set in a reference field, such as those at the bottom of the large views of the Analog Generator or Analog Analyzer panels. Refer to Appendix A of the *2700 Series User's Manual* for more information about units of measurement.

Sweeps

Most device testing with the 2700 series is performed using *sweeps*. In a sweep, a setting is stepped over a pre-defined range, while selected readings are gathered. The result is a table of data relating the readings to each value of the setting.

For instance, if you wanted to measure the frequency response of an analog amplifier, you might vary the frequency of the Analog Generator from 20 Hz to 20 kHz in 30 logarithmic steps, while monitoring the Analog Analyzer Channel A and Channel B level readings. The result would be a table of data with 30 rows (one for each frequency step) and 3 columns (one for the frequency setting, one for the Channel A level reading, and one for the Channel B level reading).

> *You can force the 2700 series to perform a "single-point sweep" for special purposes, but generally the term* sweep *indicates that a number of data points are included in the measurement.*

When a sweep is performed, a *graph* is also created. This is a graphical representation of the data obtained during the sweep. Figure 20 shows an example of a stereo frequency response graph.
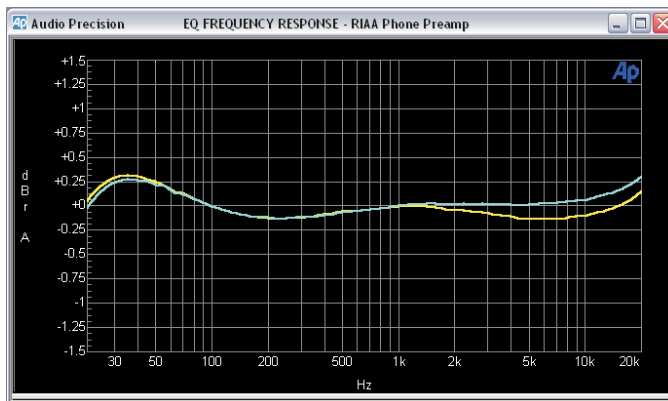
Figure 20. A typical frequency response graph.

The Sweep panel contains many more features than are described here. See Chapter 17 of the *2700 Series User's Manual* for more information.

Sweep Fundamentals

Settings and Readings

To set up a sweep, open the Sweep panel, either by choosing **Panels > Sweep**, or by clicking on the **Sweep** button. The Sweep panel is divided horizontally into two major areas: the **Data** (reading) area above, and the **Source** (setting) area below. During the sweep, the setting chosen in the **Source 1** field is varied, and the readings specified in the **Data** fields are gathered.
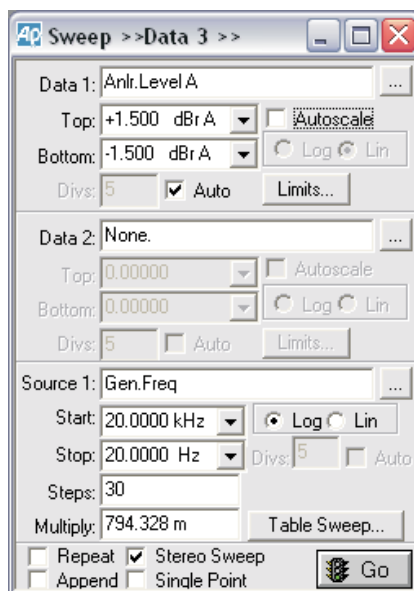
Figure 21. The Sweep panel. This configuration was used to create the graph of Figure 20.

To choose the *setting* to be varied, click on the ellipsis button to the right of the **Source 1** field. In the browser, choose an instrument from the left pane and a setting from the right pane. For instance, if you choose **Gen** from the instrument side and **Freq** from the setting side, the frequency of the Analog Generator will be varied during the sweep. This is shown in the **Source 1** field as **Gen.Freq**.
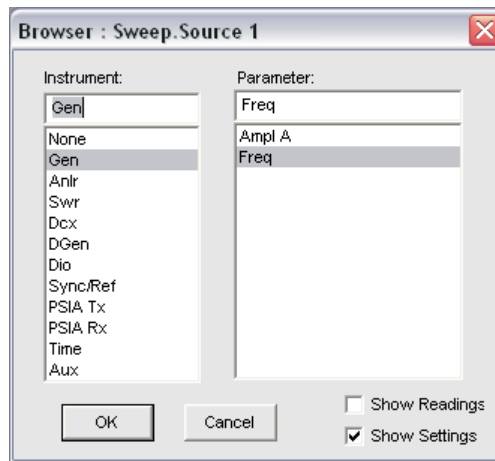
Figure 22. A Sweep browser. Appearance is similar for Source and Data browsers, for settings or for readings.

To choose a set of *readings* to be gathered, click on the ellipses button to the right of the **Data 1** field. The browser shows instruments in the left pane and readings in the right pane. If you choose the **Anlr** instrument and the **Level A** reading, the level on channel A will be recorded during the sweep. This is shown in the **Data 1** field as **Anlr.Level A**.

The Stereo Sweep checkbox near the bottom of the Sweep panel automatically ensures that readings from both channels are gathered during the sweep. For instance, if you choose **Anlr.Level A** as the reading in **Data 1** and then check the Stereo Sweep checkbox, the reading **Anlr.Level B** will also be gathered (in **Data 3**) during the sweep.
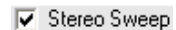
Source Range, Steps, and Spacing

During the sweep, the setting in **Source 1** is stepped from the Start value to the Stop value shown in fields below **Source 1**. The number of steps in the sweep is shown in the Steps field. A button to the right of these fields controls whether successive settings are separated *linearly* or *logarithmically*. In a linear sweep, the difference between successive settings is constant. In a logarithmic sweep, the ratio between successive settings is constant.

For example, if the Start frequency is 20 Hz, the Stop frequency is 20 kHz, there are 3 steps, and the Spacing is logarithmic, the control software will first set the generator to 20 Hz, then 200 Hz, then 2 kHz, and finally 20 kHz; that is, the *ratio* between successive frequency settings is 10.

On the other hand, if the Spacing is linear, the generator is first set to 20 Hz, then 6680 Hz, then 13340 Hz, and finally 20 kHz; that is, the *difference* between successive frequency settings is 6660 Hz.

Data Display Range

A graph is automatically created when a sweep is run. The Start and Stop values for **Source 1** define the extent of the horizontal (*x*-) axis. The extent of the vertical (*y*-) axis is determined by the values in the Top and Bottom fields beneath **Data 1**.

The units of the *y*-axis are the same as the units in the Top and Bottom fields. If you change the units of the Top or Bottom field, the units in the other field will also change. See Units on page 38 for more information on units.

Review

To set up a sweep, you must:

- *set **Source 1** to the setting to be varied during the sweep;*
- *set **Data 1** to the reading to be collected during the sweep;*
- *click Stereo Sweep if you wish to collect data from both channels;*
- *set the Start, Stop, Steps, and Lin/Log for **Source 1**; and*
- *set the Top and Bottom values and units for **Data 1**.*

*Once the sweep is set up, you can run it by clicking the **Go** button at the bottom of the Sweep panel.*

We will now perform two example sweeps: a frequency response sweep, and an amplitude linearity sweep.

Example Sweep: Frequency Response

For this example, we will measure the frequency response flatness of the 2700 series itself. We connect the analog output to the analog input, sweep the frequency of the analog generator over the audio range, and plot the level measured by the analyzer against frequency.

First, click the **New Test** button, turn the analog generator on, and set both Analog Analyzer input channels to **GenMon** so that the generator signal is sent directly to the analog input. Now click the **Sweep** button to open the Sweep panel. The default sweep is a mono frequency response sweep, so we only have to make a few modifications:

- Check the Stereo Sweep box to create a stereo sweep.

- Change the Steps from **30** to **100** for a more detailed response.

- The default range for **Data 1** is quite wide, and the units are Volts. Change the units to dBV by clicking on the arrow next to the Top field. From the list, choose the **dBV** line. (Note that the units in the Bottom

field change also.) In the Top field, type **0.1**. In the Bottom field, type **–0.1**.

You can now click the **Go** button to run the sweep. A graph is created, and the response is plotted as the sweep progresses. If you are monitoring using the internal speaker, you will hear the analog generator frequency step through the audio range. The result will be a response curve that deviates only slightly from 0 dBV, as shown in Figure 23.
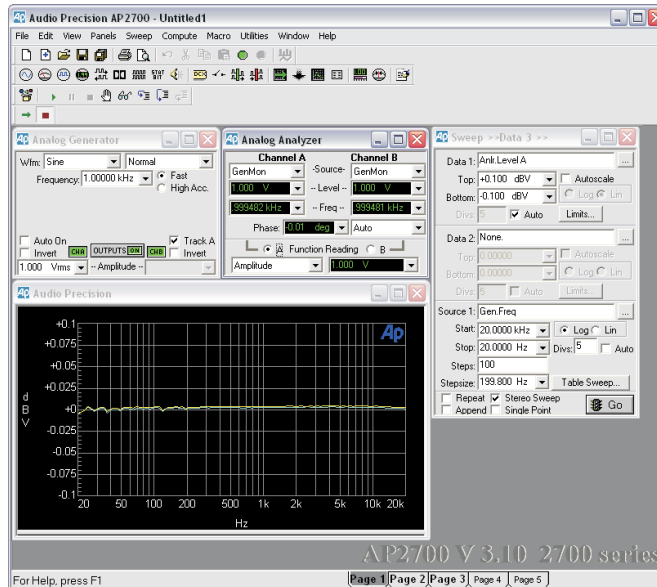
Figure 23. Current workspace after frequency response sweep, showing typical results.

Example Sweep: Amplitude Linearity

For this example, we will measure the amplitude linearity of the 2700 series instrument. With the Analog Generator output connected to the Analog Analyzer input, we sweep the amplitude of the Analog Generator over its full range, and plot the ratio of the level measured by the Analog Analyzer Level meters to the generator output level, in dB. Any amplitude non-linearity in the system will show as a deviation from 0 dB.

First, click **New Test**, turn the Analog Generator ON, and set the Analog Analyzer Source A and B to **GenMon** so that the generator signal is sent di-

rectly to the analog input. Now click the **Sweep** button to open the Sweep panel and make the following changes:

- Click the **Source 1** browser button, and choose **Gen** as the instrument and **Ampl A** as the setting.

- Select the dBg A units for **Data 1** by clicking on the arrow next to the Top field, and choosing the **dBg A** line from the list. The dBg A units (decibels relative to generator channel A) show the ratio of the reading to the level of the Analog Generator, in dB.

- Set the **Data 1** Top value to **0.1**, and the Bottom value to **–0.1**.

Click **Go** to start the sweep. The ranging relays will click as the generator changes amplitude. The result will be a curve that deviates only slightly from 0 dBg A, as shown in Figure 24.
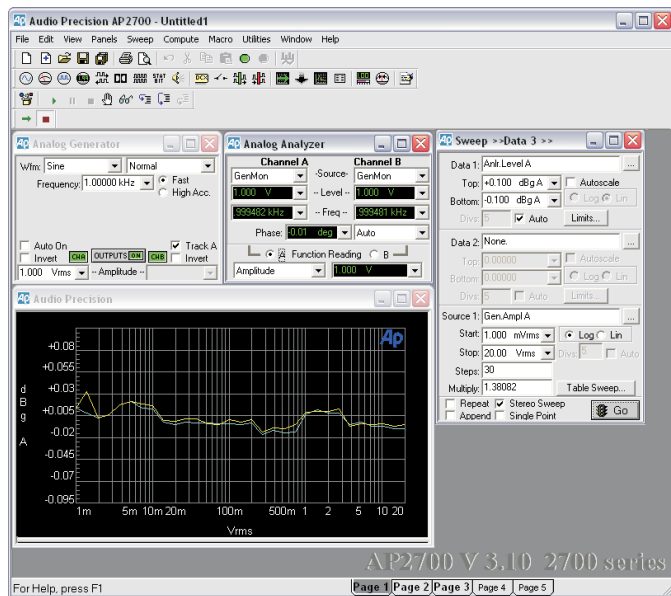
Figure 24. Current workspace after amplitude linearity sweep, showing typical results.

DSP Analysis

In addition to the hardware-implemented Analog Analyzer, the 2700 series provides a number of software-implemented analysis tools in the Digital Analyzer panel. These tools analyze digital domain signals using *digital signal processing* (DSP). The DSP Audio Analyzer, which offers similar capabilities to the Analog Analyzer, is one example. Another is the FFT Spectrum Analyzer, which we will look at next.

Analog signals must be converted from the analog domain to the digital domain for DSP analysis. When performing digital analysis of analog signals, the 2700 series uses the input circuits of the Analog Analyzer for interface and ranging, and then routes the analog signals to an analog-to-digital converter (ADC) before applying them to the Digital Analyzer for DSP analysis.

The FFT Spectrum Analyzer

Open the Digital Analyzer panel by choosing **Panels > Digital Analyzer** or by clicking the **Digital Analyzer** button.
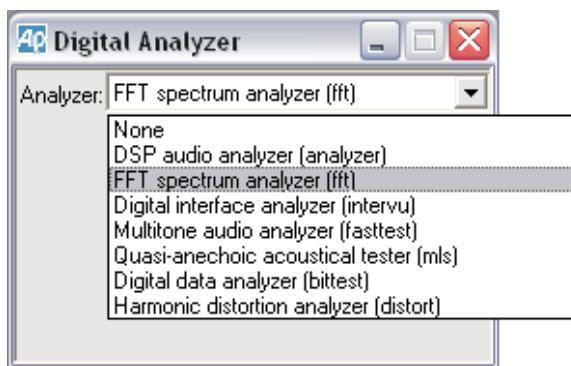
Figure 25. Digital Analyzer analysis tool selection.

At the top of the Digital Analyzer panel is a drop-down list of the DSP analysis tools available in the 2700 series.

Up to this point, we have been using the hardware-implemented Analog Analyzer to make measurements. We will now take a look at the *FFT Spectrum Analyzer*. This analysis tool displays audio signals in time (oscilloscope-style) and in frequency (spectrum analyzer-style). It operates in a different fashion from the Analog Analyzer, as explained next.

Real-Time vs. Batch-Mode Instruments

The Analog Analyzer is a *real-time* instrument. That is, it returns readings continuously, whether or not a sweep is in progress. These readings include level and frequency, as well as functions such as distortion or phase. During a sweep, readings from the audio analyzer are diverted from the panel meters to the sweep engine for graphing. After the sweep, the readings are once again displayed in the panel meters.

By contrast, the FFT Spectrum Analyzer is a *batch-mode* instrument. It does not generate readings if a sweep is not being performed. When a sweep is started, the Spectrum Analyzer first *acquires* a segment of audio by storing it in memory. This is indicated (briefly) on the Status bar as **DSP Acquiring Data**. Once the acquisition is complete, the Spectrum Analyzer processes the data. This is indicated as **DSP Transforming Data** (again, briefly) on the Status bar. Once processing is complete, the control software fetches a *batch* of readings from the Spectrum Analyzer. At that point, the data is graphed.

The FFT Concept

The FFT (fast Fourier transform) is an efficient way to decompose a periodic signal into its component frequencies. It transforms a *time-domain* signal (such as you might see in an oscilloscope trace) into a *frequency-domain* representation.

