DSP USERS MANUAL

AUDIO PRECISION SYSTEM ONE

October, 1992 Release
Software Version 2.10a
DSP Program Version 12 for all Programs Except
FASTTRIG.DSP Version 1

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TECHNICAL BULLETIN, DSP v2.10A

This bulletin describes the differences between version 2.00 and version 2.10A of the DSP programs and DSP interactions with v2.10A of S1.EXE. This bulletin also contains several detailed notes on the DSP programs which are not yet included in the DSP User's Manual. References for more detail are made to specific sections of the DSP User's Manual, September 1991.

Manual Improvements

Extensive improvements and modifications have been made to the DSP User's Manual.

Specific recommendations on loading distribution software onto the hard disk of your computer have been added to the "DSP Introduction" Chapter A, beginning on the first page of the chapter.

A "Furnished Files" section has been added at the end of each DSP-program-specific chapter, listing the names and describing the intended function of the diskette files furnished with the revised software.

Reproductions of the DSP HELP screens for each program are now included in each program-specific chapter.

A new "Arbitrary Waveform Generator" section, plus several following sections describing applications of this feature, has been added to the FFT chapter (B), starting on page B-19.

Entirely new chapters have been added describing operation of the FASTEST program (Chapter C) and the MLS program (Chapter D).

A functional block diagram of the GENANLR program analysis sections has been added as Figure F-4 on page F-3.

General DSP-Related Changes

Shaped Dither Choice Added

In addition to the TRIANGULAR and RECTANGULAR dither choices previously available for digitally-generated outputs, a new SHAPED choice has been added. SHAPED dither has a triangular probability distribution but with high-frequency boost and low-frequency cut. The resulting degradation of signal-to-noise ratio is less than with flat (white) dither for human listeners and most weighted measurements. Amplitude increases are all at frequencies above one-fourth the sampling rate (above 12 kHz with the 48 kHz sample rate), with amplitude decreases below this frequency. See the "Dither" section on page A-13 for more information.

Dither OFF

Dither may now be turned on and off independently of the word size, by use of the UTIL SERIAL-DSP DITHER OFF command. Previously, dither could only be turned off by selecting 25 in the SIZE field.

SIZE Field Sets Word Length

The SIZE field (near the bottom of the DSP panel) now controls rounding of the 24-bit internally-generated signal to the appropriate word length for the device being driven. The SIZE field also controls dither amplitude when dither is on (TRIANGULAR, RECTANGULAR, or SHAPED). Previously, truncation always took place when a less-than-24-bit device was driven from the 24-bit output of System One Dual Domain. Now, the DSP rounds to the word size specified in this field even when dither is OFF. See the "Size Field" section beginning on page A-11 for more information.

DSP-Specific Error Messages Added

A number of DSP-specific error messages have been added. Error messages are displayed when a test is attempted with a SOURCE-1 frequency value beyond the Nyquist frequency for the sample rate in use. An error is indicated if an attempt is made to display SOURCE-1 time beyond the end of the acquisition buffer. If an attempt is made to sweep a parameter of one of the real-time programs (GENANLR, HARMONIC, BITTEST) beyond its allowable frequency range, the sweep is terminated and an error message is displayed.

External DSP Sweeps Added

The SOURCE-1 EXTERNAL list of choices has been expanded from the previous FREQUENCY, LEVEL, and TIME to include all measured parameters of the real-time DSP programs (GENANLR, HARMONIC, BITTEST). Thus, the digital output signal from a digital recorder or CD player may be used to drive the horizontal axis of frequency response and linearity graphs.

DSP Generator Turns OFF at Quit

The digital generator in FFTGEN.DSP, GENANLR.DSP, FASTEST.DSP, and MLS.DSP turns off when the QUIT command is selected in S1.EXE.

Waveform Files Remain v2.00

Substantial changes were made to S1.EXE and to all DSP programs, and their version numbers were all rolled to v2.10. The .WAV file format has not been changed and thus remains v2.00.

Relative Frequency Units Added

Relative frequency units (delta%, delta Hz, OCTAVES, DECADES, etc.) have been added to the FREQ unit list of FFTGEN, FFTSLIDE, GENANLR, and the filter frequency field of HARMONIC. These units all use the value in the ANALYZER Ref FREQ field as the reference for these units.

Note on All DSP Programs

It is recommended that the DOS APPEND command be added to your AUTOEXEC.BAT file to specify the directory or directories where your DSP program files are located. Then, a DSP program will properly load with a test and properly set the test panels regardless of which directory is currently in use. See the DOS APPEND Command section of Chapter 5, "Loading the Software", in the System One User's Manual for more details on this command.

When a test is loaded with a DSP program name "attached", S1.EXE software expects to find that .DSP program in the directory specified at the time the Names Program command was originally used to attach the test. Typically, the DSP program was in the same directory as the test file and S1.EXE thus expects to find the DSP program in the current directory. If the DSP program is not in the expected directory and not in a directory specified by the DOS APPEND command, an error message will be displayed ***and all DSP panel fields will default to undesired settings***. At this stage with the test already loaded, it is too late to retrieve the proper DSP panel field settings by using Names Program to attach the DSP program, wherever it may be located. Even if the DSP program file is now located, copied into the current directory, and the Names Program command used to attach it to the test, the original DSP panel field settings have still been lost. To retrieve the proper DSP panel field settings, re-load the test from disk ***after the DSP program has been copied into the current directory (or, better, after putting the DOS APPEND command into your AUTOEXEC.BAT file and re-booting your computer)***. Then, S1.EXE will find the DSP program and will be able to set each field of the DSP panel as the creator of the test intended.

Changes Affecting FFT Programs

Flat Top Window Selection Added

A new flat top window selection (FLAT) has been added to FFTGEN.DSP and FFTSLIDE.DSP. The FLAT window has essentially zero amplitude error across the bin, so correct amplitude readings are made regardless of where the signal falls within the bin. The FLAT window has poorer "skirt" selectivity than either the HANN or BH4 window choices. Note that in v2.00 FFT programs, the FLAT selection meant no window. No window is now selected by the NONE choice. See page B-5 for a more complete discussion of the flat window, and Figure B-2 in the same section for illustrations of the shape of all windows.

<F6> Sets Graph Time and Date

In addition to <F9> setting the time and date display at the upper right of the graph display (or in the top line of tabular displays), the <F6> and <Alt><F6> keystrokes now also set the time and date.

Auto Trigger Added, "1" and "2" Trigger Thresholds Changed in FFT Programs

A new software auto trigger function has been added to FFTGEN.DSP and FFTSLIDE.DSP, in addition to the previously-available "1" and "2" selections. "1" and "2" refer to DSP input channels CH-1 and CH-2. In v2.00 FFT programs, the "1" or "2" selection would cause triggering upon any non-zero signal on the selected channel. The new AUTO selection will trigger on the positive slope from whichever channel has the higher amplitude, and will trigger on anything other than an "infinity zero" level. The "1" and "2" selections now trigger (positive slope) only when the signal amplitude of the selected channel exceeds 0.1%FS (-60 dBFS). See the "Triggering Capability, FFTSLIDE" section on page B-15 and the "Triggering Capability, FFTGEN" section on page B-17.

DGEN Trigger Source Added to FFT Programs in Dual Domain Units

A new DGEN trigger source selection has been added to the software source trigger list of FFTGEN and is also present in the new FASTEST program. A trigger event is issued once per cycle in sinewave mode of FFTGEN and once per "scan" through the arbitrary waveform buffer of either program. This selection is functional only on Dual Domain units.

Error Messages on Waveform Mismatch

When the LOAD WAVEFORM command is used to load a disk-stored single-channel waveform without using the single-channel load option, an error message "READ ERROR ON SECOND CHANNEL" is displayed.

Error Message on Loading Waveform with No DSP Program Loaded

When the LOAD WAVEFORM command is selected but no DSP program has previously been loaded, the resulting error message display is "NO DSP PROGRAM LOADED OR THIS SAMPLING RATE NOT SUPPORTED". This error message will also be displayed if a DSP program is loaded which does not support waveform download (for example, GENANLR or HARMONIC).

Arbitrary Generator Trigger Output Added on Dual Domain Units

A trigger output signal from the arbitrary waveform generator programs (FFTSLIDE and FASTEST) has been added to Dual Domain units. A pulse is generated at the rear-panel BNC TRIG OUT connector at the beginning of each scan through the arbitrary waveform generation buffer.

-AVG Function Before MAX

The "-AVG" selection, which performs a function similar to AC-coupling by subtracting the average of a waveform from every sample, now takes place before the MAX display function. Thus, when both -AVG and MAX are selected at the same time, the MAX graphs the peak of the AC signal with any DC component effectively removed.

Note on All FFT Programs

When viewing acquired signals in the time domain with the INTERPOL waveform display selection, the final seven samples in the time record will always be displayed as zero amplitude.

FFTGEN Changes

FFTGEN Amplitude Now Usable in Nested Sweeps

SOURCE-2 DSP GENAMP may now be selected instead of DATA-2, resulting in "nested sweeps" while testing digital-input devices such as D/A converters. For example, SOURCE-1 may be set to DSP FREQUENCY for an FFT display of the distortion products at the output of a D/A converter at several different digital input levels controlled by the values entered at GRAPH BOTTOM and GRAPH TOP at SOURCE-2, with the number of amplitude values between those extremes controlled by the # STEPS field.

FFTGEN Spectrum Averaging Improved

FFTGEN spectrum averaging mode now waits for a trigger for each acquisition, rather than "free-running" following the first triggered acquisition. Also, a scaling error previously present in the 1024-average selection has been corrected.

FFTSLIDE Changes

External Hardware Trigger Added for FFTSLIDE on Dual Domain Units

An external TTL-compatible hardware trigger input has been added on Dual Domain units (SYS-300 family). A signal connected to pin 3 of the rear-panel 15-pin "general purpose serial" connector can be selected as EXTERNAL in the trigger source list of FFTSLIDE. See the "Triggering Capability, FFTSLIDE" section starting on page B-15 for details.

Note on FFTSLIDE

When using the pre-trigger capability of FFTSLIDE from any analog input and a hardware trigger source, a number of samples in addition to the pre-trigger value will be acquired. This additional number of samples depends upon the sample rate in use; at the common 48 kHz sample rate, 31 samples in addition to the pre-trigger value are acquired.

Note on MLS

When viewing MLS frequency responses with the WAVEFORM selection INTERPOL, the first seven and last seven frequency bins will be displayed as zero amplitude. Assuming full memory and the 48 kHz sample rate, this will mean zero amplitude displayed from DC to 20.5 Hz and from 23.977 kHz to 23.997 kHz.

GENANLR Changes

Autoranging Added to GENANLR

Autoranging has been added to both the LEVEL and FILT LVL channels of GENANLR. Autoranging improves speed and low amplitude resolution. As previously, the D/A converter output can be used when OUTPUT SERIAL or PARALLEL is selected near the bottom of the DSP panel, to monitor digital input signal sources or the residual signal in BANDREJECT mode. Autoranging now increases the signal amplitude at the D/A converter input, providing a more useful level and over-riding the low-amplitude distortion inherent in the D/A converter.

GENANLR Panel Fields Re-Organized

More flexibility was given to GENANLR by separating the highpass filter selection from the bandpass-bandreject-weighting filter selection. See the "Filter Responses" section starting on page F-4 for description of the modified selection fields and new combinations of filters available.

GENANLR Bandpass Filter Selectivity Improved

The bandpass filter in GENANLR has been narrowed from approximately 1/10 octave (Q=15, -3 dB bandwidth = 6.8% of center frequency) in the v2.00 program to approximately 1/13 octave (Q=19, -3 dB bandwidth = 5.2% of center frequency). Figure F-9 on page F-8 shows the response of this filter and the bandreject filter.