As an example, a high-quality power amplifier fed with a sinusoidal input will produce harmonic distortion at its output that is invisible in an oscilloscope trace. If an FFT is performed on the output signal, however, the individual distortion harmonics can be seen. The pattern of harmonics provides insight into the mechanisms responsible for the distortion.

The FFT process consists of the following steps:

- *Triggering*. The system waits for a trigger event from the selected source, for instance, a positive-going zero crossing of the analog generator.

- *Acquisition*. The system stores the input audio in memory until the selected number of audio samples has been received. The more audio data acquired, the more frequency points there will be in the result, and hence the higher the resolution will be. However, longer acquisitions take more time.

- *Windowing*. Before transformation, the audio data is shaped in time by multiplying it with a pre-defined envelope. This increases the accuracy of the amplitude values returned from the FFT.

- *Transformation*. The windowed audio data is transformed by the FFT. This results in an array of points describing the magnitude and phase of each frequency in the audio signal.

- *Display.* The control software fetches the transformed audio data and graphs it. Because the number of transformed points is typically larger than the number of points in the graph, a *peak-picking* method is used to ensure that peaks in the frequency data are not missed.

For more information, see Appendix C of the *2700 Series User's Manual*.

Setting up an FFT

Click **New Test** and open the Digital Analyzer panel. After a **New Test**, the Digital Analyzer panel defaults to **None** as its analyzer. To switch to the Spectrum Analyzer, click the down arrow next to the Analyzer field at the top of the Digital Analyzer panel. From the list, choose **FFT spectrum analyzer (fft)**. The panel changes to display the analysis tool.
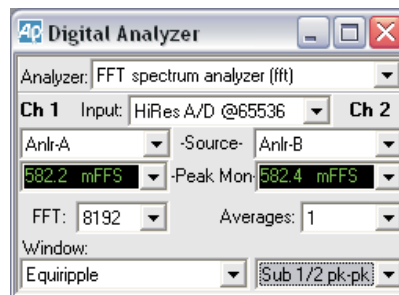
Figure 26. FFT Spectrum Analyzer panel.

Panel Fields

There are two peak monitors near the top of the panel, one for each of the stereo channels. These meters give an indication of the presence of signal at the FFT input.

Below the peak meters is an array of fields for configuring the analyzer. Here, we will use the default settings. For more details on the settings, see Chapter 11 of the *2700 Series User's Manual*.

Time vs. Frequency

As mentioned, the FFT spectrum analyzer can display audio signals in the time domain and the frequency domain. The choice of domain is determined by the **Source 1** field on the Sweep panel. If it is set to **Fft.FFT Time**, a time

domain display results. If it is set to **Fft.FFT Frequency**, a frequency domain display results.

The **Sweep Spectrum/Waveform** button in the Standard toolbar allows FFT sweeps to be set up quickly, entering **Fft.FFT Time** or **Fft.FFT Frequency** in the Source field and setting default ranges. We will use this feature to examine harmonic distortion in the 2700 series.

Quick Sweep Setup

Click **New Test**, turn the Analog Generator **ON**, and set the Analog Analyzer Source A and B to **GenMon** so that the generator signal is sent directly to the analog input.

Now open the Digital Analyzer panel and switch to the Spectrum Analyzer by selecting **FFT spectrum analyzer (fft)** from the drop-down list.
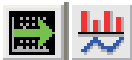
Open the Sweep panel and click the **Sweep Spectrum/Waveform** button on the Standard toolbar. The sweep is automatically set up for a stereo frequency domain display from 20 Hz to 20 kHz.

Click **Go** to perform the sweep. A graph similar to Figure 27 is created.
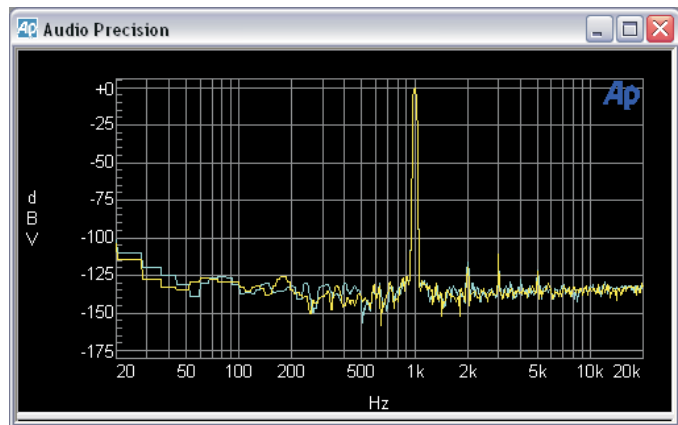
Figure 27. Typical default FFT stereo frequency domain graph.

The large peak in the middle of the spectrum is the 1 kHz, 1 V sine wave *fundamental* produced by the analog generator. There are smaller peaks spaced at 1 kHz intervals above the fundamental. These are *harmonic distortion* components generated in the 2700 series hardware. These harmonics rise out of a wideband system *noise floor.*

To look at the audio signal in the time domain, click the **Sweep Spectrum/Waveform** button again. **Source 1** changes to **Fft.FFT Time** and the graph updates to show the new data. You should see a graph similar to Figure 28.
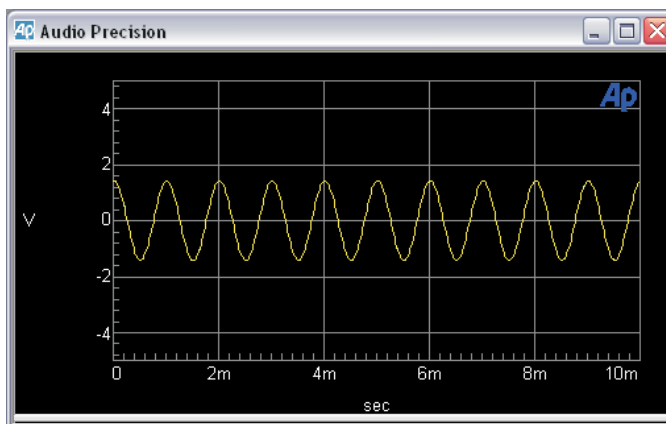
Figure 28. Typical default FFT stereo time domain graph.

The display shows a sine wave with a period of 1 ms (corresponding to a frequency of 1 kHz) and a peak excursion of 1.414 V (corresponding to an amplitude of 1 Vrms). The effect of the distortion on the waveform is not visible in the time domain.

You can switch between the time domain and frequency domain displays at any time by clicking the **Sweep Spectrum/Waveform** button.

Fine-Tuning the Display

There are several ways to adjust the graph display to more closely examine features of the acquired waveform:

- **Zoom**
  On the graph, draw a box by holding down the left mouse button and moving the mouse. When you release the button, the software zooms the selected area to the full graph size.

- **Zoomout**
  After zooming, click the right mouse button on the graph. From this right-click menu, choose **Zoomout**. The graph zooms out to the pre-zoom size.

- **Optimize**
  From the right-click menu, choose **Optimize Left Only**. The vertical axis changes to best fit the data in the graph.

For more details, see Chapter 18 of the *2700 Series User's Manual*.

Saving and Loading Tests

At any time, the current workspace setup can be saved to a file. The file contains the settings on all the panels (including panels that are not visible in the workspace), the panel positions, and references to any other files that were attached, such as limit data. The file also contains the most recent data obtained from any sweeps that were run.

When a test is loaded, the workspace reverts to the exact configuration it had when the test was saved. The data from the test is available and can be further processed. It can also be re-graphed automatically when the test is loaded. See Chapter 27 of the *2700 Series User's Manual* for configuration details.

To save the current workspace, choose **File > Save > Test**, or click the Save Test button in the Standard toolbar. To load a lest, choose **File > Open > Test**, or click the **Open Test** button in the Standard toolbar.

The Next Step

In this chapter, we have introduced two analysis tools: the Analog Analyzer, and the FFT Spectrum Analyzer. The 2700 series has other tools to measure audio signals in different ways. The *2700 Series User's Manual* describes all the analysis instruments in detail.

We have also introduced the *sweep*, which is at the heart of 2700 series automated measurements. The sweep engine and its features are described in more detail in Chapter 17 of the *2700 Series User's Manual*.

We set up tests by hand using the mouse and keyboard, and saved them to disk. The control software also contains a complete programming language called AP Basic that allows you to execute complex measurement sequences programmatically. Chapter 25 of the *2700 Series User's Manual* introduces programming and automation, the *AP Basic Manual* describes the language in detail, and the *AP Basic Extensions Reference for the 2700 Series Instruments* details the specific commands for this series.

Chapter 6
Specifications

Analog Signal Outputs

All 2700 series configurations except SYS-2720 contain an ultra-low distortion analog sine wave generator and two independent transformer-coupled output stages.

The SYS-2712 and SYS-2722 configurations also contain a dual-channel D/A signal generator for enhanced capabilities. Option "BUR" adds analog-generated sine burst, square wave, and noise signals. Option "IMD" adds analog-generated IMD test signals.

Unless otherwise noted, all specifications are valid only for outputs ≥150 µVrms [420 µVpp].

Analog Output Characteristics

Source Configuration	Selectable balanced, unbalanced, or CMTST (common mode test).
Source Impedances	
Balanced or CMTST	40 Ω (±1 Ω), 150 Ω[1] (±1.5 Ω), or 600 Ω (±3 Ω).
Unbalanced	20 Ω (±1 Ω) or 600 Ω (±3 Ω).
Max Floating Voltage	42 Vpk (outputs are isolated from each other).
Output Current Limit	Typically >80 mA.
Max Output Power into 600 Ω	
Balanced	+30.1 dBm (Rs = 40 Ω).
Unbalanced	+24.4 dBm (Rs = 20 Ω).
Output Related Crosstalk	
10 Hz–20 kHz	≤ −120 dB or 5 µV, whichever is greater.
20 kHz–100 kHz	≤ −106 dB or 10 µV, whichever is greater.

Low Distortion Sine Wave Generator

Frequency Range	10 Hz–204 kHz.
Frequency Accuracy	
High-accuracy mode	±0.03%.
Fast mode	±0.5%.

NP0020.0006.005

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[1] *200 Ω ±2 Ω with option "EURZ"*

Frequency Resolution	
High-accuracy mode	0.005%.
Fast mode	0.025 Hz, 10 Hz–204.75 Hz, 0.25 Hz, 205 Hz–2.0475 kHz, 2.5 Hz, 2.05 kHz–20.475 kHz, 25 Hz, 20.5 kHz–204 kHz.
Amplitude Range[2]	
Balanced	<10 µV–26.66 Vrms [+30.7 dBu].
Unbalanced	<10 µV–13.33 Vrms [+24.7 dBu].
Amplitude Accuracy	±0.7% [±0.06 dB] at 1 kHz.
Amplitude Resolution	0.003 dB or 0.05 µVrms, whichever is larger.
Flatness (1 kHz ref)	
10 Hz–20 kHz	±0.008 dB (typically <0.003 dB).
20 kHz–50 kHz	±0.03 dB.
50 kHz–120 kHz	±0.10 dB.
120 kHz–200 kHz	+0.2 / –0.3 dB.
Residual THD+N[3,4]	
At 1 kHz	≤(0.00025% + 1.0 µV) [–112 dB], 22 kHz BW (valid only for analyzer inputs ≤8.5 Vrms).
20 Hz–20 kHz	≤(0.0003% + 1.0 µV) [–110.5 dB], 22 kHz BW, ≤(0.0005% + 2.0 µV) [–106 dB], 80 kHz BW, ≤(0.0010% + 5.0 µV) [–100 dB], 500 kHz BW.
10 Hz–100 kHz	≤(0.0040% + 5.0 µV) [–88 dB], 500 kHz BW.

Intermodulation Distortion Test Signals

with option "IMD"

SMPTE (or DIN)

LF Tone	40, 50, 60, 70, 100, 125, 250, or 500 Hz, all ±1.5%.
HF Tone Range	2 kHz–200 kHz.
Mix Ratio	4:1 or 1:1 (LF:HF).
Amplitude Range[5]	
Balanced	30 µVpp–75.4 Vpp.
Unbalanced	30 µVpp–37.7 Vpp.
Amplitude Accuracy	±2.0% [±0.17 dB].
Residual IMD[6]	0.0015% [–96.5 dB], 60 Hz + 7 kHz or 250 Hz + 8 kHz.

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[2] *20 Hz–50 kHz only. Decrease maximum output by a factor of 2 (–6.02 dB) for the full 10 Hz–204 kHz range.*

[3] *System specification measured with the 2700 series analog analyzer set to the indicated measurement bandwidth (BW). Generator amplitude setting must be ≤12 Vrms balanced or ≤6 Vrms unbalanced for specified performance below 30 Hz. At higher amplitude settings generator THD derates to 0.0020% from 20 Hz–30 Hz.*

[4] *Individual harmonics are typically <–130 dBc at 1 kHz, and <–120 dBc from 25 Hz to 20 kHz measured with a passive notch filter and FFT analyzer.*

[5] *Calibration with other amplitude units is based upon an equivalent sinewave having the same Vpp amplitude.*

[6] *System specification measured with the 2700 series analog analyzer at any amplitude ≥200 mVrms.*

CCIF and DFD

Difference Frequency	80, 100, 120, 140, 200, 250, 500 or 1 kHz; all ±1.5%.
Center Frequency	4.5 kHz–200 kHz.
Amplitude Range[5]	
Balanced	30 µVpp–75.4 Vpp.
Unbalanced	30 µVpp–37.7 Vpp.
Amplitude Accuracy	±3.0% [±0.26 dB].
CCIF Residual IMD[6]	≤0.0004% [–108 dB], 14 kHz+15 kHz (odd order & spurious typ <0.05%).
DFD Residual IMD[6]	≤0.0002% [–114 dB], 14 kHz+15 kHz (odd order & spurious typ <0.025%).

DIM (or TIM)

Squarewave Frequency	3.15 kHz (DIM-30 and DIM-100), 2.96 kHz (DIM-B); both ±1%.
Sinewave Frequency	15 kHz (DIM-30 and DIM-100), 14 kHz (DIM-B).
Amplitude Range[4]	
Balanced	30 µVpp–75.4 Vpp.
Unbalanced	30 µVpp–37.7 Vpp.
Amplitude Accuracy	±2.0% [±0.17 dB].
Residual IMD[5]	≤0.0020% [–94 dB].

Special Purpose Signals

with option "BUR"

Sine Burst

Frequency Range	20 Hz–100 kHz.
Frequency Accuracy	Same as Sinewave.
ON Amplitude Range Accuracy, Flatness	Same as Sinewave.
OFF Ratio Range	0 dB to –80 dB.
OFF Ratio Accuracy	±0.3 dB, 0 to –60 dB.
ON Duration	1 cycle–65535 cycles, or externally gated.
Interval Range	2 cycle–65536 cycles.

Square Wave

Frequency Range	20 Hz–20 kHz.
Frequency Accuracy	Same as Sinewave.
Amplitude Range[4]	
Balanced	30 µVpp–37.7 Vpp.
Unbalanced	30 µVpp–18.8 Vpp.
Amplitude Accuracy	±2.0% [±0.17 dB] at 400 Hz.
Rise/fall time	Typically 2.0 µs.