GENANLR Reads Zero with No Input

The GENANLR real-time displays now read zero if the input channel is turned off or if no digital input signal is present, rather than retaining the last reading.

GENANLR Frequency Counter Improved

The frequency counter function in GENANLR has been improved to reduce digit bobble.

BITTEST Changes

BITTEST now holds the last error reading in the display until another error occurs. See the "Signal Analysis" section of the BITTEST chapter, on page G-4 for full details on operation of the signal and error display fields.

Error detection with the sinewave signal has been added.

The error indication display is reset whenever the input source is changed or the waveform is changed.

The word length of the output signal is now trimmed to the value in the SIZE field, even though 24-bit words are internally generated. For input signals, only the number of bits in the SIZE field (starting with the MSB) are displayed and evaluated for errors.

Serial (RS-232) Mode

S1.EXE now supports serial (RS-232) control of "S" version systems with DSP, or of remote computer/DSP unit combinations via modem. See the REMOTE MODE FOR TRANSMISSION TESTING AND LAPTOP COMPUTER OPERATION chapter starting on page 22-1 of the main System One User's Manual for full details on serial operation.

DSP, RS-232, and Speed

Most DSP-related actions are much slower with RS-232 interfaced units than with APIB units. For example, displaying a 200-point FFT graph via APIB takes about two seconds with a 386-based, coprocessor-equipped 80386 computer. With an "S" version DSP system RS-232 interfaced to a 386 laptop computer (no modem) at 9600 baud, about 18 seconds is required. Over modems at 2400 baud, about one minute 20 seconds is required. Operation of the real-time programs GENANLR.DSP and HARMONIC.DSP will be comparable in speed to non-DSP serial mode operation after the DSP program has been downloaded.

Program and waveform download and waveform upload operations over RS-232 are slow, especially at the lower baud rates. DSP programs must download from the "master" computer to the remote System One, even if a remote computer in "Run Slave" mode should have DSP programs on its disk. DSP programs are moderately large files (about 11 kbytes to 20 kbytes) and waveforms are even larger. Downloading the FFTGEN.DSP program (about 15 kbytes) across APIB requires less than two seconds. The same action to a directly-connected "S" version unit at 9600 baud takes about 20 seconds, and via 2400 baud modems requires about one minute 25 seconds.

If the same DSP program is used on two consecutive .TST files, it will not be downloaded for the second (or succeeding) .TST since S1.EXE checks to see if it is already in place. If procedures are used where some tests have DSP programs attached and others do not, the DSP tests should all be in sequence (or the DSP program should be attached to all tests even if not needed) so that only one DSP program download is required. If different tests use different DSP programs, all the tests using each particular DSP program should be in sequence.

Upon the LOAD WAVEFORM command, generator waveforms must always download over RS-232 from the controlling computer (not from the computer in Run Slave mode). SAVE WAVEFORM causes acquired waveforms to upload via RS-232 from the DSP to the controller. Generator waveforms are about 24 kbytes and thus take about 50% longer to download than the FFTGEN.DSP program example above. Saved waveforms can vary in size from a few kilobytes up to nearly 200 kbytes, depending upon which FFT program is in use and whether the entire buffer or transform buffer is saved.

Bug Fixes

On wow and flutter tests with DSP programs, the software was requiring a realtime frequency reading in order to scale the wow and flutter measurement. This created problems if the signal was no longer present while attempting to re-analyze acquired data in DSP memory. Now, the frequency reading is locked at acquisition time and used for subsequent re-analysis by <F6>.

Tests of two-channel devices, using the STEREO selection instead of DATA-2, did not interchange channels properly between the two measurements. This function is now performed properly.

"Glitches" previously present in the D/A output while monitoring digital-input signals have been removed.

DSP PROGRAM TECHNICAL BULLETIN 94-1

This technical bulletin describes the differences between this release and earlier versions of .DSP programs. DSP program versions in this release are FASTTRIG.DSP version 2 and versions 14 of BIT-TEST.DSP, FASTTEST.DSP, FFTGEN.DSP, FFTSLIDE.DSP, GENANLR.DSP, HARMONIC.DSP, and MLS.DSP.

FEATURE CHANGES

FASTTRIG.DSP Operation Without Step Table

Version 2 of FASTTRIG.DSP can now be used in RESPwW+F, DISTORT, and NOISE modes without requiring a "step table" (.SWP file). When an .SWP file is used, FASTTRIG functions largely as it did formerly. Additionally, FASTTRIG.DSP now analyzes the downloaded generator waveform file (.WAV file) for each channel to determine the frequency of each sinewave which will be present in the signal corresponding to that channel. In effect, the DSP program thus generates internally in the DSP unit "step tables" for each channel, listing the sinewave frequencies. When S1.EXE software asks the DSP unit for values to be plotted, the following actions are taken, depending upon the mode of FASTTRIG.

NORMAL mode: all bin amplitudes between the previously-requested frequency and the currently-requested frequency are examined and the peak amplitude data in that frequency range is sent from the DSP to the PC, where it will be plotted at the currently-requested frequency. This mode results in a normal FFT display with "peak-picking" to avoid missing significant signals, and would normally be used with a high # STEPS value (SWEEP DEFINITION Panel) such as 500.

RESPwW+F mode: amplitude data is measured by the DSP only in the bins where a fundamental frequency is present in the corresponding channel of the generator waveform file. If the FREQ RES field on the FASTTRIG.DSP panel contains a value greater than zero, all bin amplitudes within ±FREQ RES of each fundamental frequency are summed by the root-sum-square (RSS) technique so as to include flutter sidebands around each fundamental. If no step table is in use by S1.EXE, the frequencies at which data is requested by the PC will rarely exactly coincide with the fundamental frequencies. The DSP unit thus interpolates between the nearest measured values above and below the requested frequency value and sends the interpolated value to the PC for plotting. If the PC asks for values at frequencies below the lowest or above the highest fundamental frequency, the DSP interpolates between the amplitude of the nearest frequency and a zero amplitude at DC or one-half the sampling rate. If a step table is in use with values which exactly correspond to the fundamental frequencies in the signal, no interpolation takes place.

DISTORT mode: FFT bins are RSS-summed between each adjacent pair of frequencies requested by the PC and the summed value is sent from DSP to PC to be plotted at the higher of the two frequencies. The bin containing each fundamental is not included in the summing. If the FREQ RES value is greater than zero, all bins within ±FREQ RES percentage of each fundamental are omitted from the summing. The frequencies of fundamental-containing bins is determined in the DSP module from the generator waveforms and is specific to each channel when the generator waveforms are not identical in both channels. Thus, if a stereo waveform (some signals different between the two channels) is in use and if the cables carrying the two channels of the stereo signal are inadvertently transposed, the signals unique to each channel will be recognized as distortion in DISTORT mode. This contrasts with version 1 of FASTTRIG.DSP, where (when a step table was in use) the DSP unit ignored all bins containing PC-requested frequencies and was therefore insensitive to channel transposition if the step table included all fundamental frequencies in both channels.

```
=DSP===============LOCAL==
 DIGITAL DATA ANALYZER

INPUT 1    -29452     DEC
INPUT 2    -29452     DEC
ERRORS 1   0          DEC
ERRORS 2   0          DEC

DGEN FREQ .9999990 kHz
     AMPL +0.00       dBFS
    VALUE             OFF
SWP TIME              OFF
WAVEFORM   SINE       VAL
SIG DISPL INTERPOL    AUTO
ERR DISPL NORMAL      HOLD

RATE      48kHz
INPUT     SERIAL
  CH-1    A
  CH-2    B
OUTPUT    SERIAL   A&B
  SIZE    16

OFF HOLD
Freeze data display on error
```

BITTEST.DSP v14 Panel, Showing New Field Which Controls Display Hold Function

NOISE mode: similar to DISTORT mode, except that only "empty" FFT bins (free of generator-related signals) are summed and sent to the PC. If the FREQ RES value is greater than zero, all "empty" bins within that percentage of each fundamental are omitted from the summing.

If a .SWP file is attached to the .TST file and TABLE is selected as ON (SWEEP DEFINITION panel), the frequency values in the .SWP file determine the points at which amplitude is measured in NORMAL and RESPwW+F modes and the ranges across which distortion and noise are summed in DISTORT and NOISE modes. However, the FFT bin values containing fundamental frequencies will always be omitted from DISTORT mode measurements even if a .SWP file is in use. Thus, distortion can now be summed across frequency ranges which are selected independently from actual fundamental signal locations. For example, fundamentals could be spaced irregularly, at lesser or greater densities in different portions of the spectrum, while still summing distortion products into 1/3 octave bands by use of a .SWP file with 1/3 octave frequency values.

BITTEST.DSP Display Freeze Control

A new panel field has been added to BITTEST.DSP to permit the user to control whether the digital input value displays "freeze" upon detection of an error, or continue following changing input values. See the figure on the following page. When this new field is set to HOLD, the input signal displays will freeze upon occurrence of the first error so that the input value associated with the error can be noted. When the control field is set to OFF, the digital input value displays will continue updating even when errors occur.

BITTEST.DSP Error Indication with No Signal

Earlier versions of BITTEST.DSP did not produce an error reading if no signal was present, based on the assumption that errors are undefined without a signal. Many users requested that absence of signal should also produce an error reading, since it was otherwise possible for an entire suite of bit error tests to produce a "pass" result when the unit under test had inadvertently not been connected. Version 14 incorporates this feature, so that tests will fail if no signal is present. Thus, care should be taken that a signal is connected and present before any test is run (<F9>) when BITTEST is used.

BUG FIXES

INCORRECT DIGITAL DOMAIN SAMPLE RATE BUG FIXED

Version 1 of FASTTRIG.DSP and versions 10, 11, and 12 of the other DSP programs with digital domain output capability contained a bug which caused incorrect digital domain output sample rate. Technical Bulletin 92-2 described "work-arounds" to overcome this problem. The new version 2 of FASTTRIG.DSP and version 14 of the other DSP programs contain a cure for this problem, so work-arounds are no longer necessary.

DSP User's Manual, Table of Contents

A. DSP INTRODUCTION

The hardware architecture of System One's DSP (Digital Signal Processing) module is flexible, with powerful digital processing of signals from both digital and analog inputs and generation of signals at digital and analog outputs possible. Like any computer, software is required to instruct the DSP hardware what to do. Audio Precision makes available a number of software programs with the .DSP file name extension. This introductory chapter outlines which .DSP programs are designed for which applications and describes the features of the DSP hardware and programs which are common to most or all the programs. Following chapters will describe the unique features and applications of each of the programs in more detail.

A.1. Loading DSP Software

The DSP programs plus a wide variety of test files, procedures, sweep tables, stored waveforms, and special DSP performance checks are furnished on eight diskettes. A specific hard disk sub-directory structure is recommended, with the contents of the distribution diskettes to be copied into specific sub-directories. This section assumes that a hard-disk-based computer is to be used and that the procedures in Chapter 5 of the System One User's Manual, "Loading the Software", have been implemented. You will thus have a directory C:\AUDIO located in the root, containing S1.EXE and the non-DSP tests, procedures, and utilities except for Performance Checks. A C:\AUDIO\PERFCHEK subdirectory will contain those performance checks.