Noise Signals

White Noise	Bandwidth limited 10 Hz–23 kHz.
Pink Noise	Bandwidth limited 20 Hz–200 kHz.
Bandpass Noise	Approximately 1/3-octave (2-pole) filtered pink noise, continuously tunable from 20 Hz–100 kHz.
Generator	True random or pseudo-random.
Pseudo-Random Interval	Typically 262 ms (synchronized to the analyzer 4/s reading rate).
Amplitude Range[5]	(Approximate calibration only).
Balanced	30 µVpp–37.7 Vpp.
Unbalanced	30 µVpp–18.8 Vpp.

Graphs of Typical Analog Generator Performance
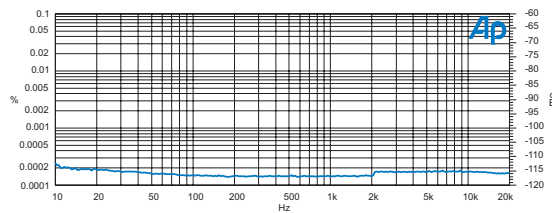
Figure 29. Typical system THD+N versus Frequency at 2 Vrms (analog sine).
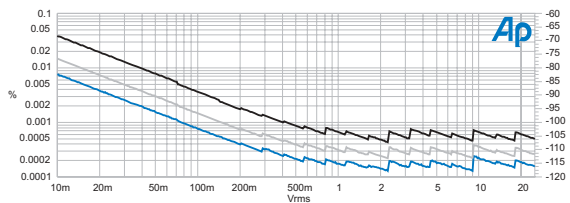
Figure 30. Typical system THD+N versus amplitude at 1 kHz. Lower trace is with 22 kHz bandwidth limiting. Middle trace is with 80 kHz.
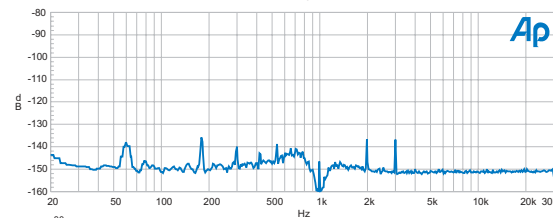
Figure 31. Typical residual THD+N spectrum at 1 kHz, 2 Vrms. (32768 point FFT of notch filter output, Sample Rate = 65.536 ks/s, 16 averages).
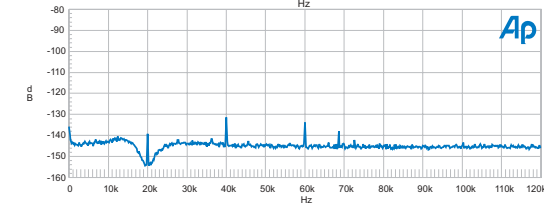
Figure 32. Typical residual THD+N spectrum at 20 kHz, 2 Vrms. (32768 point FFT of notch filter output, Sample Rate = 262 ks/s, 16 averages).
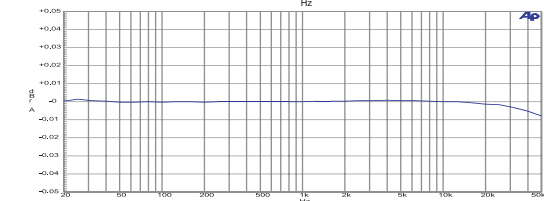
Figure 33. Typical analog system flatness at 2 Vrms signal level (measured with the analog analyzer's Level meter, dc input coupling).

D/A Generated Analog Signals

Available only in the SYS-2720 and SYS-2722 configurations. Except for arbitrary and Pass Thru waveforms, the digitally generated analog output signals and the embedded output signals are independently selectable and concurrently available. If both analog and digital outputs are selecting arbitrary waveform, it must be the same one. If Pass Thru waveforms are selected for both analog and digital generators, they must be at the same sample rate.

Common Specifications

Sample Rate	
Sine, IMD signals	fixed at 65.536 ks/s or 131.072 ks/s.
Other signals	8 ks/s–108 ks/s variable, or fixed at 65.536 ks/s or 131.072 ks/s.
Frequency Accuracy	±0.0002% [2 PPM] internal reference, lockable to external reference.
D/A Resolution	24-bit sigma-delta.

"SINE (D/A)" Signal Family

The Sine family includes "Normal," "Var Phase," "Stereo," "Dual," "Shaped Burst," and "EQ Sine." Normal and EQ Sine produce a monaural signal with the best (lowest) residual THD+N performance. EQ Sine varies the amplitude in accordance with a selected EQ file. Var Phase produces the same sine wave in both channels but with settable phase offset. Stereo provides sine waves of independently settable frequency in each channel (phase is random if both frequencies are set equal). Dual produces a monaural test signal containing a mixture of two sine waves of independently settable frequency and amplitude ratio. Shaped Burst produces a monaural sine burst signal with a raised cosine amplitude envelope (see Figure 58 on page 77).

|	
Frequency Ranges	10 Hz–30 kHz (65.536 ks/s), or[7] 10 Hz–60 kHz (131.072 ks/s).
Frequency Resolution	Sample Rate ÷ 2^{23}. [0.0078 Hz in the 30 kHz range].
Flatness (1 kHz ref)	
20 Hz–20 kHz	±0.01 dB.
10 Hz–30 kHz	±0.03 dB.
30 kHz–50 kHz	±0.10 dB (typically –0.5 dB at 60 kHz).
THD+N[8] (20Hz–20kHz)	
30 kHz range	0.0007% [–103 dB].
60 kHz range	0.0014% [–97 dB].
Variable Phase Range	–180.0 to +179.9 deg.
Dual-Sine Ratio Range	0 dB to –100 dB, usable to –138 dB.
Shaped Burst Interval	2 cycle–65536 cycles.
Shaped Burst On Time	1 to (number of interval cycles minus 1).

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[7] *Significant generator alias products will appear for frequency settings above 29.5 kHz in the '30k' range and 53.5 kHz in the '60k' range.*

[8] *System specification measured with the 2700 series analog analyzer set for a 22 kHz measurement BW.*

"IMD (D/A)" Signal Family

SMPTE/DIN Test Signal

LF Tone	40 Hz–500 Hz.
HF Tone	2.00 kHz–50 kHz.
Mix Ratio	4:1 or 1:1 (LF:HF).
Residual IMD[7]	≤0.0010% [–100 dB], 60/7 kHz or 250/8 kHz.

CCIF/DFD Test Signal

Difference Frequency	80 Hz–2 kHz.
Center Frequency	4.50 kHz to >50 kHz.
Residual CCIF IMD[9]	CCIF: ≤0.0004% [–108 dB], 14 kHz/15 kHz. DFD: ≤0.0002% [–114 dB], 14 kHz/15 kHz.

DIM Test Signal

Squarewave Frequency	3.15 kHz for DIM30 and DIM100, 2.96 kHz for DIMB.
Sinewave Frequency	15.00 kHz for DIM30 and DIM100, 14.00 kHz for DIMB.
Residual IMD[8]	≤0.0020% [–94 dB].

Other Signals

Arbitrary and Multitone Waveforms ("Arb Wfm")

Signal	Determined by the associated file specified in the panel drop-down box.
Frequency Range	20 Hz to 47% of Sample Rate.
Length	256 points–16384 points per channel. Utility is provided to prepare waveform from frequency, amplitude, and phase data.
Frequency Resolution	Sample Rate ÷ Length [2.93 Hz at 48 ks/s for a waveform 16384 points in length].
Maximum Number of Tones	(Length / 2) –1 [8191 for Length = 16384].

Maximum Length Sequence ("MLS")

Sequences	Four pink, four white.
Sequence Length	"32k" (32767) or "128k" (131071).
Frequency Range	10 Hz to 43% of Sample Rate, ±0.1 dB.

Special Signals

Polarity	Sum of two sine waves phased for reinforcement with normal polarity.
Pass Thru	Passes the embedded audio signal from the rear panel Reference Input. Ratio of reference rate to output Sample Rate may not exceed 8:1.

[9] *System specification measured with the 2700 series analog analyzer at any voltage ≥ 200 mVrms.*

Squarewave

Frequency Range	20 Hz–20.0 kHz.
Rise Time	Typically 2.0 µs.

Noise Signal

Pseudo-random white

Analog Analyzer

All 2700 series configurations except SYS-2720 contain an input module with two independent auto-ranging input stages, each having its own level (rms) and frequency meters; a phase meter; plus a single channel multi-function analyzer module providing additional signal processing and gain stages. Standard analog analyzer functions include amplitude and noise (both wideband and selective), THD+N, and crosstalk.

The SYS-2712 and SYS-2722 configurations add dual-channel A/D converters for FFT and other special forms of analysis. Option "IMD" adds inter-modulation distortion measurement capability. Option "W&F" adds wow & flutter measurement capability.

Unless otherwise noted, all specifications assume dc coupling, rms detection, and auto-ranging operation.

Analog Input Characteristics

Input Ranges	40 mV–160 V in 6.02 dB steps.
Maximum Rated Input	230 Vpk, 160 Vrms (dc to 20 kHz), overload protected in all ranges.
Input Impedance	
Balanced	200 kΩ / 95 pF (differential).
Unbalanced	100 kΩ / 185 pF.
Terminations	Selectable 600 Ω or 300 Ω, each ±1%, 1 Watt [+30 dBm] maximum power.
CMRR[10]	
40 mV–2.5 V ranges	≥80 dB, 10 Hz–20 kHz.
5 V and 10 V ranges	≥65 dB, 10 Hz–20 kHz.
20 V–160 V ranges	≥50 dB, 10 Hz–1 kHz.
Input Related Crosstalk	
10 Hz–20 kHz	≤ –140 dB or 1 µV, whichever is greater.
20 kHz–100 kHz	≤ –126 dB or 2.5 µV, whichever is greater.

Level Meter Related

Measurement Range	5 mV–160 V for specified accuracy and flatness, usable to <100 µV.
Resolution (full scale)[11]	
4/s and 8/s	1/40,000 [0.00022 dB].
16/s	1/20,000 [0.00043 dB].
32/s	1/10,000 [0.00087 dB].
64/s	1/5,000 [0.0017 dB].
128/s	1/2,500 [0.0035 dB].
Accuracy (1 kHz)	±0.5% [±0.05 dB].

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[10] *Not valid below 50 Hz with ac coupling.*

[11] *Resolution within a given range is equal to its full scale value multiplied by the fraction indicated for the selected reading rate. (Example: 40 mV input range reading resolution = 4 µV, using the 32/s reading rate). Numerical displays using a dB unit are rounded to the nearest 0.001 dB.*

Flatness (1 kHz ref)[12]	
20 Hz–20 kHz	±0.008 dB (typically <0.003 dB).
15 Hz–50 kHz	±0.03 dB.
10 Hz–120 kHz	±0.10 dB.
120 kHz–200 kHz	+0.2 / –0.3 dB (typically <–0.5 dB at 500 kHz).

Frequency Meter Related

Measurement Range	10 Hz–500 kHz.
Accuracy	±0.0006% [±6 PPM].
Resolution	6 digits + 0.000244 Hz.
Minimum Input	5 mV.

Phase Measurement Related

Measurement Ranges	±180, –90 / +270, or 0 / +360 deg.
Accuracy[13]	
10 Hz–5 kHz	±0.5 deg.
5 kHz–20 kHz	±1 deg.
20 kHz–50 kHz	±2 deg.
Resolution	0.1 deg.
Minimum Input	5 mV, both inputs.

Wideband Amplitude/Noise Function

Measurement Range	<1 µV–160 Vrms.
Accuracy (1 kHz)	±1.0% [±0.09 dB].
Flatness (1 kHz ref)[11]	
20 Hz–20 kHz	±0.02 dB.
15 Hz–50 kHz	±0.05 dB.
50 kHz–120 kHz	±0.15 dB.
120 kHz–200 kHz	+0.2 dB / –0.3 dB (typically < –3 dB at 500 kHz).

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[12] *Derate flatness above 5 kHz by an additional ±0.02 dB in the 20 V, 40 V, 80 V, and 160 V input ranges.*

[13] *Both analyzer input channels must have same coupling (ac or dc) selection. Accuracy is valid for any input signal amplitude ratio up to ±30 dB.*

Bandwidth Limiting Filters	–see Figure 34
LF –3 dB	<10 Hz, 22 Hz per IEC468 (CCIR), 100 Hz ±5% (3-pole), or 400 Hz ±5% (3-pole).
HF –3 dB	22 kHz per IEC468 (CCIR), 30 kHz ±5% (3-pole), 80 kHz ±5% (3-pole), or >500 kHz.
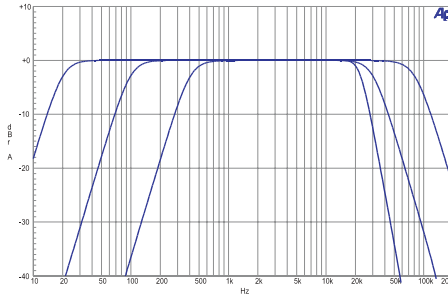
Figure 34. Typical responses of the standard band-limiting filters.

Optional Filters	up to 7 *(see section on Option Filters)*.
Detection	RMS (τ = 25 ms or 50 ms), Average, QPk per IEC468 (CCIR), Pk (pseudo-peak), or S-Pk (0.7071 × Pk reading).

Residual Noise	
22 Hz–22 kHz BW	≤1.0 µV [–117.8 dBu].
80 kHz BW	≤2.0 µV [–111.8 dBu].
500 kHz BW	≤6.0 µV [–102.2 dBu].
A-weighted	≤0.7 µV [–120.9 dBu].
CCIR-QPk	≤3.5 µV [–106.9 dBu].

Bandpass Amplitude Function

Tuning Range (f_o)	10 Hz–200 kHz.
Tuning Accuracy	±2%.
Bandpass Response	1/3-octave class II (4-pole), < –32 dB at 0.5 f_o and 2.0 f_o.
Accuracy (at f_o)	±0.3 dB, 20 Hz–120 kHz.
Residual Noise	
10 Hz–5 kHz	≤0.25 µV [–130 dBu].
5 kHz–20 kHz	≤0.5 µV [–124 dBu].
20 kHz–200 kHz	≤1.5 µV [–114 dBu].

Bandreject Amplitude Function

Tuning Range (f_o)	10 Hz–200 kHz.
Tuning Accuracy	±2%.
Bandreject Response	typically –3 dB at 0.73 f_o & 1.37 f_o, –20 dB at f_o ±10%, –40 dB at f_o ±2.5%.
Accuracy	±0.3 dB, 20 Hz–120 kHz (excluding 0.5 f_o–2.0 f_o).

THD+N Function

Fundamental Range	10 Hz–200 kHz.
Measurement Range	0%–100%.
Accuracy	±0.3 dB, 20 Hz–120 kHz harmonics.
Measurement Bandwidth	
LF –3 dB	<10, 22, 100, or 400 Hz.
HF –3 dB	22k, 30k, 80k, or >500 kHz (Option filter selection also affects bandwidth).
Residual THD+N[14]	
At 1 kHz	≤(0.00025% + 1.0 µV) [–112 dB], 22 kHz BW (valid only for analyzer inputs ≤8.5 Vrms).
20 Hz–20 kHz	≤(0.0003% + 1.0 µV) [–110.5 dB], 22 kHz BW, ≤(0.0005% + 2.0 µV) [–106 dB], 80 kHz BW, ≤(0.0010% + 5.0 µV) [–100 dB], 500 kHz BW .
10 Hz–100 kHz	≤(0.0040% + 5.0 µV) [–88 dB], 500 kHz BW .
Minimum Input	5 mV for specified accuracy, usable to <100 µV with fixed notch tuning.
Notch Tuning Modes	Counter Tuned, Sweep Track, AGen-Track (analog generator), DGen-Track (digital generator), or Fixed (set by direct entry).
Notch Tracking Range	±2.5% from fixed setting.