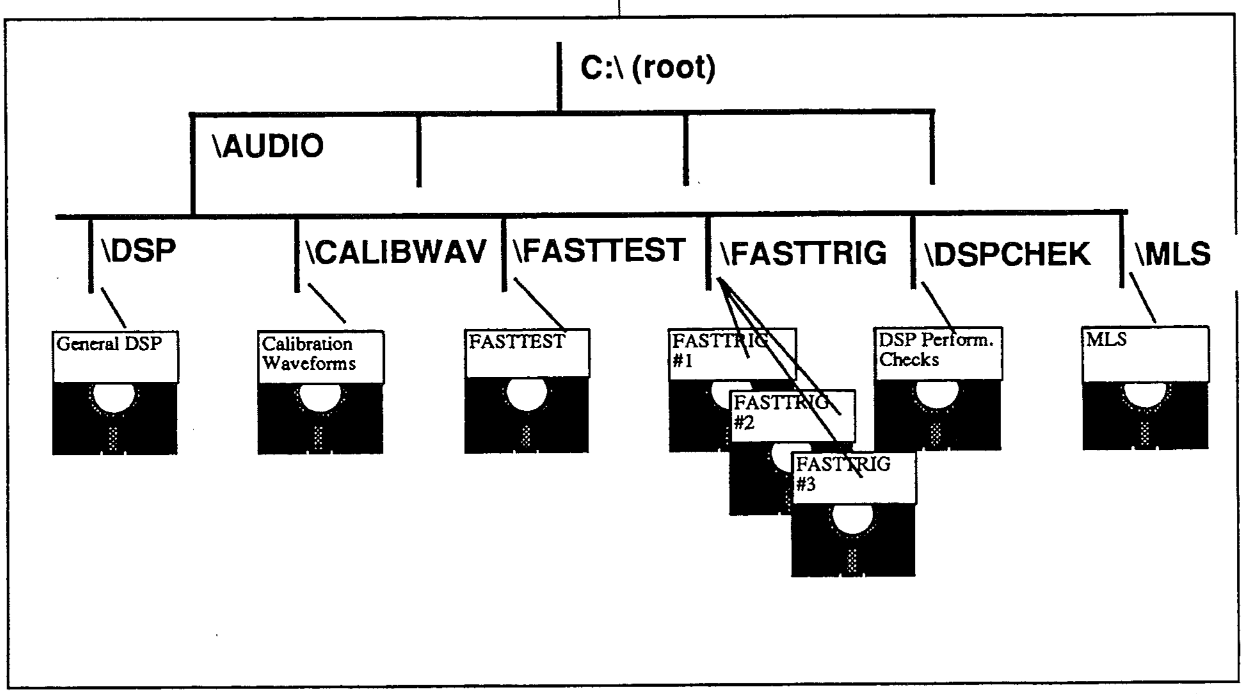


Figure A-1 Recommended Hard Disk Directory Structure and File Location

Make new sub-directories under the C:\AUDIO sub-directory with the names \DSP, \CALIBWAV, \FASTTEST, \FASTTRIG, \MLS, and \DSPCHEK. Copy the contents of the "General DSP" diskette into C:\AUDIO\DSP. Copy the "Calibration Waveforms" diskette into C:\AUDIO\CALIBWAV. Copy the "FASTTEST" diskette files into C:\AUDIO\FASTTEST. Copy the three "FASTTRIG" diskettes into C:\AUDIO\FASTTRIG. Copy the "MLS" diskette files into C:\AUDIO\MLS. Copy the "DSP Performance Checks" diskette into C:\AUDIO\DSPCHEK. See Figure A-1 for a schematic representation of this structure.

A.1.1. DOS PATH and APPEND Commands

In order that S1.EXE can be run, DSP programs can be named, and the key FASTTEST.DSP sweep tables used from any directory on the hard disk, add PATH and APPEND commands to your AUTOEXEC.BAT file (or append additional sub-directory path names to those commands if they already exist). See the discussions of the PATH and APPEND commands in Chapter 5 of the System One User's Manual. The only directory necessary in the PATH command is C:\AUDIO; as recommended in the System One User's Manual. The following sub-directories may be added to the APPEND command in order that all DSP programs can be loaded and waveform and sweep files used regardless of which sub-directory is current:

APPENDC:\AUDIO\DSP;C:\AUDIO\MLS;C:\AUDIO\FASTTEST;C:\AUDIO\FASTTRIG;C:\AUDIO\CALIBWAV;

Note that the APPEND statement is to be typed on one line of the batch file.

However, the APPEND command has the disadvantage that it will prevent files of the same name from being used in more than one of the subdirectories in the APPEND list. If you attempt to create and save a file whose name already exists in one of the other directories listed, you will receive a warning that the name exists and prompt you for a "Y" to overwrite. If you supply the "Y", the message "File Not Found" will be given and the file will not be saved. Therefore, it may be more satisfactory to list only one subdirectory in the APPEND command and to locate all .DSP programs plus any other data files which are to be universally accessible in that one subdirectory.

Note that the PATH and APPEND commands make executable and data files accessible from any sub-directory, but they will not be displayed on the S1.EXE LOAD, and NAMES directories unless they are in the current directory. Thus, you can move into any directory, type S1 <Enter>, and S1.EXE software will load. The NAMES PROGRAM command will not show any .DSP files unless you are in the directory where they are located, but if you type in the DSP program name, the program will load. Similarly, LOAD PROCEDURE will only show procedure names in the current directory. If you type in the name of a procedure in one of the directories listed in the APPEND command, that procedure will load and run including all test files, DSP program files, sweep tables, limit files, etc., as long as they are all located in one or more of the directories listed in the APPEND command.

A.2. Applications vs DSP Programs

A.2.1. Waveform Display

For time-domain display of waveforms up through 80 kHz (digital storage oscilloscope mode), FFTSLIDE.DSP is recommended. Signal amplitude range is that of the analog front end of System One, from a few microvolts up through approximately 160 Volts RMS. FFTSLIDE.DSP permits triggering from positive or negative slope of the signal or from the System One generator or ac mains. Pre-trigger can be used to retain signal prior to the trigger and thus display the signal which may have caused the trigger event. FFTGEN.DSP also includes waveform display capability, but with less powerful triggering capabilities. FFTGEN.DSP is recommended for digital-domain waveform display, since it includes sinewave generation features and thus can perform the entire stimulus-response capability in the digital domain.

A.2.2. Spectrum Analysis

FFT spectrum analysis through 80 kHz is offered by both FFTSLIDE.DSP and FFTGEN.DSP. Again, FFTSLIDE.DSP has the more powerful and flexible triggering capabilities. FFTSLIDE.DSP also offers the ability to do spectrum analysis of any section of an acquired waveform of up to 640 ms at usual sampling rates. FFTGEN.DSP offers averaging of the FFT magnitudes, which reduces the variance of random noise in the signal.

A.2.3. Individual Harmonic Analysis

HARMONIC.DSP offers a highly-selective bandpass filter for analog signals which can track (at fundamental or any harmonic through the 9th) the analog generator or the frequency of an externally-furnished signal.

A.2.4. Selective Amplitude Measurements

HARMONIC.DSP and either FFTSLIDE.DSP or FFTGEN.DSP can offer selective amplitude measurement at specific frequencies. The two FFT programs offer the greatest selectivity, effectively down to less than three Hz bandwidth across the 20 kHz audio spectrum. HARMONIC.DSP, while not as selective, is a real-time program which can thus offer rapidly-updating measurements for adjustments.

A.2.5. Rapid Frequency Response, Distortion, Phase, and Noise Measurements

FASTTEST.DSP and FASTTRIG.DSP combine FFT analysis with the ability to simultaneously generate large numbers of sinewaves across the audio spectrum from down-loaded waveform files. These programs thus provide a "parallel" testing concept for audio devices, stimulating the device simultaneously at all frequencies of interest, rather than the usual "serial" testing in which a single sinewave is swept or stepped across the audio band, one frequency at a time. FASTTEST.DSP can stimulate a device and acquire signal in 1-2 seconds, then analyze the acquired signal during the next 2-5 seconds for total distortion, frequency response, interchannel phase, stereo separation, and noise in the presence of signal. FASTTRIG.DSP has the further ability to continually monitor incoming signal and capture a brief burst of signal when it matches the reference signal previously downloaded to FASTTRIG. The burst of test signal can be as short as 250 milliseconds for most applications and can provide useful testing with bursts as short as 40 milliseconds.

A.2.6. Quasi-Anechoic Loudspeaker Response Testing

MLS.DSP generates Maximum Length Sequence noise as the stimulus for a loudspeaker under test. MLS.DSP then takes the output signal from a measurement microphone, cross-correlates the measured signal with the transmitted signal to obtain impulse response, and performs an FFT of the impulse response. Any specific time section of the impulse response may be selected for analysis. Typically, the "first arrival" signal is selected, prior to the arrival of the first reflection, providing a quasi-anechoic response measurement independent of the room in which the measurements are made.

A.2.7. Digital Domain Performance Testing

GENANLR.DSP is designed for stimulus-response testing of digital audio devices. Sinewave generation is provided plus the ability to measure weighted and unweighted noise, frequency response, gain and loss, level, THD+N, and crosstalk in the digital domain. For mixed-domain measurements (A/D, D/A, etc.), the analog stimulus and response capability of System One is used in conjunction with GENANLR.DSP.

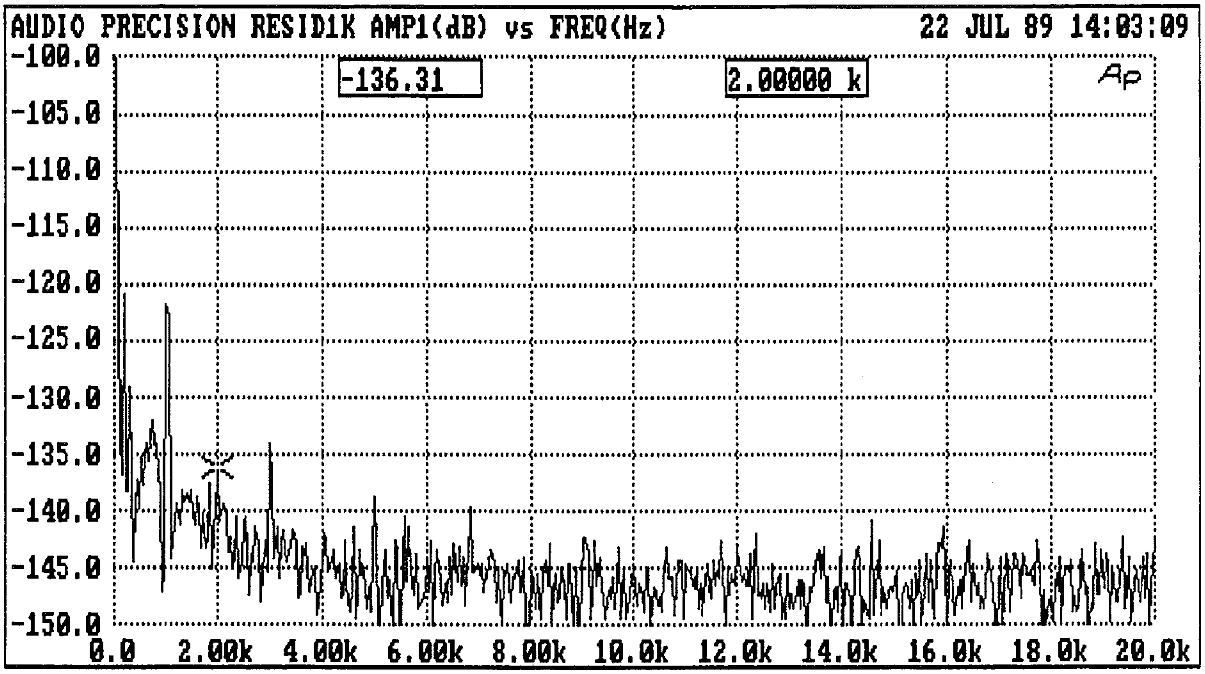


Figure A-2 FFT Spectrum Analysis of Residual Distortion of Analog Generator and Analyzer