Crosstalk Function

Frequency Range	10 Hz–200 kHz.
Accuracy[14]	±0.4 dB, 20 Hz–120 kHz.
Residual Crosstalk[15]	
10 Hz–20 kHz	≤ –140 or 1 µV.
20 kHz–100 kHz	≤ –126 dB or 2.5 µV .

IMD Measurements

with option "IMD"

Option "IMD" adds the capability to measure intermodulation distortion (IMD) using three of the most popular techniques. The demodulated IMD signal can also be selected for FFT analysis in SYS-2712 and SYS-2722 configurations.

SMPTE (DIN) IMD Function

Test Signal Compatibility	Any combination of 40 Hz–250 Hz (LF) and 2 kHz–100 kHz (HF) tones, mixed in any ratio from 0:1 to 8:1 (LF:HF).
IMD Measured	Amplitude modulation products of the HF tone. –3 dB measurement bandwidth is typically 20 Hz–750 Hz.

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[14] *System specification measured with the 2700 series analog generator and the analog analyzer set to the indicated measurement bandwidth (BW). Generator amplitude setting must be ≤12 Vrms balanced or ≤6 Vrms unbalanced for specified system performance below 30 Hz. At higher amplitude settings generator THD derates to 0.0020% from 20 Hz–30 Hz.*

[15] *Uses the 1/3-octave bandpass filter to enhance the measured range in the presence of wideband noise. Alternate (interfering) channel input must be ≥ 5 mV.*

Measurement Range	0%–20%.
Accuracy	±0.5 dB.
Residual IMD[16]	≤0.0015% [–96.5 dB], 60/7 kHz or 250/8 kHz.

CCIF and DFD IMD Functions

Test Signal Compatibility	Any combination of equal amplitude tones from 4 kHz–100 kHz spaced 80 Hz–1 kHz.
IMD Measured	
CCIF	2nd order difference frequency product relative to the amplitude of either test tone.
DFD	u_2 (2nd order difference frequency product) per IEC 268-3 (1986).
Measurement Range	0%–20%.
Accuracy	±0.5 dB.
Residual IMD[15]	CCIF ≤0.0004% [–108 dB], 14 kHz + 15 kHz, DFD ≤0.0002% [–114 dB], 14 kHz + 15 kHz.

DIM (TIM) IMD Function

Test Signal Compatibility	2.96 kHz–3.15 kHz squarewave mixed with 14 kHz–15 kHz sine wave (probe tone).
IMD Measured[17]	u_4 and u_5 per IEC 268-3 (1986).
Measurement Range	0%–20%.
Accuracy	±0.7 dB.
Residual IMD15	≤0.0020% [–94 dB].

Wow & Flutter Measurements

with option "W&F"

Option "W&F" adds the capability to make both conventional wow & flutter and scrape flutter measurements (using the technique developed by Dale Manquen of Altair Electronics, Inc.). The demodulated W&F signal can also be selected for FFT analysis in SYS-2712 and SYS-2722 configurations.

Test Signal Compatibility	
Normal	2.80 kHz–3.35 kHz.
"High-band"	11.5 kHz–13.5 kHz.
Measurement Range	0%–1.2%.
Accuracy (4 Hz)	±(5% of reading + 0.0005%).
Detection Modes	IEC/DIN (quasi-peak per IEC-386), NAB (average), JIS (per JIS 5551).

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[16] *System specification measured with the 2700 series analog generator at any valid input level ≥ 200 mVrms.*

[17] *IEC 268-3 defines nine possible DIM products. The 2700 series IMD option analyzer is sensitive only to the u4 and u5 products using the simplified measurement technique proposed by Paul Skritek. DIM measurements using this technique will typically be 6 dB–8 dB lower (better) than the results obtained using FFT-based techniques which sum all nine products.*

Response Selections	
Weighted	4 Hz bandpass per IEC/DIN/NAB.
Unweighted	0.5 Hz–200 Hz.
Scrape[18]	200 Hz–5 kHz.
Wideband[17]	0.5 Hz–5 kHz.
Residual W+F	
Weighted	≤0.001%.
Unweighted	≤0.002%.
Scrape or Wideband	≤0.005%.
Minimum Input	5 mV, 20 mV for specified residual.
Settling Time	
IEC/DIN or NAB	Typically 3 seconds–6 seconds.
JIS	Typically 15 seconds–20 seconds.

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[18] *Operational only with high-band test signals (11.5 kHz–13.5 kHz). Upper –3 dB rolloff is typically 4.5 kHz using 12.5 kHz.*

Option Filters

Up to seven option filters can be installed in the analog analyzer for weighted noise or other special measurements. Only one option filter may be enabled at a time, and it is cascaded with the standard bandwidth limiting filters. The following tables list only the most popular types. Consult Audio Precision for custom designs.

Weighted Noise Measurement

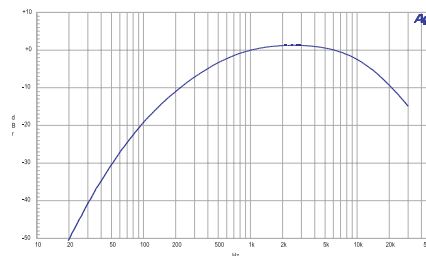|		
FIL-AWT	"A" weighting per IEC Rec 179.	*–see Figure 35*
FIL-CCR	Weighting per IEC468 (CCIR) and DIN 45404 (Also for CCIR/ARM).	*–see Figure 36*
FIL-CIT	Weighting per CCITT Rec P53.	*–see Figure 37*
FIL-CMS	"C-message" per BSTM 41004 and ANSI/IEEE Std 743-1984.	*–see Figure 38*
FIL-CWT	"C" weighting per IEC Rec 179.	*–see Figure 39*

Figure 35. FIL-AWT. ANSI-IEC "A" Weighting Filter.
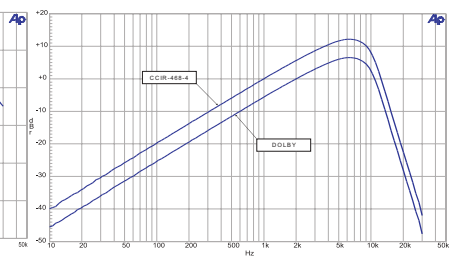

Figure 36. FIL-CCR. IEC468 (CCIR)/ DIN 45404 Noise Weighting Filter.
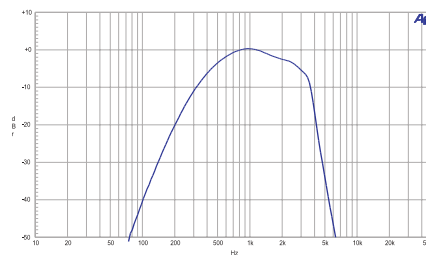
Figure 37. FIL-CIT. CCITT P53 Noise Weighting Filter.
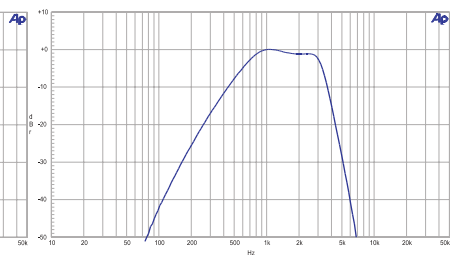
Figure 38. FIL-CMS.C-Message Weighting Filter (ANSI/IEEE 743-1984).
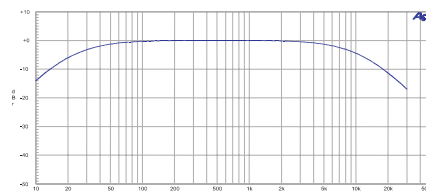
Figure 39. FIL-CWT. "C" Weighting (IEC-179).

Precision De-emphasis Family

FIL-D50	50 µs ±1%.	–see Figure 40
FIL-D50E	50 µs ±1% + 15.625 kHz notch.	
FIL-D50F	50 µs ±1% + 19.0 kHz notch.	–see Figure 41
FIL-D75	75 µs ±1%.	–see Figure 42
FIL-D75B	75 µs ±1% + 15.734 kHz notch.	–see Figure 43
FIL-D75F	75 µs ±1% + 19.0 kHz notch.	–see Figure 44
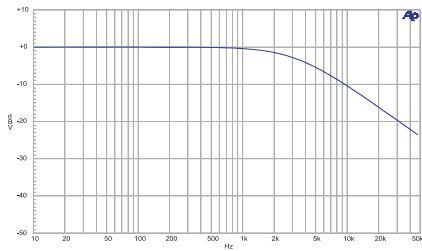
Figure 40. FIL-D50. 50 µs De-emphasis Filter.
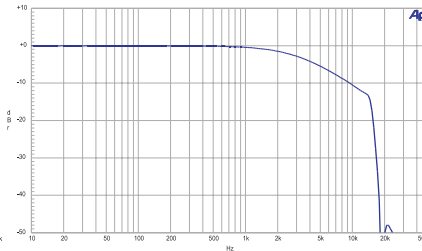
Figure 41. FIL-D50F. 50 µs with 19 kHz (FM) notch De-emphasis Filter.
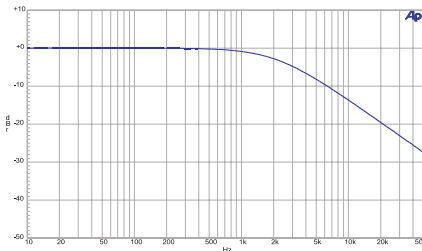
Figure 42. FIL-D75. 75 µs De-emphasis Filter.
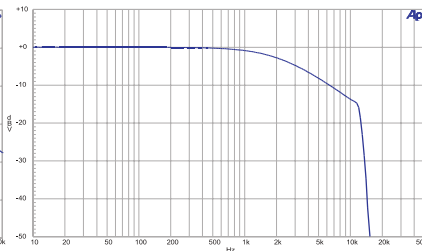
Figure 43. FIL-D75B. 75 µs with 15.734 kHz (NTSC) notch De-emphasis Filter.
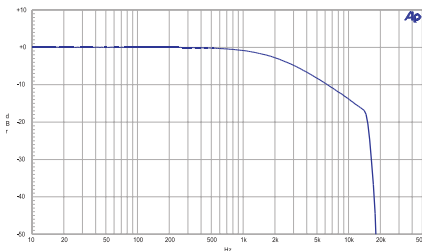
Figure 44. FIL-D75F. 75 µs with 19 kHz (FM) notch De-emphasis Filter.

Very Sharp Cutoff Low-Pass Filter Family

FLP-B20K	±0.1 dB, 10 Hz–20 kHz; >60 dB attenuation at 24 kHz and higher. Complies with AES17. *–see Figure 45*
FLP-B40K	±0.1 dB, 10 Hz–40 kHz; >60 dB attenuation at 48 kHz and higher. Complies with AES17.
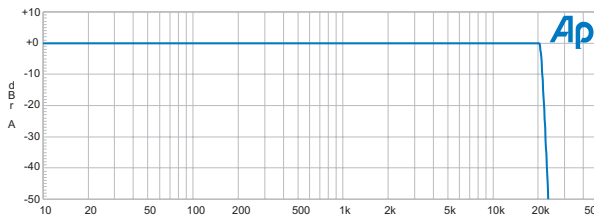
Figure 45. FLP-B20K "Brick Wall" 20 kHz low pass filter. Complies with requirements of AES17 for D/A converter THD+N measurements.

General Purpose Low-Pass

FLP-300	300 Hz ±3%, 5-pole.	
FLP-400	400 Hz ±3%, 5-pole.	
FLP-500	500 Hz ±3%, 5-pole.	
FLP-1K	1 kHz ±3%, 5-pole.	*–see Figure 46*
FLP-3K	3 kHz ±3%, 7-pole.	
FLP-4K	4 kHz ±3%, 7-pole.	
FLP-8K	8 kHz ±3%, 7-pole.	*–see Figure 47*
FLP-50K	50 kHz ±5%, 3-pole.	
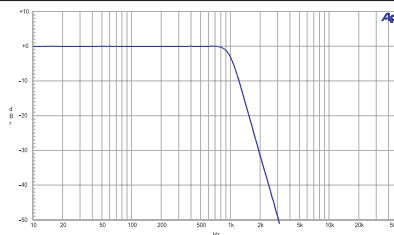
Figure 46. 1 kHz 5-pole Low Pass Filter.
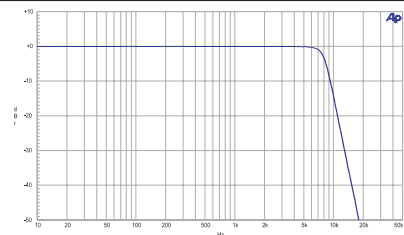
Figure 47. 8 kHz 7-pole Low Pass Filter.

General Purpose High-Pass

FHP-70	70 Hz ±3%, 8-pole.	
FHP-400	400 Hz ±3%, 9-pole.	*–see Figure 48*
FHP-2K	2 kHz ±3%, 9-pole.	
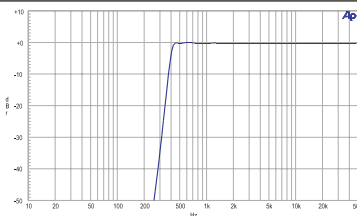
Figure 48. FHP-400. 400 Hz 9-pole High Pass Filter.

1/3-Octave (Class II) Bandpass Family

Family Response	Class II (4-pole) ± 0.2 dB from $0.97\, f_o$–$1.03\, f_o$; <-12 dB at $0.8\, f_o$ and $1.25\, f_o$; <-32 dB at $0.5\, f_o$ and $2.0\, f_o$. –see Figure 49
FBP-120	f_o = 120 Hz.
FBP-250	f_o = 250 Hz.
FBP-300	f_o = 300 Hz.
FBP-400	f_o = 400 Hz.
FBP-500	f_o = 500 Hz.
FBP-600	f_o = 600 Hz.
FBP-800	f_o = 800 Hz.
FBP-1000	f_o = 1.00 kHz.
FBP-1200	f_o = 1.20 kHz.
FBP-1500	f_o = 1.50 kHz.
FBP-2000	f_o = 2.00 kHz.
FBP-3000	f_o = 3.00 kHz.
FBP-4000	f_o = 4.00 kHz.
FBP-5000	f_o = 5.00 kHz.
FBP-6000	f_o = 6.00 kHz.
FBP-8000	f_o = 8.00 kHz.
FBP-10000	f_o = 10.0 kHz.
FBP-12500	f_o = 12.5 kHz.
FBP-15000	f_o = 15.0 kHz.
FBP-20000	f_o = 20.0 kHz.
FBP-30000	f_o = 30.0 kHz.

Figure 49. FBP-xxxx.
Normalized Response of 1/3-
Octave Band Pass Filters

Receiver Testing

FIL-RCR	200 Hz–15 kHz + 19.0 kHz notch.	*–see Figure 50*
FIL-IECR	20 Hz–15 kHz + 15.625 kHz notch	*–see Figure 51*
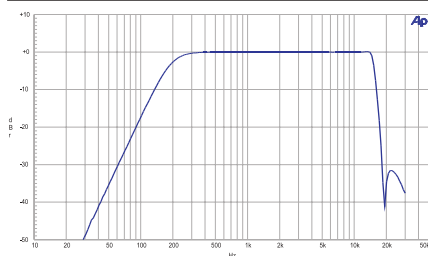
Figure 50. FIL-RCR. 200 Hz to 15 kHz with 19 kHz (FM) notch.
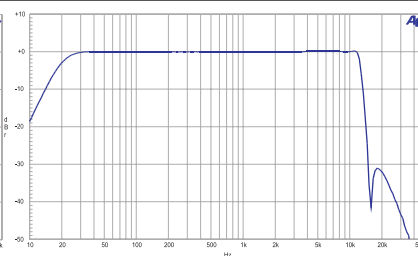
Figure 51. FIL-IECR. 20 Hz to 15 kHz with 15.625 kHz (PAL) notch.

Miscellaneous

FBP-500X	High-Q 500 Hz bandpass for CD DAC linearity measurements.	*–see Figure 52*
FIL-USR	Kit for building custom filters.	
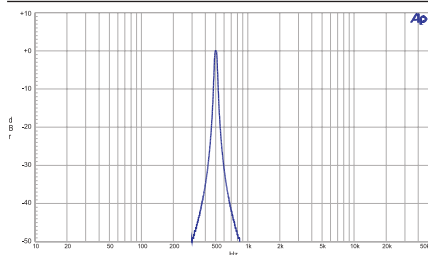
Figure 52. FBP-500X. High-Q 500 Hz Band Pass Filter (for CD linearity testing).

Option S-AES17

Option S-AES17 adds the capability to insert a 20 kHz or 40 kHz low-pass filter following the selected analog input preamplifier, but before any signal processing within the analog analyzer. It enables accurate noise and THD+N measurements of sigma-delta converters and switching power amplifiers that contain large amounts of unwanted energy above the normal audio bandwidth.

> *High performance sigma-delta converters and switching power amplifiers often contain out-of-band energy that can exceed the in-band audio signal. Standard bandwidth limiting and noise weighting filters will not give accurate measurements due to their relatively low roll-off rates.*

Option S-AES17 also includes the FLP-B20K and FLP-B40K option filters. These have been designed to work in tandem with the selectable pre-analyzer filters to provide THD+N measurements in accordance with AES17-1998.

Pre-Analyzer Filter Response

(also affects the LEVEL and FREQUENCY meters of the selected channel)

20 kHz	±0.10 dB, 10 Hz–20 kHz (typ –3 dB at 25 kHz, <–60 dB above 60 kHz).
40 kHz	±0.10 dB, 10 Hz–40 kHz (typ –3 dB at 50 kHz, <–60 dB above 120 kHz).
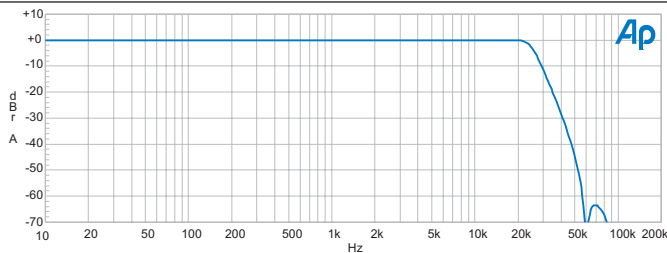
Figure 53. Typical response of 20 kHz "pre-analyzer" filter.

Residual THD+N (1 kHz)

"20k AES17" mode	≤(0.00030% + 1.0 µV) [–110.5 dB].
"40k AES17" mode	≤(0.00040% + 1.4 µV) [–108 dB].

DSP Analysis of Analog Signals

Available only in SYS-2712 and SYS-2722 configurations. Signals connected to the analog analyzer input connector may be routed through stereo A/D converters for enhanced analysis capabilities. There are two selectable converters. The high-resolution converter ("HiRes A/D") is optimized for signal analysis and FFT displays up to 30 kHz. It offers the best residual noise and distortion performance. The high bandwidth converter ("HiBW A/D") is optimized for signal analysis up to 120 kHz.

The term "f_s" refers to sample frequency (also called sample rate), in hertz.

High Resolution Converter

A/D Resolution	24-bit sigma-delta.
Sample Rate (f_s)	8 ks/s–108 ks/s variable; or 65.536 ks/s fixed.
Flatness (1 kHz ref)	±0.01 dB to 0.45× SR or 20 kHz, whichever is lower.
Alias Rejection[19]	typically >115 dB for signals >0.554 x f_s.
Distortion	−105 dB for f_s ≤65.536 ks/s, −102 dB for f_s up to 100 ks/s.

High Bandwidth Converter

A/D Resolution	16-bit sigma-delta.
Sample Rate (f_s)	16 ks/s–200 ks/s variable; or 131.072 ks/s, or 262.144 ks/s fixed.
Flatness (1 kHz ref)	±0.01 dB to 20 kHz, ±0.10 dB to 120 kHz (262.144 ks/s).
Alias Rejection[18]	typically >85 dB for signals > 0.540 x f_s.
Distortion	−92 dB for f_s ≤200 ks/s, −90 dB with f_s = 262.144 ks/s.

FFT Signal Analyzer

(With "FFT" DSP program)

Acquisition Length	800 samples to 4 M samples in 15 steps.
Transform Length	256–32768 samples in binary steps.
Processing	48 bit.
Amplitude Accuracy	±0.09 dB, 20 Hz–20 kHz, Flat-top or Move to Bin Center windows.
Averaging	1–4096 averages in binary steps. Averaging is power-based (frequency domain), or synchronous (time domain).

Waveform Display Modes
   Time Domain
   Frequency Domain
	
	Normal, Interpolate, Peak or Max.
	Peak pick (highest bin amplitude is displayed between the requested graph points).

Frequency Display Modes	Peak pick, individual bin.

---

[19] *Alias rejection is provided by digital filters within the A/D converters.*

Windows –see Figures 54 and 37.	Blackman-Harris (4-term with –92 dB sidelobes), Hann, Flat-top, Equiripple (AP design with –160 dB sidelobes), None, None, move to bin center, Hamming, Gaussian, Rife-Vincent 4-term, Rife-Vincent 5-term.

Move to bin center Window
    Frequency Range
 	
    Spurious Products
	$\pm4\%$ of input frequency, 7th FFT bin (low limit),
	    to 0.45 x f_s (high limit).
	
	<–120 dB.
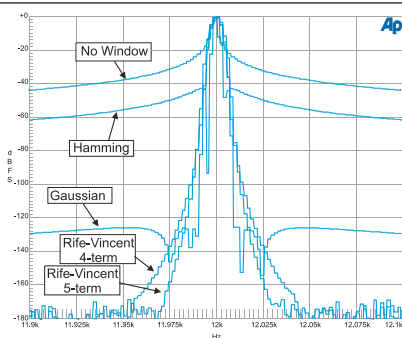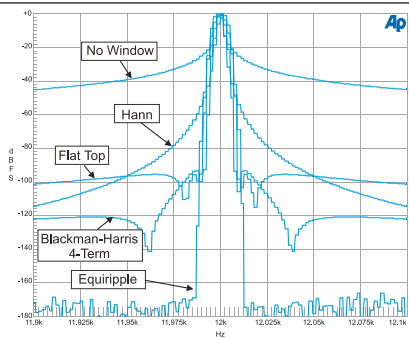
Figure 54. Windowing functions for FFT (A)

Figure 55. Windowing functions for FFT (B)

DSP Audio Analyzer

with "Analyzer" DSP program

Wideband Level/Amplitude

Accuracy (1 kHz)	±0.09 dB [$\pm1.0\%$]
Frequency Range	<10 Hz to 45% of Sample Rate [10 Hz–21.6 kHz at 48 ks/s].
High pass Filters	<10 Hz 4-pole, 22 Hz 4-pole, 100 Hz 4-pole, 400 Hz 4-pole (4-pole Butterworth or 10-pole elliptic if no other filters are enabled).
Low pass Filters	$F_S/2$ (maximum bandwidth), 20 kHz (6-pole elliptic), 15 kHz (6-pole elliptic).
Weighting Filters	ANSI-IEC "A" weighting, per IEC Rec 179, CCIR QPk per IEC468 (CCIR), CCIR RMS per AES17, C-message per IEEE Std 743-1978, CCITT per CCITT Rec. O.41, "F" weighting corresponding to 15 phon loudness contour, –see Figure 56 HI-2 Harmonic weighting.
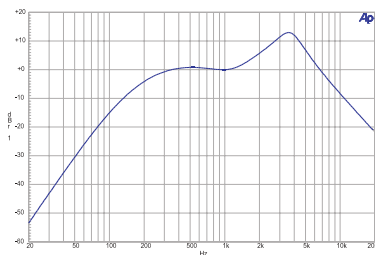
Figure 56. Digital Analyzer
F-weighting curve.

Narrow Band Amplitude

Frequency Range	<10 Hz to 47% of Sample Rate [10 Hz–22.56 kHz at 48 ks/s].
Filter Shape	10-pole, Q=19 (BW = 5.3% of f_o). *–see Figure 57*
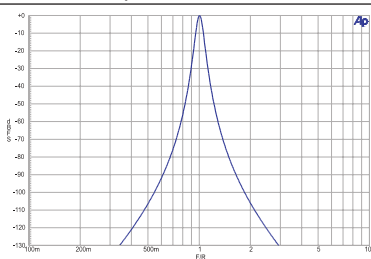
Figure 57. Digital Domain
Bandpass filter response.

THD+N Measurements

Frequency Range	<10 Hz to 47% of Sample Rate [10 Hz–22.56 kHz at 48 ks/s].
High pass Filters	<10 Hz (4-pole), 22 Hz (4-pole), 100 Hz (4-pole), 400 Hz (4-pole Butterworth).
Low pass Filters	F_S/2 (maximum bandwidth), 20 kHz (6-pole elliptic), 15 kHz (6-pole elliptic).
Weighting Filters	ANSI-IEC "A" weighting, per IEC Rec 179, CCIR QPk per IEC468 (CCIR), CCIR RMS per AES17, C-message per IEEE Std 743-1978, CCITT per CCITT Rec. O.41, "F" weighting corresponding to 15 phon loudness contour, *–see Figure 29* HI-2 Harmonic weighting.

Frequency Measurements

Range	<10 Hz to 47% of Sample Rate [10 Hz–23.0 kHz at 48 ks/s].
Accuracy	±0.01% of reading or 0.0001% of Sample Rate, whichever is greater.
Resolution	0.003% of reading or 0.0001% of Sample Rate, whichever is greater.

Phase Measurements

Measurement Ranges	±180, –90/+270, or 0/+360 degrees.
Accuracy[20]	
10 Hz–5 kHz	±0.5 degree.
5 kHz–20 kHz	±1 degree.
20 kHz–50 kHz	±2 degrees.
Resolution	0.01 degree.
Minimum Input	1 mV, both inputs.

SMPTE IMD Measurements

|	
Test Signal Compatibility	Any combination of 40 Hz–250 Hz (LF) and 2 kHz to 45% of Sample Rate (HF) tones, mixed in any ratio from 1:1 to 5:1 (LF:HF).
IMD Measured	Amplitude modulation products of the HF tone. –3dB measurement bandwidth is 10 Hz–750 Hz.
Measurement Range	0%–20%.
Accuracy	±0.5 dB.
Residual IMD[21]	≤0.0025%, 60 + 7 kHz or 250 + 8 kHz.

Quasi-Anechoic Acoustical Tester

With "MLS" DSP program

Signals	Four pink sequences, four white sequences.
Frequency Range	(Sample Rate ÷ 2000) to (Sample Rate ÷ 2).
Frequency Resolution (Max)	1.465 Hz at 48.0 ks/s.
Acquisition Length	32767 or 131071 samples.
FFT Length	32768.
Energy Time Windows	half Hann, Hann, <240 Hz to >8 kHz, <120 Hz to >16 kHz.
Time Windows (percent of data record to transition from 0 to full amplitude)	<5%, <10%, <20%, <30%.
Averaging	1–4096 averages in binary steps. Averaging algorithm is synchronous.

---

[20] *Both analog analyzer input channels must have same coupling (ac or dc) selection, and both DSP analyzer input channels must have same coupling (ac or dc) selection. Accuracy is valid for any input signal amplitude ratio up to ±30 dB. Upper frequency range limited to 45% of Sample Rate.*

[21] *System specification measured with the 2700 series analog generator. Valid for input levels ≥200 mVrms.*

Multitone Audio Analyzer

With "FASTTEST" DSP program

Acquisition Length	512–32768 samples in binary steps.
Transform Length	512–32768 samples in binary steps.
Processing	48 bit.
Measurements	Level vs frequency (Response), Total distortion vs frequency, Noise vs frequency, Phase vs frequency, Crosstalk vs frequency, Masking curve.
Frequency Resolution	(Sample Rate ÷ Transform Length) [1.465 Hz with f_s = 48 ks/s & Transform Length = 32768].
Frequency Correction Range	±3%.
Distortion	≤−115 dB.

Digital Signal Generator

Available only in the SYS-2720 and SYS-2722 configurations. The 2700 series digital generator consists of a DSP signal generator, hardware impairment generators and processing (for example, variable risetime and induced jitter), and digital output stages supporting the most popular formats.

Except for arbitrary and Pass Thru waveforms, the embedded output signals and the digitally generated analog output signals are independently selectable and concurrently available. If both digital and analog outputs are selecting arbitrary waveform, it must be the same one. If Pass Thru waveforms are selected for both analog and digital generators, they must be at the same sample rate.

Interface Signal Characteristics

Output Formats	Balanced XLR (AES/EBU per AES3-r1997), Dual Connector XLR, Unbalanced BNC (SPDIF-EIAJ per IEC-60958), Dual Connector BNC, Optical (Toslink®) per IEC-60958, General purpose parallel, or Serial interface to chip via optional PSIA-2722.
Sample Rate ("SR")	
Range	28 kHz–200 kHz for fully specified performance. Usable from 16 kHz–216 kHz.
Resolution	<0.0001 Hz.
Accuracy	±0.0002% [±2 PPM], lockable to external reference.
Output Amplitude	
Balanced (XLR)	
Range	0 Vpp–10.20 Vpp into 110 Ω. Output is unspecified above 8.00 Vpp if SR >108 kHz.
Resolution	40 mV, 4 mV below 1 Vpp.
Accuracy	±(10% + 80 mV).
Unbalanced (BNC)	
Range	0 Vpp–2.55 Vpp into 75 Ω. Output is unspecified above 2.00 Vpp if SR >108 kHz.
Resolution	10 mV, 1 mV below 0.25 Vpp.
Accuracy	±(8% + 20 mV).
Optical	Fixed using Toslink® TOTX142L.
Output impedance	
Balanced (XLR)	Nominally 110 Ω.
Unbalanced (BNC)	Nominally 75 Ω.
Channel Status Bits	Full implementation per IEC 60958, English language decoded, Professional or consumer or hex formats; independent in each channel.
User Bits	Set to 0.
Validity Flag	Selectable-set or cleared, common to both outputs.