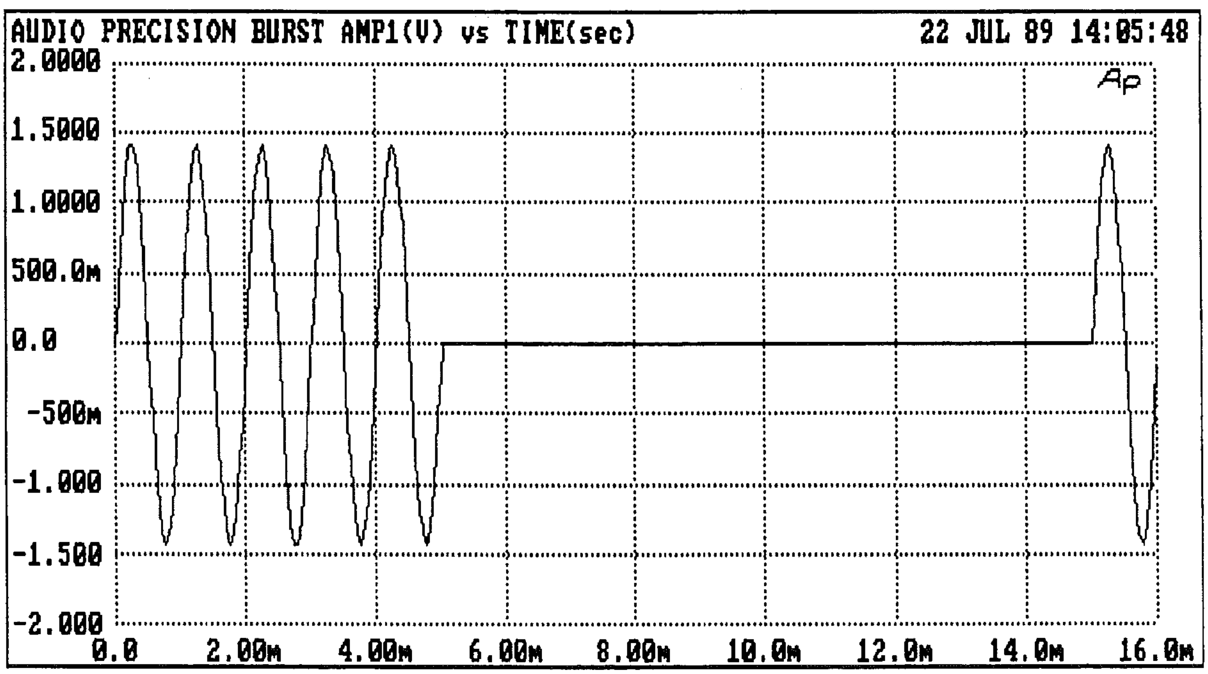


Figure A-3 Sine Burst from System One BUR-GEN Option

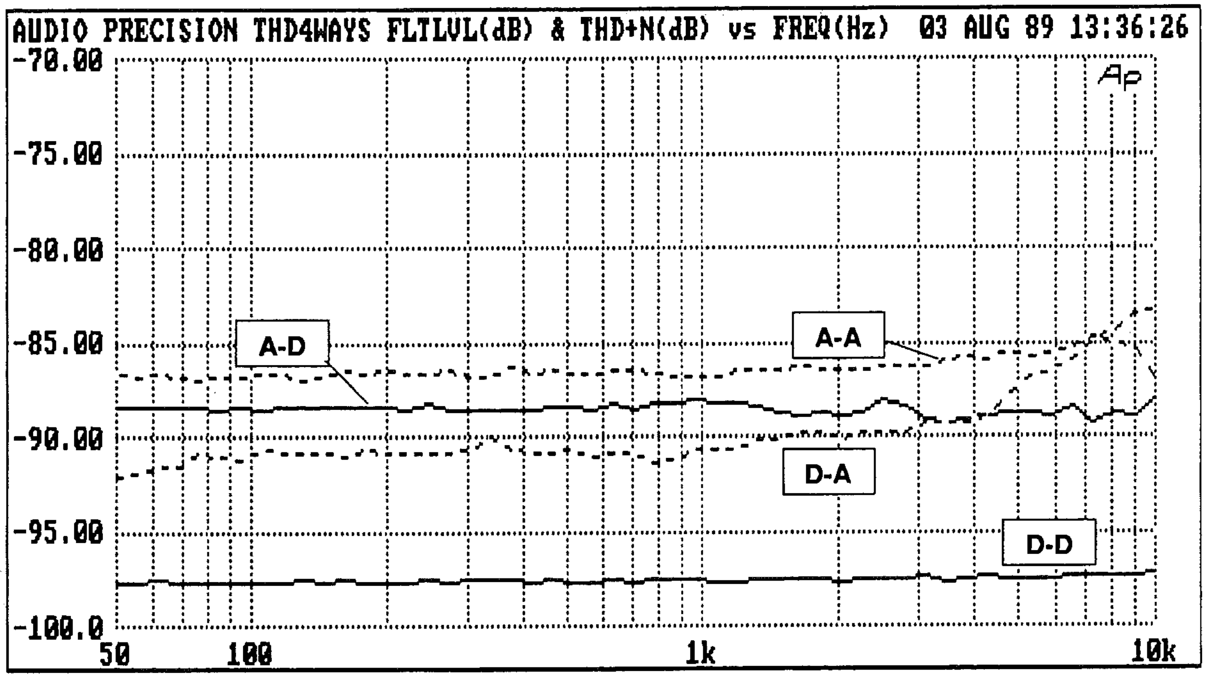


Figure A-4 RDAT THD+N vs Frequency In All Combinations of Analog and Digital Inputs and Outputs

A.2.8. Digital Interface, Transmission, and Storage Medium Error Testing

BITTEST.DSP generates a number of special signals such as pseudorandom noise, walking bit patterns, and "digital dc" in addition to sinewaves. The analysis section of BITTEST.DSP then analyzes any of these waveforms for bit errors. Any amount of time delay is acceptable, so any of these signals may be recorded and later reproduced for error-checking of the recording process and storage medium.

A.3. Typical DSP Applications

The Digital Signal Processor modules are installed in the lower left-hand compartment of System One, underneath the generator. When the DSP modules are installed, System One becomes an SYS-200 series ("System One + DSP") or SYS-300 series ("System One Dual Domain"). Both these series can make a variety of enhanced audio measurements on analog audio signals. Figure A-2, for example, is a high-resolution spectrum analysis via FFT (Fast Fourier Transform) program of the residual distortion of a typical System One generator and analyzer at 1 kHz. The fundamental signal, attenuated by the analog notch filter, is visible at -122 dB. Several harmonics are visible in the -135 to -140 dB area, plus the noise floor at about -145 dB.

Figure A-3 is an example of the waveform display function possible with the FFT programs. The signal is the SINE BURST mode available from System One's BUR-GEN option. SYS-200 and SYS-300 series can acquire waveforms, save them to the computer disk, and later download the waveforms to the DSP for further waveform display or FFT analysis.

The SYS-300 series is further distinguished by its ability to measure and/or stimulate digital audio devices in the digital domain, in parallel and several serial formats. This capability, added to the existing System One capability for audio signal generation and measurement in the analog domain, allows the

SYS-300 models to test analog and digital audio devices in any of the four possible combinations; A/A, A/D, D/A, or D/D. Audio Precision has coined the term Dual Domain™ to describe the mixed analog and digital testing. Figure A-4 shows THD+N versus frequency for a 16-bit PCM digital recorder, measured in each of these possible combinations of domain. The D/D curve at -98 dB demonstrates in practical measurement the theoretical quantization noise and distortion value of a 16-bit PCM system. The D/A curve shows the recorder's output DAC adding about 6 dB distortion above theoretical minimum. The A/D and A/A curves show the recorder's input A/D converter to be the dominant source of distortion, adding some 10-12 dB above theoretical minima.