Interface Signal Impairments

Variable Rise/Fall Time	12 ns–100 ns, ±(10% + 2 ns).
Cable Simulation	Approximates the signal degradation of 40 m–100 m of typical digital audio cable.
Induced Jitter (sine wave)	
Frequency Range[22]	2 Hz–200 kHz.
Amplitude Range[21]	0 UI–0.127 UI in 0.0005 UI steps, 0.13 UI–1.27 UI in 0.005 UI steps, 1.3 UI–12.7 UI in 0.05 UI steps.
Accuracy	±(10% + 2.0 ns), peak calibrated.
Flatness[23]	±1 dB, 100 Hz–50 kHz.
Residual Jitter[24]	≤600 ps (700 Hz–100 kHz analyzer bandwidth), ≤1.0 ns (50 Hz–100 kHz analyzer bandwidth).
Spurious Jitter Products	Typically 30 dB below jitter signal or 0.001 UI, whichever is larger.
Normal Mode Noise	
Waveform	Pseudo-random pulse train.
Balanced (XLR)	0 Vpp–2.55 Vpp in 10 mV steps, ±(10% + 100 mV).
Unbalanced (BNC)	0 mVpp–635 mVpp in 2.5 mV steps, ±(10% + 25 mV).
Common Mode Signal	Applies to Balanced (XLR) outputs only.
Frequency Range	20 Hz–100 kHz.
Amplitude Range	0 Vpp–20.4 Vpp in 80 mV steps, ±(10% + 160 mV).
Delay from rear panel Reference Output	–64 UI to +63.5 UI in 0.5 UI steps.

Embedded Signal Generation

Encoding is selectable 8–24 bit Linear, μ-Law, or A-Law

Sine Family Common Characteristics

Waveforms	Sine, Sine Burst (rectangular envelope), Variable Phase Sine (two sine waves of same frequency but settable phase), Stereo Sine (independent frequency and amplitude in each channel), Dual Sine (sum of two sine waves with variable ratio), Sine + Offset, and Shaped Sine Burst (raised cosine envelope).
Frequency Range	10 Hz to 47% of Sample Rate [22.56 kHz at 48 ks/s].
Frequency Resolution	Sample Rate ÷ 2^{23} [0.006 Hz at 48 ks/s].
Flatness	±0.001 dB.

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[22] *Combinations of jitter amplitude and frequency must not result in greater than 50% reduction in transmitted bit width.*

[23] *System specification at 32, 44.1, 48, 88.2, 96, 176.4, and 192 ks/s. Flatness of the jitter generator will exhibit degradation at other sample rates.*

[24] *Applies to electrical signal formats only. Specification subject to the following conditions: (1) all jitter generator impairments must be turned off or disabled including the delay from reference output feature; (2) the digital input signal must be ≥ 1.5 Vpp (AES/EBU) or ≥ 300 mVpp (SPDIF-EIAJ) and have a risetime ≤ 20 ns. Jitter via the optical path may be considerably worse with embedded audio or dither enabled (limited by Toslink® technology).*

Harmonics/Spurious Products	≤0.000001% [−160 dB].

Variable Phase Sine Wave

Phase Range	±180 deg.
Phase Resolution	0.01 deg.

Sine + Offset

Offset Amplitude	Sine amplitude + |offset amplitude| ≤100% F_S.

Sine Burst and Shaped Sine Burst

Envelope	Rectangular for Sine Burst, Raised cosine for Shaped Burst. −*see Figure 58*
Interval	2 cycles–65536 cycles.
Burst On	1 to (number of Interval cycles minus 1).
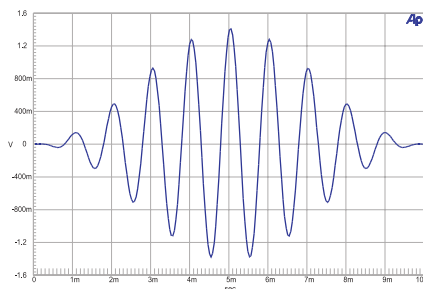
Figure 58. Shaped Sine Burst signal.
(1 kHz, 10 cycles)

Square Wave

Frequency Range	≤1 Hz to 1/6 Sample Rate. Frequencies are limited to even integer sub-multiples of the Sample Rate.
Even Harmonic Content	≤0.000001% [−160 dB].

SMPTE/DIN Waveform

Upper Tone Range	2 kHz to 47% of Sample Rate [22.56 kHz at 48 ks/s].
Lower Tone Range	40 Hz–500 Hz.
Amplitude Ratio	1:1 or 4:1 (LF:HF).
Distortion/Spurious	≤0.000001% [−160 dB] at 4:1 ratio.

CCIF and DFD IMD Waveforms

Center Frequency Range	3.00 kHz to (47% of Sample Rate –½ IM freq.).
IM Frequency Range	80 Hz–2.00 kHz.
Distortion/Spurious	≤0.000001% [–160 dB].

DIM IMD Waveform

Square/Sine Frequencies	Determined by Sample Rate (see Note below).
Distortion/Spurious	≤0.000001% [–160 dB].

The DIM test signal consists of a square wave and a sine wave mixed in a 4:1 amplitude ratio. Since digital square waves are generated by alternately turning the output on and off for the same number of sample periods, the frequencies achievable are limited to even sub-multiples of the Sample Rate. Because of this constraint, the square wave frequency is chosen first to be as close to the "ideal" analog test frequency as possible. The sine wave frequency is then chosen based upon the ideal sine/square frequency ratio. The following table lists some examples for the DIM and DIMB signals:

DIM: "ideal" square frequency = 3150, sine/square frequency ratio = 100/21		
Sample Rate	Square Wave Frequency	Sine Wave Frequency
44100	3150	15000
48000	3000	14285.7
DIMB: "ideal" square frequency = 2960, sine/square frequency ratio = 175/37		
44100	3150	14898.65
48000	3000	14189.19

Noise

Types	Pink, White, Burst, USASI.

Special Signals

Monotonicity	Low level staircase waveform for D/A linearity testing.
J-Test	Produces a maximum amount of data-induced jitter on low-bandwidth transmission links.
Polarity	Two sinewaves phased for reinforcement with normal polarity.
Walking Ones	A single binary one value "walked" from LSB to MSB.
Walking Zeros	A single binary zero value "walked" from LSB to MSB.
Constant Value (Digital dc)	32-bit resolution when using triangular dither.

Random (Bittest)	Pseudo-random binary states of all bits.
Pass Thru	Passes the signal from the rear panel Ref Input. Accepts sample rates from 27 kHz–200 kHz and outputs at programmed sample rate. Ratio of rates may not exceed 7.75:1.

Quasi-Anechoic Acoustical Tester (MLS)

(Also see MLS in Digital Analyzer section, page 84)

Signals	Four pink sequences, four white sequences.
Frequency Range	dc to Sample Rate ÷ 2.
Sequence Length	32767 samples or 131071 samples.

Arbitrary and Multitone Waveforms ("Arb Wfm")

Stored waveform consisting of multiple sine waves, each at independent frequency, amplitude, and phase

Signal	Determined by the associated file specified in the panel drop-down box.
Frequency Range	dc to Sample Rate ÷ 2.
Length	256 points–16384 points per channel. Utility is provided to prepare waveform from user specified frequency, amplitude, and phase data.
Frequency Resolution	Sample Rate ÷ Length [2.93 Hz at 48 ks/s for a waveform 16384 points in length].
Maximum Number of Tones	(Length / 2) –1 [8191 for Length = 16384].

Dither

(May be enabled for all waveforms except Monotonicity, J-Test, Walking Ones and Zeroes, and Random)

Probability Distribution	Triangular or rectangular; pseudo random, independent for each channel.
Spectral Distribution	Flat (white) or Shaped (+6 dB/oct).
Amplitude	8 bit–24 bit, or OFF.

Pre-Emphasis Filters

(all waveforms)

Filter Shape	50/15 µs or J17.
Response Accuracy	±0.02 dB, 10 Hz to 45% of Sample Rate.
Residual Distortion	≤0.00003% [−130 dB].

Digital Analyzer

Available only in the SYS-2720 and SYS-2722 configurations.

Digital Interface Signal Measurements

Input Sample Rate	
Range	28 kHz–200 kHz for fully specified performance. Usable from 16 kHz–216 kHz.
Accuracy	
Int. Reference	±(0.0003% + 1 digit) [±3 PPM].
Ext. Reference	±(0.0001% + 1 digit) [±1 PPM].
Input Amplitude	
Balanced (XLR)	0 Vpp–10.00 Vpp, ±(5% + 25 mV).
Unbalanced (BNC)	0 Vpp–2.5 Vpp, ±(5% + 6 mV).
Optical	Displays voltage of internal logic signal (not linearly related to optical input power).
Jitter Measurement	
Range[25]	0 UI–0.200 UI [0 ns–32 ns at 48k, 0 ns–8.0 ns at 192k, etc.].
Accuracy	±(10% + 1.0 ns), peak calibrated.
Detection	"Pk" (for measurements per AES-3), or "Avg" (for transfer function measurements).
Bandwidth	<50 Hz–100 kHz, 120 Hz–100 kHz, 700 Hz–100 kHz, or 1.2 kHz–100 kHz.
Flatness[26]	±1 dB, 100 Hz–50 kHz.
Residual Jitter[27]	≤600 ps with "700 Hz–100 kHz" bandwidth, ≤1.0 ns with "50 Hz–100 kHz" bandwidth.
Jitter Spectrum	Spurious products are typically 40 dB below jitter signal or <0.0003 UI [–70 dBUI], whichever is larger.
Common Mode Amplitude	0 Vpp–20.0 Vpp, ±(10% + 50 mV), 100 Hz–100 kHz.
Channel Status Bits	Full implementation, English language decoded (Professional or Consumer) hex formats, independent in each channel.
User Bits	Not displayed.
Validity Flag	Displayed for each channel.
Parity	Displayed for total signal (both channels combined).
Output to Input Delay	Measures propagation from the rear panel AES/EBU Reference Output to the input.
Range	–12.7 to +115.1 UI [–10% to +90% of frame] in seconds, 60 ns resolution.

[25] *The maximum measureable jitter increases to 1.000 UI for sine wave jitter signals below 1 kHz decreasing to 0.200 UI at or above 8 kHz.*

[26] *System specification at 32, 44.1, 48, 88.2, 96, 176.4, and 192 ks/s using the "50 Hz to 100 kHz" analyzer bandwidth selection. Flatness of the jitter generator will exhibit degradation at other sample rates.*

[27] *Applies to electrical signal formats only. Specification subject to the following conditions: (1) all jitter generator impairments must be turned off or disabled including the delay from reference output feature; (2) the digital input signal must be ≥ 1.5 Vpp (AES/EBU) or ≥ 300 mVpp (SPDIF-EIAJ) and have a risetime ≤ 20 ns. Jitter via the optical path may be considerably worse with embedded audio or dither enabled (limited by Toslink® technology).*

Signal Confidence	Displayed for total signal (both channels combined)
Receiver Lock	Displayed for total signal (both channels combined).
Coding Error	Displayed for total signal (both channels combined).

Digital Interface Analyzer

with "INTERVU" DSP program

The Digital Interface Analyzer (Intervu) operates in conjunction with an 8-bit A/D converter clocked at 80.0 MHz. The analyzer can display the interface signal in time or frequency domain, as an eye pattern, or probability graphs of amplitude or pulse width. The Digital Interface Analyzer can also demodulate the jitter signal and display it in time or frequency domain or as a histogram. The jitter signal or the data on the interface may be reproduced through the monitor loudspeaker.

AES/EBU Input Voltage	
Balanced	0 Vpp–10.00 Vpp, ±(10% + 50 mV).
Unbalanced	0 Vpp–2.5 Vpp, ±(8% + 12 mV).
Jitter Amplitude	0 UI–5 UI pk, ±(5% + 0.015 UI).
Residual Jitter	≤0.01 UI (50 Hz–1 MHz BW).
Spurious Jitter Products	≤0.001 UI, or ≤–60 dB below jitter signal.
Common Mode Amplitude	0 Vpp–20.00 Vpp, ±(30% + 50 mV), 20 kHz–1 MHz.
Jitter Probability Display	256 bins, auto-ranging.
Input Probability Display	256 bins, auto-ranging.
Bit Width Probability Display	32768 bins.
Risetime	Typically 17 ns–20 ns.
Acquisition time / memory	19.66 ms / 1,572,864 samples.

Embedded Audio Measurements

With "ANALYZER" DSP program

Wideband Level/Amplitude

Range	–120 dBFS to 0 dBFS (usable to –140 dBFS).
Frequency Range	10 Hz to 45.8% of Sample Rate, [10 Hz–20.2 kHz at 44.1 ks/s], [10 Hz–22.0 kHz at 48 ks/s], [10 Hz–44.0 kHz at 96 ks/s].
Accuracy	±0.01 dB.
Flatness	±0.01 dB, 15 Hz–22 kHz (<10 Hz high-pass filter selection).

High pass Filters	<10 Hz (4-pole), 22 Hz (4-pole), 100 Hz (4-pole), 400 Hz (4-pole Butterworth, or 10-pole elliptic if no other filters are enabled).
Low pass Filters	$f_S/2$ (maximum bandwidth), 20 kHz (6-pole elliptic), 15 kHz (6-pole elliptic).
Weighting Filters	ANSI-IEC "A" weighting, per IEC Rec 179, CCIR QPk per CCIR Rec. 468, CCIR RMS per AES17, C-message per IEEE Std 743-1978, CCITT per CCITT Rec. O.41, "F" weighting corresponding to 15 phon loudness contour, *–see Figure 56, page 72* HI-2 Harmonic weighting.
Residual Noise (at 48 ks/s and 96 ks/s f_s)	−141 dBFS unweighted, −144 dBFS A-weighted, −140 dBFS CCIR RMS, −130 dBFS CCIR QPk, −142 dBFS 20 kHz LP, −143 dBFS 15 kHz LP, −139 dBFS "F" weighting, −152 dBFS CCITT, −151 dBFS C Message.

Narrow Band Amplitude

Frequency Range	10 Hz to 40% of Sample Rate, [10 Hz–17.6 kHz at 44.1 ks/s], [10 Hz–19.2 kHz at 48 ks/s], [10 Hz–38.4 kHz at 96 ks/s].
Filter Shape	10-pole, Q=19 (BW = 5.3% of f_o). *–see Figure 57, page 72*
Residual Distortion	≤−150 dBFS.

THD+N Measurements

Frequency Range	<10 Hz to 47% of Sample Rate, [10 Hz–19.9 kHz at 44.1 ks/s], [10 Hz–21.6 kHz at 48 ks/s], [10 Hz–43.2 kHz at 96 ks/s].
Residual THD+N	≤−138 dBFS. *–see Figure 59, page 85*
High pass Filters	<10 Hz (4-pole), 22 Hz (4-pole), 100 Hz (4-pole), 400 Hz (4-pole Butterworth).
Low pass Filters	$F_S/2$ (maximum bandwidth), 20 kHz (6-pole elliptic), 15 kHz (6-pole elliptic).
Weighting Filters	Same as Wideband Level/Amplitude.
Residual Noise	Same as Wideband Level/Amplitude.