A.4. DSP Architecture

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

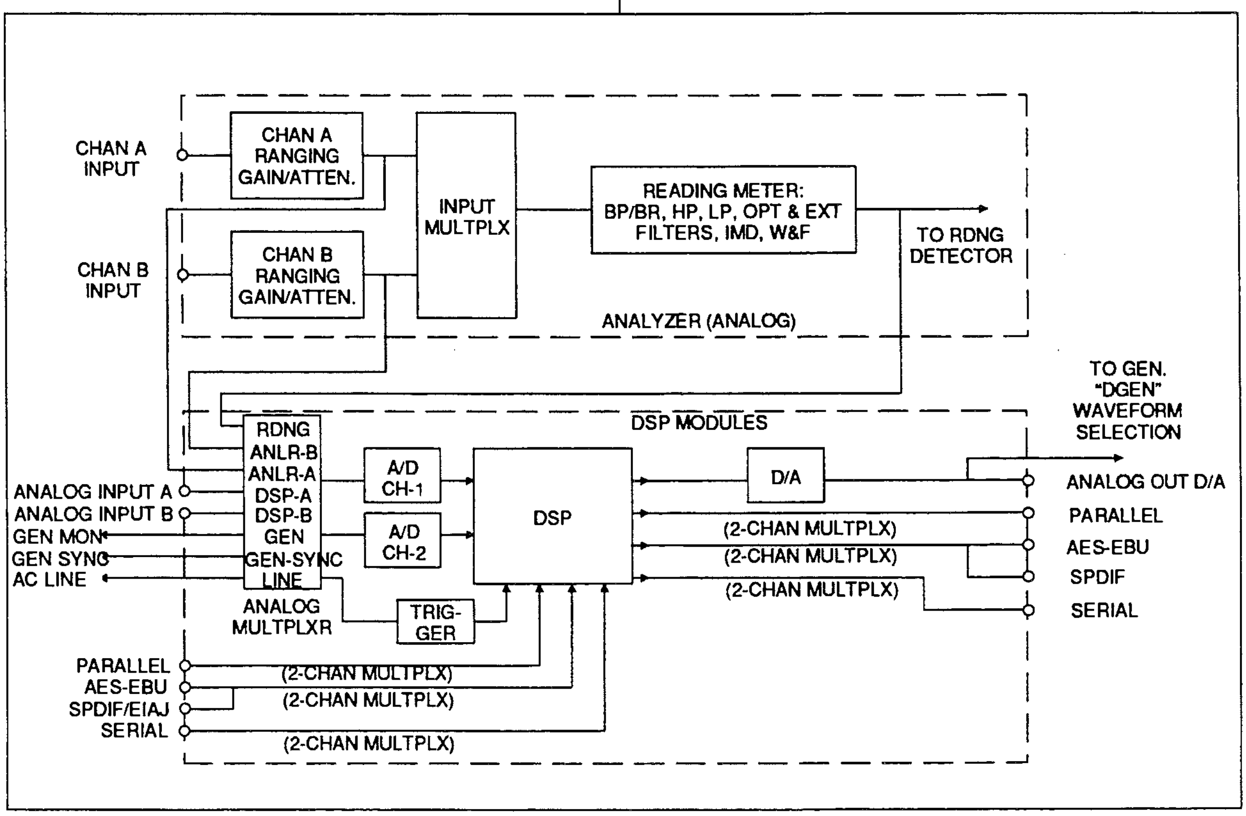


Figure A-5 Simplified Block Diagram, DSP Inputs and Outputs

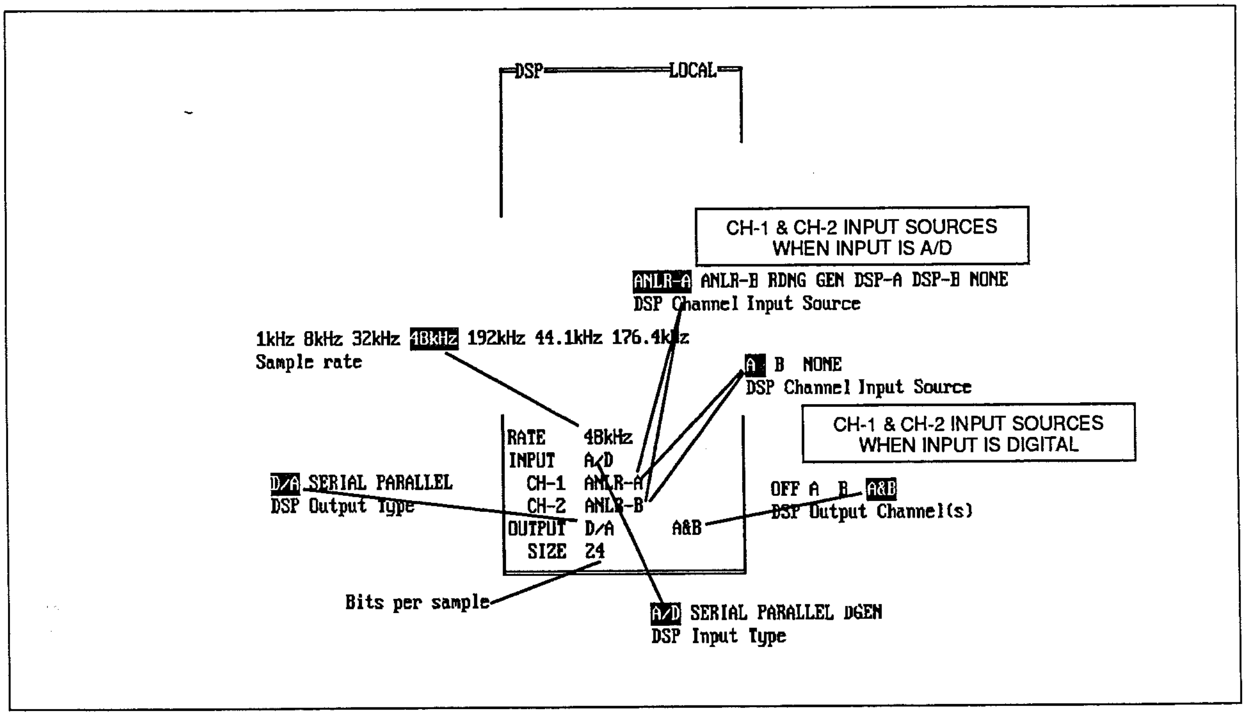


Figure A-6 DSP Panel, Fields Common to All Programs. Not All Sample Rates Are Available to All Programs

CHANNEL B, and READING monitor connectors of the analyzer, the MONITOR connector of the generator, and two additional BNC input connectors dedicated to the DSP.

The SYS-300 (Dual Domain) series can additionally acquire and generate 24-bit digital audio signals in the AES/EBU and SPDIF serial formats at front panel connectors, and general-purpose serial and parallel-format signals at connectors at the rear of the enclosure. See Figure A-5 for a simplified block diagram of the signal connections to and from the DSP modules. Note that the BNC connectors labeled ANALOG INPUTS A and B are unbalanced, DC-coupled inputs of 2 Volts RMS full scale. This allows measurements below the low frequency limit of the normal analog analyzer AC coupled inputs. The "TRIGGER" block is a hardware trigger circuit which shares the same interrupt with the output signal circuitry. Thus, programs which have signal generation capability have only software triggering capability.

A.5. Downloading DSP Programs

As a special-purpose digital computer, the DSP module requires programs to tell it what to do. These files (identified by the file extension .DSP) are downloaded from the personal computer to the DSP modules within System One by the NAMES PROGRAM command. If a .TST file is loaded to which a DSP program had previously been attached via the NAMES PROGRAM command, the DSP program automatically downloads when the .TST file is loaded into memory. The download is not performed if that same DSP program is already in place from the previous test.

The DSP panel is reproduced in Figure A-6. Until a DSP program is downloaded to the DSP module, most of the panel is blank. The control and display fields which load into the DSP panel depend upon which program is downloaded. Only the fields at the bottom of the panel are common to all DSP programs. These fixed fields permit selection of the sample rate at which the input data is acquired, whether the input is analog, serial digital,

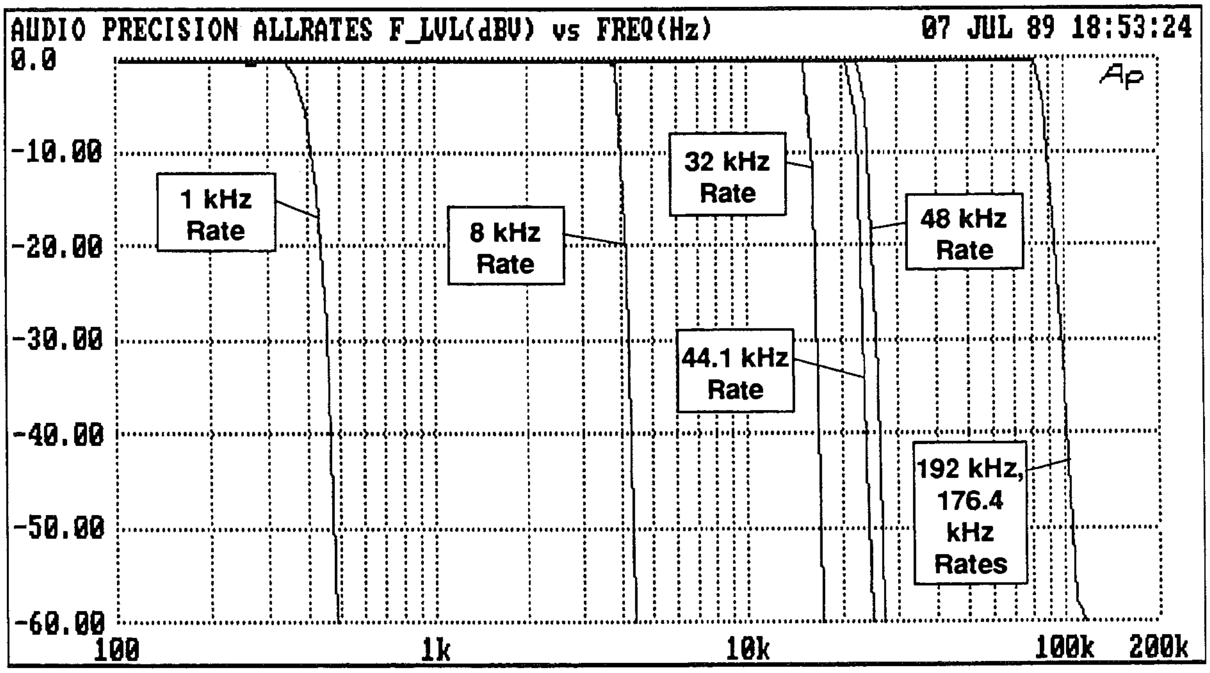


Figure A-7 Frequency Response vs Sampling Rate, DSP

or parallel digital, and what signal drives each input. If a generator capability is included in the DSP program, these fields will select the destination of the internally-generated signal, its sample rate, and dither amplitude.

As with other System One software panels, simple descriptions about the function of individual fields on the DSP panel may be found at the bottom left of the panel when the cursor is placed on each field.

A.6. DSP Help Screens

The HELP DSP menu selection presents a screen of unique information which is loaded by each .DSP program selected. This information describes general operation of the program, limitations on use of inputs and outputs, and cautions regarding setup. The assignment of readings to fields on the Sweep Settling Panel is specified on the HELP DSP screen for easy reference. If a question arises on operation of any .DSP program, the HELP screen should be the first place to look for information.

A.7. DSP Input Operation

A.7.1. Rate vs Bandwidth

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

The bandwidth available with any rate cannot exceed half the sample rate, as originally shown by Nyquist. In practice, the useful bandwidth is somewhat less than half the sample rate. System One's DSP module, when acquiring analog signals, actually functions with the A/D converters always operating at a 192 kHz or 176.4 kHz sample rate and with an anti-alias low-pass filter bandwidth from zero to 80 kHz. When a lower sample rate is selected, a second DSP chip functions as a decimator, effectively scaling down both the sample rate and the anti-alias filter corner frequency to lower values. Figure A-7 shows the typical measured frequency response of the analog input channels at each sample rate. Typical -3 dB points are 375 Hz at the 1 kHz rate, 3.94 kHz at the 8 kHz rate, 15.7 kHz at the 32 kHz rate, 21.6 kHz at the 44.1 kHz rate, 23.5 kHz at the 48 kHz rate, and 85.3 kHz at the 176.4 and 192 kHz rates. Linear phase acquisition of analog signals is only available at sample rates of 48 kHz and lower. The 1 kHz sample rate is intended only for acquisition and spectral analysis of very low frequency signals such as wow and flutter.

The bandwidth of the signal being acquired may be further limited by the processing selected in the System One analog hardware. For example, when acquiring through the READING (RDNG) signal path, the lowpass, highpass, tunable bandpass, and any optional filters selected will all affect the signal. When measuring Intermodulation signals from the READING meter, the measurement bandwidth will be limited by the IMD mode selected. Consult the System One User's Manual for information on the measurement bandwidth in any analyzer mode.

A.7.2. DSP Input Signal Selection

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

The INPUT fields near the bottom of the DSP panel permit selection of the signal type and source. See Figure A-5 for a graphic explanation of its function. The DSP is a full two-channel device, with two digital input channels and two separate A/D converters and anti-alias filters for two analog signals. Note that the DSP channels have been named CH-1 and CH-2, to avoid confusion with the analog analyzer's A and B input channels.

If A/D is selected as INPUT, one or two channels of analog signal will be converted to digital samples by the two 16-bit A/D converters in the DSP unit and presented to the DSP chip for analysis. When A/D is selected, the CH-1 and CH-2 choices available on the next two lines are ANLR-A, ANLR-B, RDNG, GEN, DSP-A, DSP-B, and OFF. ANLR-A and ANLR-B are buffered ac-coupled signals taken from the analog analyzer CHANNEL A and CHANNEL B inputs, respectively, following the input ranging circuits but before any filtering. RDNG is the ac-coupled analog analyzer READING meter signal, following all analog processing in the analyzer. The RDNG signal thus follows the analog analyzer's tunable bandpass-bandreject filter, any standard high-pass and low-pass filters selected, and any plug-in option filter or external filter selected. In the intermodulation distortion modes, the RDNG signal follows the AM detection (SMPTE/DIN mode) and filtering (all IMD modes) of the intermodulation analyzer board. In wow and flutter mode, the RDNG signal is the weighted or unweighted (as selected) output of the wow and flutter discriminator.

GEN refers to the analog generator's MONITOR OUTPUT signal. This is an approximate 2.8 Volt p-p constant-amplitude version of the analog generator output waveform which is present even when OFF has been selected on the GENERATOR panel. This provides a convenient way to look at the waveform of a signal going out the System One analog generator output. DSP-A and DSP-B refer to the dc-coupled, unbalanced BNC input connectors labeled ANALOG INPUTS A and B on the DSP module.

In System One Dual Domain models (SYS-300 series), if SERIAL, PARALLEL, or DGEN are chosen as the source on the INPUT line, the choices at CH-1 and CH-2 become A, B, and NONE. The parallel input connector is located on the instrument rear panel. SERIAL includes three serial data formats, selected by the UTIL SERIAL-DSP MODE menu command. The formats are the AES/EBU digital audio format at a front panel female XLR connector, the SPDIF/EIAJ (Sony Philips Digital Interface) digital format at a front-panel RCA phono type connector (with an optical input connector functionally in parallel with the SPDIF/EIAJ input connector after serial number SYS1-32214), and a general-purpose SERIAL format at a rear-panel connector. On units before S/N SYS1-32214, the AES/EBU and SPDIF/EIAJ inputs are effectively in parallel and cables should not be connected to both at the same time. On units S/N SYS1-32214 and later, the optical, coaxial, and AES/EBU inputs may all have cables connected at the same time. System One Dual Domain will sense the presence of signal and select one of these inputs according to the following priority if two or more have signal present simultaneously: AES/EBU first, optical second, coaxial last.