Frequency Measurements

Range	10 Hz to 47% of Sample Rate, [10 Hz–21.0 kHz at 44.1 ks/s], [10 Hz–23.0 kHz at 48 ks/s], [10 Hz–46.0 kHz at 96 ks/s].

Accuracy	±0.01% of reading or 0.0001% of Sample Rate, whichever is greater.
Resolution	0.003% of reading or 0.0001% of Sample Rate, whichever is greater.

Phase Measurements

Measurement Ranges	±180, –90/+270, or 0/+360 degrees.
Accuracy[28]	±0.05 degree, 10 Hz to 45% of Sample Rate.
Resolution	0.01 degree.
Minimum Input	–30 dBFS, both inputs.

SMPTE IMD Measurements

Test Signal Compatibility	Any combination of 40 Hz–250 Hz (LF) and (2 kHz to 45% of Sample Rate) (HF) tones, mixed in any ratio from 1:1 to 5:1 (LF:HF).
IMD Measured	Amplitude modulation products of the HF tone. (–3 dB measurement bandwidth is 10 Hz–750 Hz.)
Measurement Range	0%–20%.
Accuracy	±0.5 dB.
Residual IMD	≤–130 dB at 0 dBFS, 60 + 7 kHz or 250 + 8 kHz ≤–110 dB at –25 dBFS, 60 + 7 kHz or 250 + 8 kHz.

Embedded Audio, FFT Spectrum Analyzer

with "FFT" DSP program (48-bit processing)

Acquisition Length	800 samples–4 M samples in 15 steps.
Transform Length	256–32768 samples in binary steps.
Windows (see Figures 54 and 55, page 71)	Blackman-Harris (4-term with –92 dB sidelobes), Hann, Flat-top, Equiripple (AP design –160 dB sidelobes), None, None, move to bin center, Hamming, Gaussian, Rife-Vincent 4-term, Rife-Vincent 5-term.
Amplitude Accuracy	±0.001 dB, 20 Hz–20 kHz, with Flat-top window.
Phase Accuracy[29]	±0.05 degree, 10 Hz to 45% of Sample Rate.
Resolution	0.01 degree.
Averaging	1–4096 averages in binary steps. Averaging is power-based (frequency domain), or synchronous (time domain).

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[28] *Both DSP analyzer input channels must have the same coupling (ac or dc) selection. Accuracy is valid for any input signal amplitude ratio up to ±30 dB.*

[29] *Both DSP analyzer input channels must have same coupling (ac or dc) selection. Accuracy is valid for any input signal amplitude ratio up to ±30 dB*

Distortion Products	≤−160 dB.
Display Modes	
Time Domain	Normal, Interpolate, Peak or Max.
Frequency Domain	Peak pick, individual bin, Smoothing.
Move to bin center Window	
Frequency Range	7th bin to 45% of Sample Rate, and ±4% of input frequency[21.6 kHz at 48 ks/s].
Amplitude Accuracy	±0.025 dB.
Spurious Products	≤−120 dB.

Embedded Audio, Multitone Audio Analyzer

with "FASTTEST" DSP program (48 bit processing)

Acquisition Length	512–32768 samples in binary steps.
Transform Length	512–32768 samples in binary steps.
Measurements	Level vs frequency, Total distortion vs frequency, Noise vs frequency, Phase vs frequency, Crosstalk vs frequency, Masking curve.
Frequency Resolution	Sample Rate ÷ 2^{15} [1.465 Hz with 48 ks/s].
Frequency Correction Range	±3%.
Distortion	≤−140 dB.

Embedded Audio, Quasi-Anechoic Acoustical Tester

with "MLS" DSP program

| Signals	Four pink sequences and four white sequences, selected by triggering generator MLS setting.
Frequency Range	(Sample Rate ÷ 2000) to (Sample Rate ÷ 2).
Frequency Resolution (Max)	1.465 Hz at 48 ks/s.
Acquisition Length	32767 or 131071 samples, selected by triggering generator MLS setting.
FFT Length	32768.
Energy Time Windows	half Hann, Hann, <240 Hz >8 kHz, <120 Hz >16 kHz.
Time Windows (percent of data record to transition from 0 to full amplitude)	<5%, <10%, <20%, <30%.
Averaging	1–4096 averages in binary steps, synchronous.

Graphs of Typical Digital Domain Performance
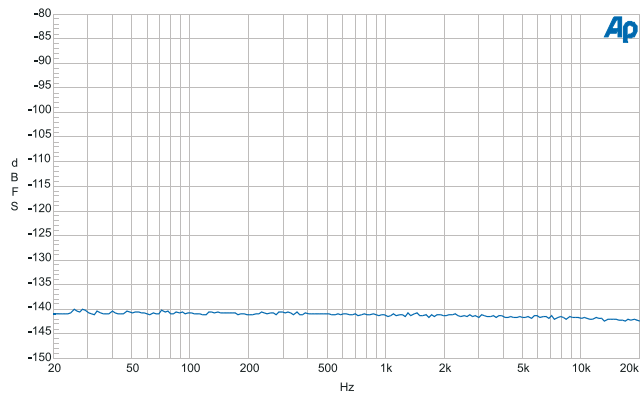
Figure 59. Typical Digital Domain system residual THD+N showing components below –140 dB.
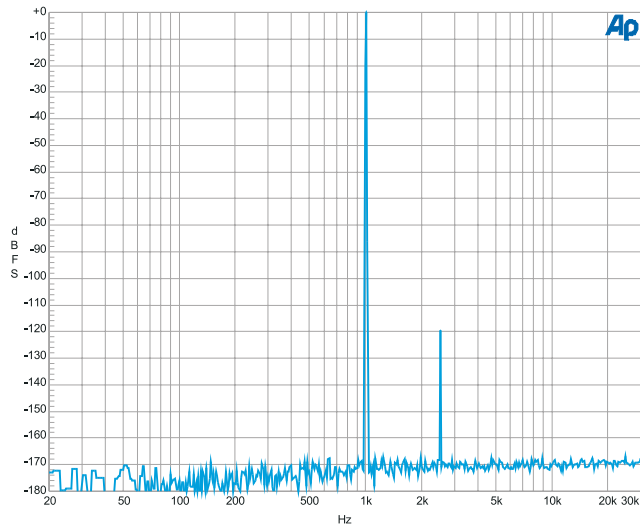

Figure 60. Illustration of typical Digital Domain FFT dynamic range. Signal is 0 dB 1 kHz with a secondary signal at –120 dB and 2.5 kHz.

Front Panel Auxiliary Signals

Figure 61. Monitors panel.

Generator Monitors

(all units except SYS-2720.)

Channel A	Buffered version of the channel A analog generator signal. Amplitude is typically 2.8 Vpp.
Channel B	Buffered version of the channel B analog generator signal. Amplitude is typically 2.8 Vpp.

Generator Auxiliary Signals

(all units except SYS-2720.)

Sync Output	LSTTL compatible signal that is intended to be used as a trigger for stable oscilloscope displays.
Trig/Gate Input	LSTTL compatible input, functional with option "BUR" only.

Analyzer Signal Monitors

(all units except SYS-2720.)

Channel A	Buffered version of the channel A analog input signal. Amplitude is typically 0 Vpp–3.6 Vpp.
Channel B	Buffered version of the channel B analog input signal. Amplitude is typically 0 Vpp–3.6 Vpp.
Reading	Buffered version of the analog analyzer output signal after all filtering and gain stages. Amplitude is typically 0 Vpp–3.6 Vpp.

Digital Signal Monitors

(SYS-2720 & SYS-2722 only.)

Via four 24-bit D/A converters. Function monitored depends upon analyzer program loaded; for example, noise and distortion products after the notch filter are monitored with "DSP Audio Analyzer" in its THD+N function.

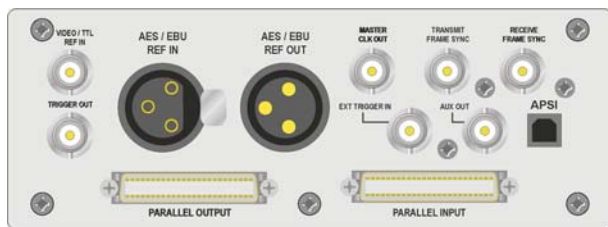|	
Channel 1	Buffered version of the digital channel 1 signal.
Channel 2	Buffered version of the digital channel 2 signal.
Reading 1	Distortion of the digital channel 1 signal.
Reading 2	Distortion of the digital channel 2 signal.

Rear Panel Auxiliary Signals

Figure 62. Miscellaneous digital I/O Panel (rear of SYS-2720 and SYS-2722 instruments only).

Reference Input ("REF IN") Characteristics

The reference input provides the ability to synchronize the internal sample rate generator to an external signal. The sample rate need not be at 1:1 ratio to the reference, but will be locked to the reference over the fully specified range of sample rate and Reference inputs. PLL bandwidth is approximately 5 Hz.

Input formats	28 kHz–200 kHz AES/EBU, NTSC, PAL, or SECAM video, or 8 kHz–10 MHz square wave.
Sample Rate Resolution	
8 kHz–65 kHz	1/128 Hz [0.0078125 Hz].
65 kHz–256 kHz	1/32 Hz [0.03125 Hz].
256 kHz–1 MHz	1/8 Hz [0.125 Hz].
1 MHz–4 MHz	½ Hz [0.5 Hz].
4 MHz–10 MHz	2 Hz.
Input Amplitude	200 mVpp–5.0 Vpp.
Input Impedance	Selectable terminated or high impedance.
AES/EBU (XLR)	Nominally 110Ω or 5kΩ.
Video/TTL (BNC)	Nominally 75Ω or 5kΩ.
Lock Range	±0.0015% [±15 PPM].
Input Delay from Reference Display	–12.7 to +115.1 UI [–10 to +90% of frame] in seconds, ±60 ns, AES/EBU signals only.
Reference Rate Display	Measures approximate reference input rate.

Reference Output ("REF OUT") Characteristics

Intended to drive devices under test that require their own reference input. The reference output signal is not jittered.

Output format	AES/EBU (per AES3-r1997).
Output Sample Rates	8 kHz–216 kHz, locked to front panel output.
Status Bits	Format "Professional," Type "Grade 2 reference," Origin "SYS2," Reliability flags implemented, CRCC implemented, Time of Day not implemented, Sample Count not implemented.

Output Delay from Reference Output	–64 to +63.5 UI in 0.5 UI steps, ±(5% + 0.5 UI).
Residual Jitter	≤600 ps (700 Hz–100 kHz BW).

Miscellaneous Digital I/O

Transmit Frame Sync	Square wave at the programmed internal sample rate (SR).
Receive Frame Sync	Square wave at the frame rate of the received AES/EBU signal.
Master Clock Out	Square wave at 256 x System Sample Rate (SSR). Selectable between jittered and un-jittered signals.
Auxiliary Input	LSTTL compatible trigger input for DSP program data acquisition.
Auxiliary Output	HCMOS signal, function under DSP program control.
Trigger Output	HCMOS signal, coincident with period of generated signal waveform.
Parallel Input and Output	For use with optional PSIA-2722 accessory.

Audio Monitor

All configurations contain an internal loudspeaker and headphone jack for listening to the generator, analyzer, or digital signal monitor points, including noise and distortion following analog or digital notch filters or the AES/EBU jitter signal. Use of the audio monitor does not preclude the use of any measurements.

Power Output	Typically 1 Watt.

General/Environmental

Power Requirements	100/120/230/240 Vac (−10%/+6%), 50/60 Hz, 240 VA max.
Temperature Range	
Operating	+5°C to +40°C.
Storage	−40°C to +75°C.
Humidity	90% RH (non-condensing).
Altitude	2000 m.
EMC[30]	Complies with 89/336/EEC, CISPR 22 (class B), and FCC 15 subpart J (class B).
Dimensions	
Width	41.9 cm [16.5 inches].
Height	14.6 cm [5.75 inches] bench-top (feet attached) 3U [5.25 inches] rack-mount.
Depth	34.5 cm [13.6 inches].
Weight	Approximately 15.4 kg [34 lbs].
Safety	Complies with 73/23/EEC, 93/68/EEC, and EN61010-1 (1990) + Amendment 1 (1992) + Amendment 2 (1995). Installation Category II, Pollution Degree 2.

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[30] *Emission and immunity levels are influenced by the shielding performance of the connecting cables. The shielding performance will depend on the internal design of the cable, connector quality, and the assembly methods used. EMC compliance was demonstrated using Audio Precision pn 4155.0117 cables for all ports using XLR-type connectors, Audio Precision pn 4155.0301 cable for the Master Clk port, and Pasternack Model No. PE3067-36 for all remaining ports using BNC-type connectors.*

Chapter 7

GPIB Configuration

Introduction

As an option, 2700 series instruments can be ordered in GPIB configurations, models SYS-27xxG. GPIB capabilities can also be added to instruments after purchase with the installation of a GPIB upgrade at the Audio Precision factory or at a factory-authorized representative.

The GPIB configuration brings the IEEE-488.2 interface standard, also known as the General Purpose Interface Bus or GPIB, to the 2700 series. Hardware for GPIB consists of a circuit board mounted on a connector panel assembly that is fitted in the reserved space on the instrument rear panel, as shown below.
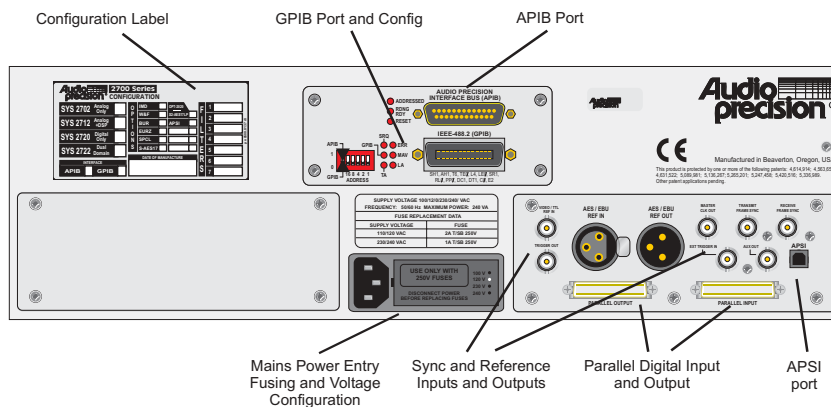

Figure 63. Rear panel, 2700 series instrument with GPIB Option.

APIB or GPIB?

A 2700 series instrument is normally controlled by Audio Precision's AP2700 control software using the APIB interface bus. In the GPIB configuration, the instrument can be controlled by either GPIB or APIB as selected by a switch on the GPIB hardware mounted on the instrument rear panel.

The GPIB configuration makes it possible to integrate 2700 series hardware into a system that may include other GPIB-compatible instruments and devices under operated by your choice of GPIB control software.

Audio Precision does not provide control software for GPIB applications. We do offer full documentation of the 2700 series functions controllable by GPIB in the *2700 Series GPIB Programmer's Reference Manual*, which contains a listing of all 2700 series GPIB commands and detailed explanations of command parameters. The reference manual comes with a companion CD-ROM that contains sample programs written for Microsoft Visual Basic. These programs provide examples of how to control a 2700 series instrument to perform many common audio measurements.