DGEN refers to the digital generator output. The DGEN selection may be thought of as functionally similar to the GEN MONITOR path between the analog generator and analog analyzer. A and B refer to the two multiplexed channels available in any of these digital formats. These are often assigned, respectively, as left and right channels of audio in a stereo recording or transmission.

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

A.7.2.1. Front-Panel Digital Inputs Below Serial Number SYS1-32214

On Dual Domain units with serial numbers below SYS1-32214, the DSP module front panel includes XLR female and male connectors for the AES/EBU input and output, female RCA phono connectors for the SPDIF/EIAJ input and output, and three BNC connectors for the two A/D inputs and the D/A output. The AES/EBU and SPDIF/EIAJ input connectors are functionally in parallel and should not both be connected at the same time. The AES/EBU input impedance is 240 Ohms, as specified in the original AES/EBU standard. If an optical panel option retrofit has been installed, the unit will function as described below for serial numbers of SYS1-32214 and above.

A.7.2.2. Front-Panel Digital Inputs From Serial Number SYS1-32214

Starting with S/N SYS1-32214, optical connectors were added for SPDIF/EIAJ input and output. The optical and coaxial (RCA) connectors are functionally in parallel, both for the input and output, so no "optical" software selection is required under the UTIL SERIAL MODE menu selection. Cables may be connected to one, two, or all three inputs simultaneously. When signals are present at two or more of the input connectors, System One decides which signal to use depending upon a priority list. Top priority is the AES/EBU input, second priority is the optical SPDIF/EIAJ input, and the lowest priority is the coaxial SPDIF/EIAJ (RCA phono) input.

A.7.2.3. AES/EBU Input Termination

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

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

A.8. DSP Output Operation

Some of the .DSP programs include signal generation capability. Signals generated in any of the DSP hardware models can be furnished in analog format. In a Dual Domain unit, they can also be furnished in digital formats. Some DSP programs may use the analog output capability simultaneously with the digital output. Consult the chapter for the DSP program being used for specific information.

A.8.1. Analog Outputs

When the OUTPUT D/A selection is made near the bottom of the DSP panel, analog output from the digital generator is available via an internal 16-bit D/A converter with approximately 2 Volts RMS full scale output. Distortion of the D/A converter is typical of 16-bit converters at about -85 to -90 dB, considerably inferior to the System One analog generator. The channel selections A, B, A&B on the OUTPUT line of the DSP panel all have the same effect when D/A is selected, since there is only a single D/A output.

A.8.2. Generator Analog Output via D/A BNC Connector

This analog output signal may be obtained at the ANALOG OUTPUT D/A connector of the DSP module, ground-referenced, with a typical two Volt RMS amplitude at 0 dBFS (100% FS) amplitude. Amplitude control at this connector is available only by varying the AMPLITUDE field on the DSP panel. Thus, fewer bits of the output D/A are in use at lower amplitudes. Distortion as a percentage of the signal amplitude will increase at lower amplitudes. Likewise, resolution decreases at lower amplitudes. This output is DC coupled and the signal is inverted relative to the digital outputs and to the analog generator output.

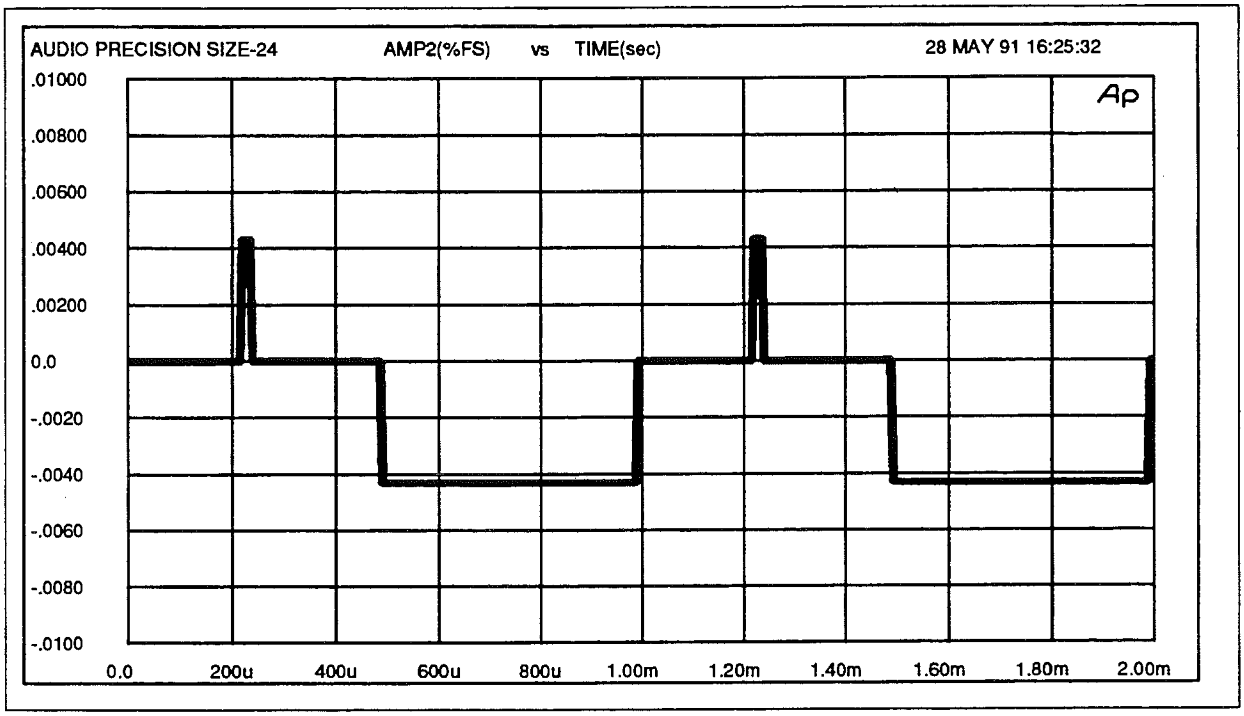
A.8.3. Generator Analog Output via Analog Generator Output Stage

The D/A output signal may also be routed through System One's analog generator amplitude control circuitry, power amplifier and output transformer, and circuitry for selection of source impedance, balanced vs unbalanced connections, and floating vs ground-referenced connection. To route the D/A signal through the analog generator outputs, select WAVEFORM DGEN on the GENERATOR panel. The digital generator output AMPLITUDE field on the DSP panel will then be over-ridden (set to maximum) so that all control of amplitude comes from the AMPLITUDE field on the GENERATOR panel. Frequency is controlled from the FREQUENCY field on the DSP panel. In this mode, the digital signal constantly operates at the full 16-bit resolution of the D/A. Thus, the output waveform distortion is essentially unaffected by the AMPLITUDE selected on the GENERATOR panel. Resolution is also improved at low amplitudes.

When the D/A output is routed through the analog generator output stage by the WAVEFORM DGEN selection, the GENERATOR panel OUTPUT fields provide normal stereo mode selection.

A.8.4. Digital Audio Outputs

SERIAL or PARALLEL may also be selected on the OUTPUT line if a Dual Domain (SYS-300) system is in use. The general-purpose parallel output connector is located on the rear panel. The UTIL SERIAL-DSP MODE command of the menu permits selection among three serial formats and connectors. The general-purpose serial output and input share a 15-pin D-sub connector on the rear panel. The AES/EBU output is a male XLR connector on the front panel. On units before serial number SYS1-32214, the AES/EBU and SPDIF/EIAJ (RCA phono) connectors are effectively in parallel and should not be used at the same time. Starting with serial number SYS1-32214, an optical output connector was added to the front panel. On these later units, the optical, SPDIF/EIAJ coaxial (RCA), and AES/EBU connectors are driven in parallel, are



A-8 Extremely Low Amplitude (-90.31 dBFS) Sinewave with SIZE Setting 24, Passed Through 16-Bit System. Truncation Visible as Wide Negative Portions, Narrow Positive Portions

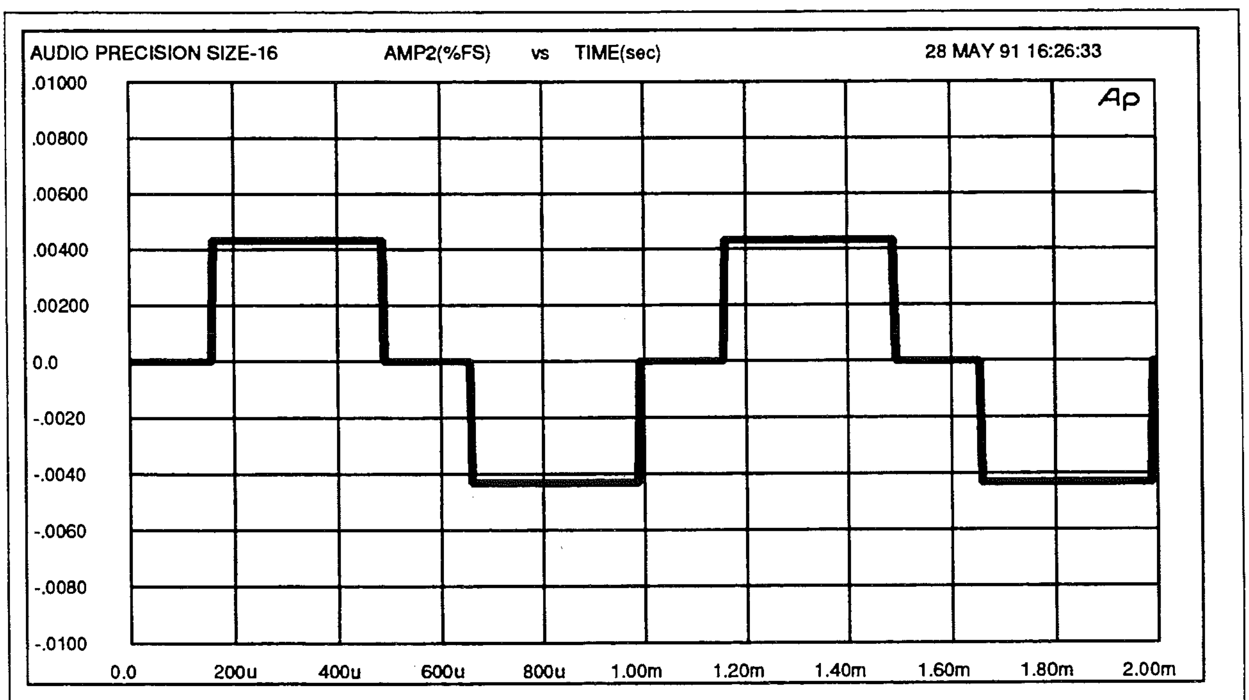


Figure A-9 Same Sinewave with SIZE Setting 16, Passed Through 16-Bit System. DSP Rounds Output Signal to 16 Bits, Signal is Symmetrical

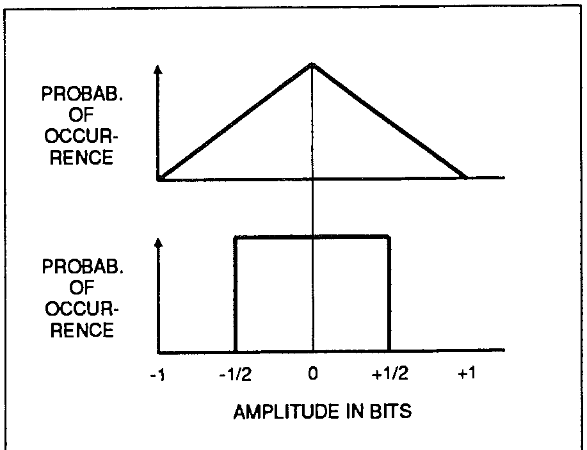


Figure A-10 Triangular (Above) and Rectangular (Below) Dither Probability Functions

all buffered and may all be used simultaneously. All digital formats are multiplexed dual channel, 24-bit linear 2's complement format.

Since the digital interfaces are all two-channel multiplexed signals, the field at the right end of the OUTPUT line permits the signal to be present as channel A, channel B, both, or off.

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

A.8.5. Size Field

In except BITTEST.DSP the SIZE field at the bottom of the DSP panel determines what word length the generated output signal will be rounded to, and also controls the amplitude of any dither used (see below). If the external digital device under test has lower resolution, it should be connected to the most significant bits of System One Dual Domain's output. Setting the SIZE field to the word length of the external signal then causes the DSP to round to that word length, producing a properly-symmetrical signal. When using the AES/EBU, optical or SPDIF outputs the bits below this point will be zeroed. If the SIZE field is set to a longer word length, the output word from the DSP generator is effectively truncated by its connection to a device with a shorter word length. When these signals are input through the AES/EBU, SPDIF or optical ports the bits below this point will be zeroed before being measured.

Figures A-8 and A-9 show a low-amplitude (-90.31 dBFS) 1 kHz sine wave passed through a 16-bit device under test. In Figure A-8, the SIZE field was set to 24 bits. The DSP rounds to 24 bits, but truncation to 16 bits occurs in passing through the external 16-bit device. Only the single sample at sinewave positive peak reaches the +1 LSB level, with truncation forcing all other samples to lower digital values. In Figure A-9, with the SIZE set to 16 bits, each 24-bit DSP generator output sample is rounded to the nearest 16-bit value and the resulting signal is properly symmetrical between +1 LSB and -1 LSB.

A.8.6. Dither

If dither has not been turned off by use of the UTIL SERIAL-DSP DITHER OFF menu command, the SIZE field at the bottom of the DSP panel controls the amplitude of dither added to the digital generator output (except during use of BITTEST.DSP which has no dither available). Dither is noise combined with the signal to improve linearity, reduce distortion at low amplitudes, and extend the linear operating range below the theoretical minimum for undithered PCM signals of any particular resolution. The additional noise is introduced before quantizing and serves to randomize the quantization distortion and produce an undistorted signal with a slightly higher noise floor. The digital generator generates a 24-bit resolution signal at all times. When the digital device under test has less than 24-bit resolution, only the higher (most significant) bits from the generator will be used. If an undithered signal is desired, turn dither off via the UTIL SERIAL-DSP

DITHER command. To set dither at any desired bit level, enter that bit number in the SIZE field. For example, proper dither for a 16-bit system is obtained by entering 16 in this field. ***When the D/A converter is selected as the output port, the SIZE field is over-ridden and internally set to 16 bits, regardless of the panel SIZE setting.***

Three choices of dither are available; TRIANGULAR, RECTANGULAR, and SHAPED (plus OFF). The selection between them is made via the UTIL SERIAL-DSP DITHER menu command. Note that the dither selected also appears at the parallel digital output, even though the menu command implies serial. TRIANGULAR dither is the default choice. TRIANGULAR and RECTANGULAR refer to the shape of the probability functions of the two types of dither; see Figure A-10 for an illustration of these probability functions. TRIANGULAR dither can add or subtract up to one bit peak amplitude at the selected SIZE. RECTANGULAR dither can add or subtract only 1/2 bit amplitude. Thus, TRIANGULAR dither increases the noise floor more than RECTANGULAR. However, RECTANGULAR dither will result in variations of the noise floor as a function of signal amplitude, while the noise floor is independent of signal with TRIANGULAR dither. Shaped dither is TRIANGULAR probability function dither which has further been spectrally-shaped by decreasing its amplitude at low frequencies and increasing it at high frequencies. The audible effect of SHAPED dither is thus significantly reduced, since the human ear is less sensitive at the higher frequencies where the dither is greater.

Note that only one dither generator is used for the two channels so that a subtraction of the two channel signals by the device under test will produce zero.

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

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═══════════ DSP DIGITAL SERIAL I/O CONFIGURATION AND INFORMATION ═══════════


DSP digital interface mode = AES/EBU
DSP dither type            = Triangular

DSP Receive Sync           = LOCKED
DSP Transmit Sync          = UNLOCKED

    BYTE                                         SAMPLE                    CRCC
    ZERO                                         ADDRESS