With this information and some knowledge of 2700 series theory and operation, you can develop custom software to control your GPIB system. A recommended path of software development is discussed in the following topics.

The GPIB Software Development Process

A practical working knowledge of your instrument is required in order to develop effective GPIB software for it. Figure 64 illustrates a typical software
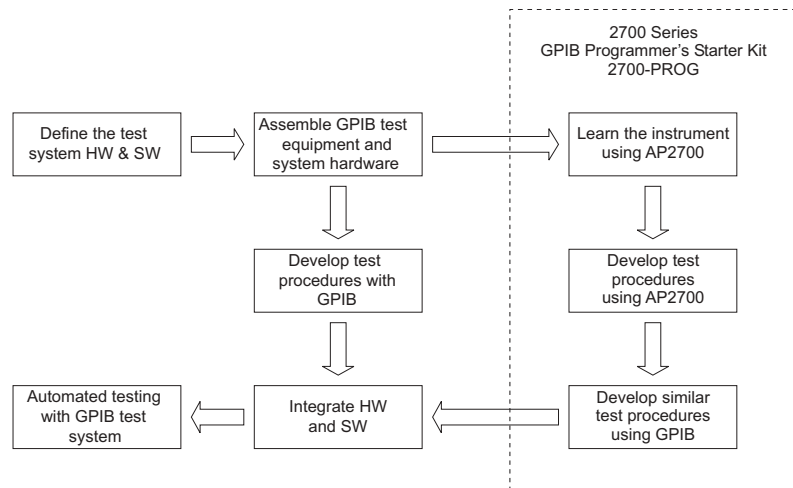

Figure 64. Recommended GPIB software development process using 2700 series control software.

development scenario in which the 2700 series control software is used to develop expertise with the 2700 series instrument before attempting to integrate the instrument into a larger system controlled by a GPIB interface software program.

Since the GPIB programming commands for the 2700 series are modeled directly on the 2700 series control software user interface, the fastest path for GPIB software development involves the use of the control software as a learning tool.
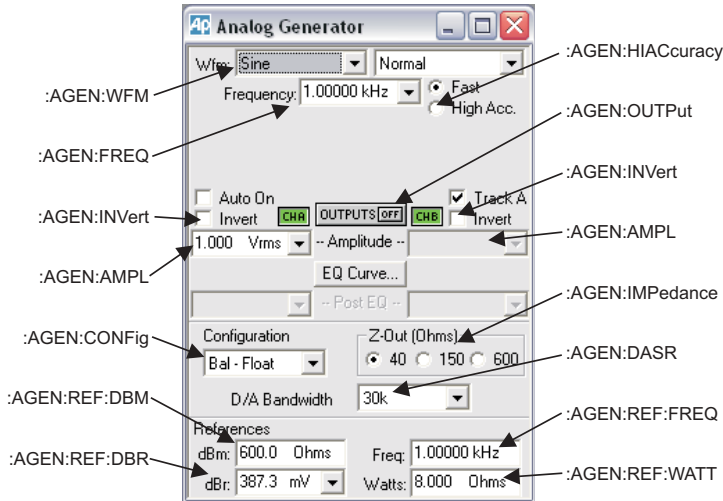

Figure 65. 2700 series Analog Generator panel with GPIB command call-outs.

Working in the 2700 series control software you can develop test methods for your DUT and build confidence that your methods are correct. You can use AP Basic and the Learn Mode button on the tool bar to develop AP Basic macros that perform the tests you wish to develop for GPIB. As you learn how your instrument best operates under AP2700control, the tasks involved in developing equivalent code for the GPIB port will be greatly simplified.

While developing tests in AP2700, refer to the *2700 Series GPIB Programmer's Reference Manual* for the equivalent GPIB commands. It contains detailed information on every GPIB command for the 2700 series, along with general programming information, code examples and reference material.

The command description sections of the *2700 Series GPIB Programmer's Reference Manual* show AP2700 software control panels for each of the major features of the 2700 series. Each panel is illustrated with GPIB command callouts to help you convert your AP2700 software setup to GPIB commands. Figure 65 illustrates this for the 2700 series Analog Generator panel.

Using both GPIB and APIB for Software Development

Every 2700 series instrument has an APIB port on the rear panel. The GPIB option adds a GPIB port and configuration DIP switches and indicators. The rear panel switches provide a means to select which port is to be used to control the instrument, and to assign the instrument's GPIB address. During the software development process you may switch between the two control modes in order to take advantage of the interactive development environment of the 2700 series control software.

The two ports cannot control the instrument simultaneously. You must select either GPIB or APIB as the control port. When the GPIB port is in control, the instrument's APIB port must be disconnected from the APIB interface card in the computer.
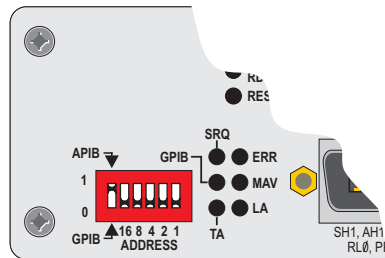

Figure 66. 2700 series GPIB/APIB interface panel, detail; shown in APIB mode.

To use APIB to control your instrument, select APIB on the rear-panel switch as shown in Figure 66 (left-most switch clicked up to "1"), and con-
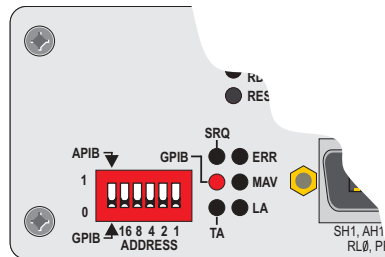

Figure 67. 2700 series GPIB/APIB interface panel, detail; shown in GPIB mode.

nect a cable from the APIB interface card on the controlling computer to the instrument's APIB port.

To use GPIB to control your instrument, connect an appropriate cable from the GPIB interface card on the controlling PC to the instrument's GPIB port, and select **GPIB** on the rear-panel switch as shown in Figure 67 (left-most DIP switch clicked down to "0"). An LED will light to indicate that GPIB is selected.
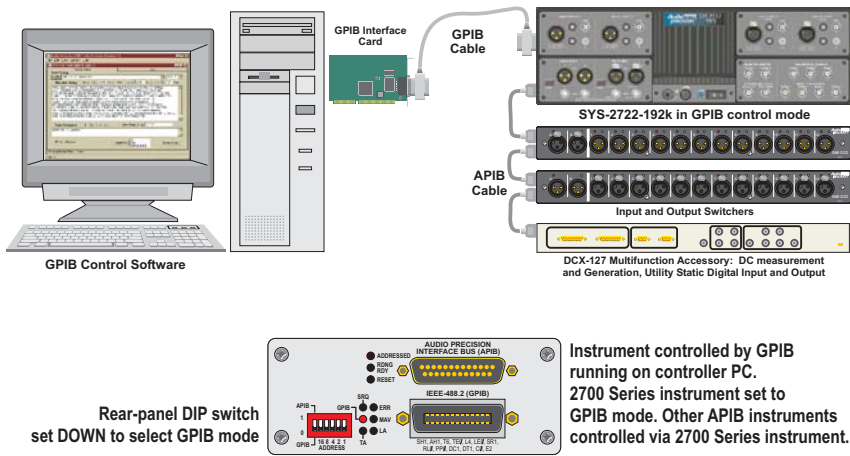
Figure 68. Control bus connections to 2700 series instrument with GPIB option in GPIB control mode. Switcher and DCX-127 connected to the instrument APIB port.
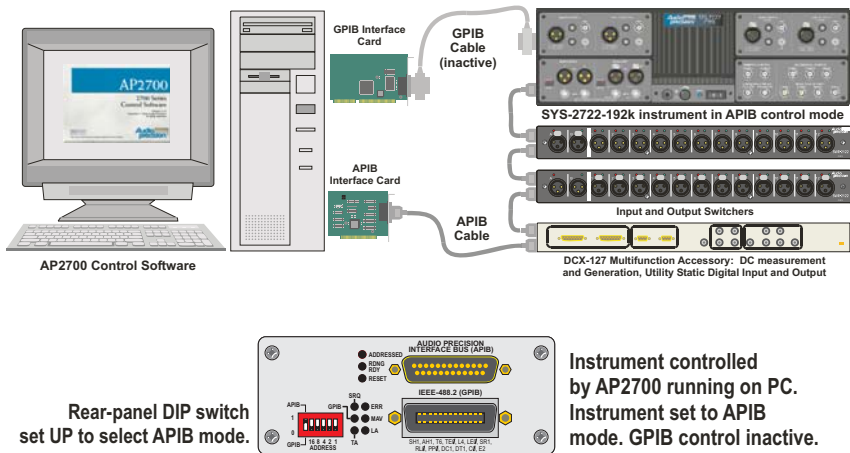

Figure 69. Control bus connections to 2700 series instrument with GPIB option in APIB control mode. PC APIB cable connected first to switcher and DCX-127, then connected to the instrument APIB port.

☞ **The instrument firmware EEROM settings for stored calibration constants can be corrupted by running in GPIB mode while connected to a PC on the APIB connector. This may result in an inoperable instrument.**

Always disconnect the PC from the APIB bus *before* switching to GPIB mode.

Always connect the PC to the APIB bus *after* switching to APIB mode.

Whether controlled by APIB or GPIB, a 2700 series instrument still uses the APIB bus to control other Audio Precision APIB instruments and accessories, such as the DCX-127 or the SWR-2122 family of switchers. Figure 68 illustrates how to connect these devices to a 2700 series instrument when the GPIB port is the control port; Figure 69 illustrates how to connect an APIB controller to this system when the GPIB port is disabled.

Establishing GPIB Communication

GPIB Connection

A GPIB-equipped 2700 series instrument has a 24-pin GPIB-compatible connector on the rear panel. This D-shell connector conforms to the mechanical requirements of IEEE-488.1-1987. The instrument is to be connected to the instrument controller via an appropriate cable. The instrument controller (a personal computer) must have a corresponding GPIB interface port. Standard
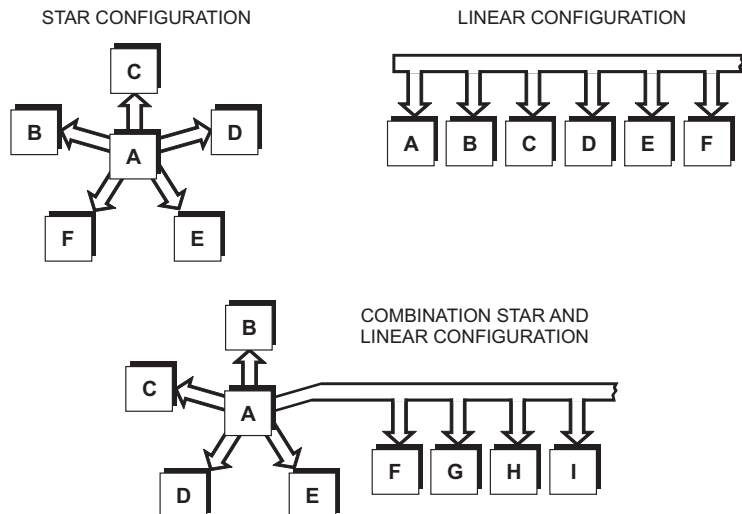

Figure 70. GPIB devices may be connected in star, linear, or combination configurations.

GPIB cables are designed so that they can be stacked if needed to connect multiple instruments into your GPIB system.

When connecting instruments into a GPIB system, observe the following rules:

- Connect and disconnect instruments from the bus only when the mains power to all instruments in the system is **off**.

- Assign a unique GPIB address to each instrument (device) on the bus.

- Devices may be connected in a star or linear configuration (see Figure 70), or a combination of star and linear configurations.

- Do not attach more than 15 devices (including the controller) to one bus.

- Cable length between devices must not exceed 2 meters (6 feet).

- One device must be attached to the bus for every two meters (6 feet) of cable.

- Total cable length must not exceed 20 meters (66 feet).

- At least two-thirds of the devices on the bus must be powered up for proper system operation.

GPIB Address and Control Mode Switch

Like all devices on a GPIB bus, the 2700 series instrument must be set to a unique GPIB address.

An address select switch bank with six slide switches sets the GPIB address and the control mode. The switch is shown below in Figure 71. The five switches to the right of the bank set the 5-bit binary primary address of the instrument. Legal addresses are 0 through 30. Set each switch up for a binary 1 or down for a binary 0.

The left-most switch in the bank sets the control mode, independent of the GPIB address switch settings. Click the switch to 0 (down) to set the control mode to GPIB.

> *The APIB port must not be connected to an Audio Precision APIB interface card when the GPIB mode is selected. Review the cautions in the previous section.*

Set the switch to 1 (up) to set the control mode to APIB. The APIB mode disables the GPIB interface (the pins are all set to the high-impedance state) and enables APIB to control the instrument.
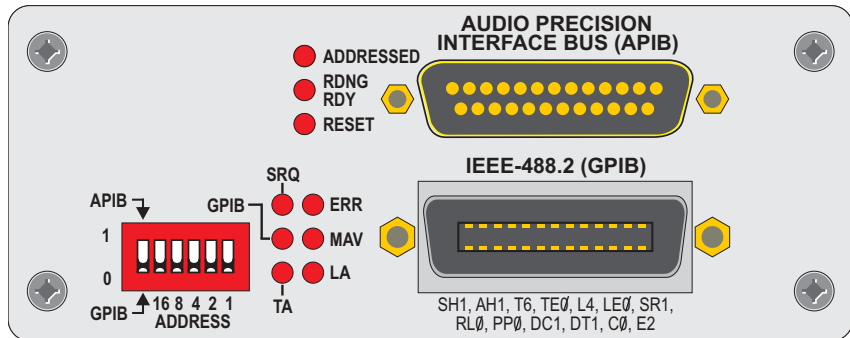
Figure 71. 2700 series GPIB/APIB interface panel, showing address DIP switch, GPIB status LEDs, APIB and GPIB ports.

GPIB Status LEDs

The six GPIB Status LEDs shown in Figure 71 indicate the current status of the GPIB bus. When an LED is illuminated, the status indicated is as follows:

- **SRQ** (SRQ interrupt)
  The instrument has asserted the SRQ line in order to request service from the GPIB controller.

- **GPIB**
  The instrument is under the control of the GPIB interface, set by the control mode switch. The APIB connector will be driven internally by the instrument GPIB interface board.

- **TA** (Talk Addressed)
  The instrument is talk addressed by the GPIB controller.

- **LA** (Listen Addressed)
  The instrument is listen addressed by the GPIB controller.

- **MAV**
  The instrument has one or more bytes in the output queue and has asserted the MAV (Message AVailable) bit in the Status Byte Register.

- **ERR**
  The instrument has detected an error condition that has not been reported to the system controller. This LED goes off when the error condition has been reported in response to the :ERRM? or :ERRS? error queries.

GPIB Program Message Terminators

When the instrument is addressed as a listener, two program message terminators (PMT) are supported:

- EOI line asserted with the NL character (ASCII 10, the linefeed character);

■ EOI line asserted with the last byte of a message.

The NL character alone without EOI is not supported.

When the instrument is addressed as a talker, one PMT is supported:

EOI line asserted with the NL character (ASCII 10, the linefeed character).