      0  1  2  3  4  5  6  7  8  9  10 11 12 13 14 15 16 17 18 19 20 21 22 23
RCV : 04 00 00 00 00 00 00 00 00 00 00 00 00 00 A3 11 1C 00 00 00 00 00 00 4C
XMIT: 04 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00 00
═══════════════════════════════════════════════════════════════════════════
SERIAL-DSP: TRANSMIT RECEIVE MODE DITHER
Enter channel status data to transmit        AUDIO PRECISION SYSTEM ONE, v 2.10
```

Figure A-11 Screen Display, UTIL AES/EBU TRANSMIT

A.9. AES/EBU and SPDIF/EIAJ Interfaces

A.9.1. AES/EBU Status Bytes

The AES/EBU digital audio transmission standard (AES3-1985, also ANSI S4.40-1985) reserves 24 8-bit status bytes in addition to two channels of digitized audio signals. The UTIL SERIAL-DSP RECEIVE menu command permits display of received status bytes at the AES/EBU and SPDIF/EIAJ output connectors of Dual Domain units. The UTIL SERIAL-DSP TRANSMIT menu command permits control over the status bytes transmitted at the AES/EBU and SPDIF/EIAJ output connectors of Dual Domain units.

The AES/EBU standard defines the use of many of the status bytes. Some of those status bytes are also defined in some of the consumer digital audio transmission standards which are very similar to the AES/EBU standard. Use of byte zero (and sometimes bytes 1-3) is critical to proper interfacing to digital audio devices, and several of the other bytes may be useful.

A.9.2. Byte Zero

Byte zero defines (when transmitted) and displays (when received) a number of important parameters including use and type of emphasis, selected sample rate, and consumer vs professional use. In order to successfully transmit digital data to a device such as a digital recorder, first use SERIAL-DSP RECEIVE to display the codes coming from the machine while in playback mode. Duplicating those codes via SERIAL-DSP TRANSMIT will then normally cause the machine to accept the digital signal from System One Dual Domain. Changes from the received code will be necessary if, for example, the default playback mode uses emphasis but the user wishes to test without emphasis. Some of the common two-character hexadecimal codes for byte zero are shown in the following table:

Hex Code	Emph.	Pro/Consumer	Misc
00	no	consumer	copy inhibit
04	no	consumer	copy OK
08	50/15	consumer	copy inhibit
0C	50/15	consumer	copy OK
01	no	professional	rate n/i
0D	50/15	professional	rate n/i
81	no	professional	48 kHz
8D	50/15	professional	48 kHz
41	no	professional	44.1 kHz
4D	50/15	professional	44.1 kHz
C1	no	professional	32 kHz
CD	50/15	professional	32 kHz

The "rate n/i" entry under Misc means that the sample rate is not indicated in the status bytes and therefore will be determined by the clock rate of the transmitted signal. In practice, all receivers use the transmitted signal's clock rate to set sample rate, since the rates must match to allow reception of data. The 50/15 entry under emphasis means that 50/15 microsecond pre-emphasis (CD type) is used during recording and matching de-emphasis is used during playback.

A.9.3. Sample Address Code

The Dual Domain interface also generates a "sample address code" and transmits it on bytes 14-17. This code serves a similar function to a recording index counter on analog tape recorders. The code is reset to zero whenever a UTIL RESTORE menu command is issued. These bytes are not defined in any of the consumer versions of the interface.

A.9.4. CRCC Code

Byte 23 is a CRCC (cyclic redundancy check character) code. This code is computed by the Dual Domain interface for each status block and transmitted on byte 23. A receiving device could use this byte to detect errors occurring during transmission or recording and reproduction of the status bytes. This byte is not defined in the consumer versions of the interface.

A.9.5. Other Bytes

Any of the 24 bytes except bytes 14-17, which carry the automatically-generated sample address code, may have any hex value entered and thus transmitted until changed. Refer to the AES/EBU or EIAJ standards for the definition of these other bytes.

A.9.6. UTIL SERIAL-DSP Menu Commands

The UTIL SERIAL-DSP TRANSMIT menu command is used to set status bytes and cause them to be transmitted until further notice. See Figure A-11 for a typical screen when UTIL SERIAL-DSP TRANSMIT is selected. The rows near the bottom of the screen (visible as soon as the UTIL SERIAL-DSP menu command is executed) display the last-received and the last-transmitted set of status bytes. To change to another transmitted set of bytes, further select TRANSMIT and enter the desired hex codes at the bottom of the screen. The <Home> and <End> keys may be used to jump the cursor to the left or right end of the sequence of bytes. The horizontal arrow keys move the cursor one character at a time. The cursor is normally in Overtype mode, but can be toggled between Insert and Overtype modes with the <Ins> key. When the characters to be transmitted are entered as desired, pressing the <Enter> key causes this set of status bytes to be transmitted until further notice. Bytes 14-17 are controlled by the sample address code generator and cannot be set by keyboard entry.

The UTIL SERIAL-DSP RECEIVE command is used to update the upper line display of the screen. Each time the <Enter> key is pressed in UTIL SERIAL-DSP RECEIVE mode, the status bytes of the currently-received sample are acquired and placed into the display. If the device under test implements the sample address code (or if an XLR-XLR cable is connected directly from the AES output to AES input), bytes 14-17 will be seen as an increasing hexadecimal count as the generated sample address code increases.

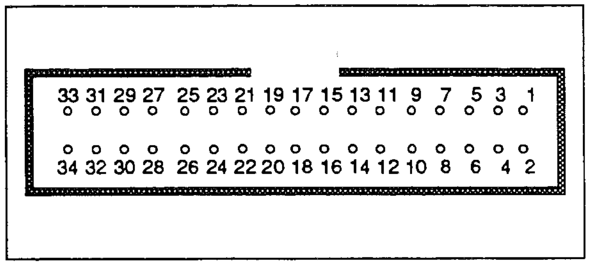


Figure A-12 Pin Numbering, Parallel Input and Output Connectors

To determine the status bytes required by a device under test before it will pass data it is often helpful to look at the status bytes 0-3 received from the device. They may then be entered into the UTIL SERIAL-DSP TRANSMIT command bytes 0-3. This will usually make a device under test accept the data.

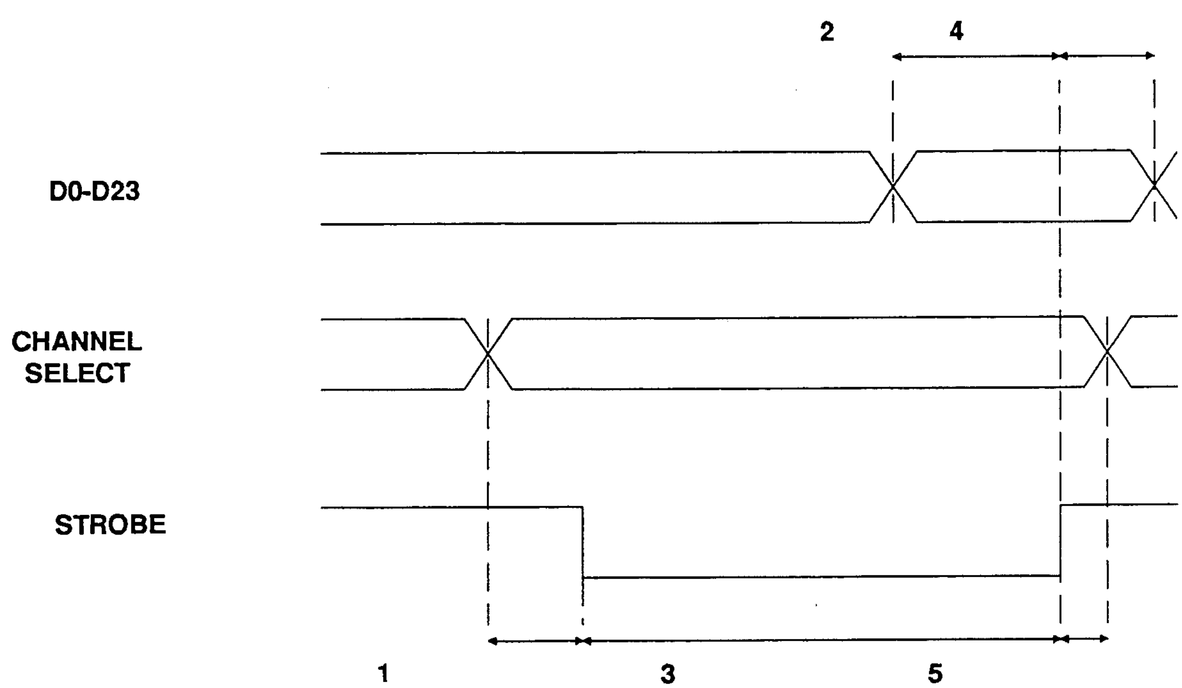
A.10. Parallel Digital Interface

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

A.10.1. Parallel Port General Information

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

There are 24 data lines, a channel select line, a strobe line and a sample clock line. The remaining pins are grounds, plus one +5 V power line. The table below lists the pinout of both input and output connectors. All lines are independent between the two connectors except the sample clock output, which is shared. Figure A-12 illustrates the physical orientation of the pins on the connectors when viewed from the rear of the instrument. Note the location of the key slot in the System One connector which prevents reverse installation of the mating cable connector. The cable connector key is a small raised portion in the center of one of the wide surfaces of the cable connector. It is highly recommended that mating connectors be used which have the corresponding key. This will prevent mis-insertion and reduce the risk of damage to the system or to the device under test.



1 CHANNEL SELECT-TO-STROBE SETUP TIME 22 nS MINIMUM
2 DATA-TO-STROBE SETUP TIME 33 nS MINIMUM
3 STROBE PULSE WIDTH 100 nS MINIMUM
4 STROBE-TO-DATA HOLD TIME 17 nS MINIMUM
5 STROBE-TO-CHANNEL SELECT HOLD TIME 8 nS MINIMUM

Figure A-13 Timing Diagram, Parallel Intput Port

Parallel Interface Connector Pin Assignments

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

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

All integrated circuits which connect directly to connector pins have been placed in sockets for easy replacement in the event of damage. All input and output data bits are processed through 74ACT670 Dual Port Register Files. Control lines are buffered with 74AC244 Octal Buffers.

A.10.2. Parallel Input

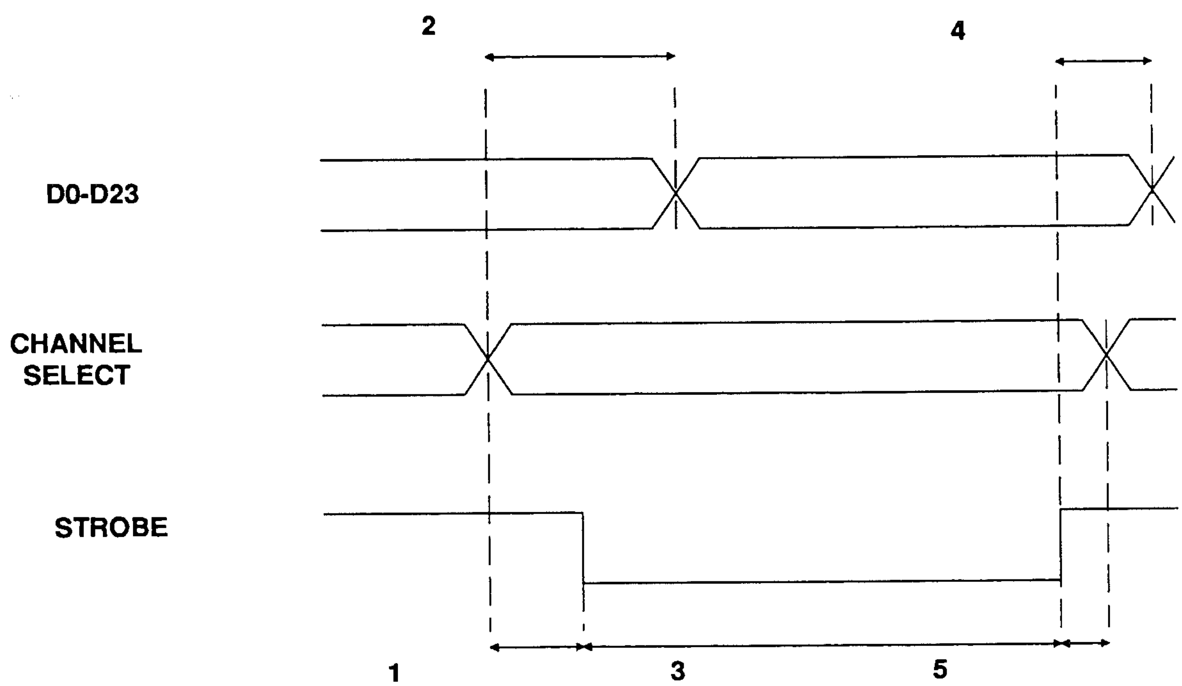
The parallel input accepts 24-bit data on the data lines and latches it when the strobe line makes a transition from low to high. The data is latched into the channel A input when the channel select line is held low and the strobe makes a transition from low to high. The data is latched into the channel B input when the channel select line is held high and the strobe makes a transition from low to high. Latching channel B data also generates an interrupt for the System One, causing it to read both channels of data just written to the latches. The channel select line must be held either high or low during the entire time the strobe line is low. A transition on the channel select line while the strobe line is low will cause an incorrect latch operation. Figure A-13 illustrates the correct operation of the parallel input port and identifies the timing specifications for the operations. Note that data must always be supplied to Channel B since its latch operation triggers the DSP to read the data. If a single channel interface is desired, it must therefore be configured as Channel B and the channel select line connected to +5V.

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

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

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

If some of the bits (especially the sign bit) are allowed to leak into the analog signal it will result in odd harmonic distortion (3rd, 5th, 7th, etc.) in the



1 CHANNEL SELECT-TO-STROBE SETUP TIME 0 nS MINIMUM
2 CHANNEL SELECT-TO-DATA VALID TIME 39 nS MAXIMUM
3 STROBE PULSE WIDTH 100 nS MINIMUM
4 STROBE-TO-NEW DATA VALID TIME 70 nS MAXIMUM
5 STROBE-TO-CHANNEL SELECT HOLD TIME 0 nS MINIMUM

Figure A-14 Timing Diagram, Parallel Output Port measured results. Excessive distortion is often a clue to examine the leakage paths in the device under test.

A.10.3. Parallel Output

Note that dither is also fully operational for the parallel digital output signal. See the dither section above on page A-13 for details. The output bits below the dither point are not truncated by the dsp. If they need to be zeroed it should be done in the wiring from the parallel port to the device under test.

The output port provides 24-bit data on the data lines at all times. The channel select line determines which channel's data appears on these lines. The read strobe line causes both channels of data to be updated, and signals the System One processor that another pair of data words are required. The data is buffered with a two stage FIFO buffer which guarantees that data will be available when needed rather than after the processor's service delay. As long as the maximum sample rate restriction is obeyed the two-stage buffer will never empty. Figure A-14 illustrates correct operation of the parallel output port and identifies timing specifications for the operations.

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

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

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

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

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

Burr-Brown PCM78 A-D Converter Fixture

Connector pin	function	System One Parallel In Connector function	pin
		Ground	1
		Ground	2
B1		MSB (bit 23)	3
B2		Bit 22	4
B3		Bit 21	5
B4		Bit 20	6
B5		Bit 19	7
B6		Bit 18	8
B7		Bit 17	9
B8		Bit 16	10
B9		Bit 15	11
B10		Bit 14	12
B11		Bit 13	13
B12		Bit 12	14
B13		Bit 11	15
B14		Bit 10	16
B15		Bit 9	17
B16		Bit 8	18
	JUMPER	Bit 7	19
		Bit 6	20
		Bit 5	21
		Bit 4	22
		Bit 3	23
		Bit 2	24
		Bit 1	25
		LSB (Bit 0)	26
GND		Ground	27
NC		Channel Select In	28
	JUMPER	Ground	29
DV		Data Strobe In	30
NC		+5 Volts	31
TO CONVERT COMMAND (BNC)	910 Ω	Sample ClockOut	32
	0.001 uF	Ground	33
	SCHOTTKY	Ground	34

All pins labeled Ground on System One connector should be grounded at the Burr-Brown board. Bits 0 through 7 on System One connector must be grounded.

Figure A-16 Connection Diagram, Burr-Brown PCM78 A-D to DSP Parallel Digital Input

Crystal CDB 5326 A-D Converter Fixture

Crystal Connector pin	function	System One Parallel In Connector function	pin
		Ground	1
		Ground	2
8	D15 (MSB)	MSB (bit 23)	3
10	D14	Bit 22	4
12	D13	Bit 21	5
14	D12	Bit 20	6
16	D11	Bit 19	7
18	D10	Bit 18	8
20	D9	Bit 17	9
22	D8	Bit 16	10
24	D7	Bit 15	11
26	D6	Bit 14	12
28	D5	Bit 13	13
30	D4	Bit 12	14
32	D3	Bit 11	15
34	D2	Bit 10	16
36	D1	Bit 9	17
38	D0 (LSB)	Bit 8	18
		Bit 7	19
		Bit 6	20
		Bit 5	21
	JUMPER	Bit 4	22
		Bit 3	23
		Bit 2	24
GND		Bit 1	25
		LSB (Bit 0)	26
		Ground	27
6	X (L/R)	Channel Select In	28
		Ground	29
4	LCLK	Data Strobe In	30
	NC	+5 Volts	31
	NC	Sample ClockOut	32
		Ground	33
		Ground	34

Connect U4 pin 16 (VD+) on Crystal board to point X (pin 6) for channel control.

Pins 7, 9, 11, 13, 15, 17, 19, 21, 23, 25, and 27 on Crystal connector must be grounded.

All pins labeled Ground on System One connector must be grounded. Bits 0 through 7 on System One connector must be grounded.

Figure A-15 Connection Diagram, Crystal Semiconductor CDB 5326 A-D Converter Board and DSP Parallel Input

dbx A-D Converter Fixture

dbx Connector pin	dbx Connector function	System One Parallel In Connector function	System One Parallel In Connector pin
44	Ground	Ground	1
43	Ground	Ground	2
1	MSB	MSB (bit 23)	3
2	Bit 18	Bit 22	4
3	Bit 17	Bit 21	5
4	Bit 16	Bit 20	6
5	Bit 15	Bit 19	7
6	Bit 14	Bit 18	8
7	Bit 13	Bit 17	9
8	Bit 12	Bit 16	10
9	Bit 11	Bit 15	11
10	Bit 10	Bit 14	12
11	Bit 9	Bit 13	13
12	Bit 8	Bit 12	14
13	Bit 7	Bit 11	15
14	Bit 6	Bit 10	16
15	Bit 5	Bit 9	17
16	Bit 4	Bit 8	18
17	Bit 3	Bit 7	19
18	Bit 2	Bit 6	20
20	Bit 1	Bit 5	21
22	LSB (Bit 0)	Bit 4	22
31	Ground	Bit 3	23
32	Ground	Bit 2	24
33	Ground	Bit 1	25
34	Ground	LSB (Bit 0)	26
30	Ground	Ground	27
NC	JUMPER	Channel Select In	28
24	Ground	Ground	29
25	INV Clock Out	Data Strobe In	30
NC		+5 Volts	31
NC		Sample ClockOut	32
45	Ground	Ground	33
46	Ground	Ground	34
29	Output Enable	Ground	34
33	Ground	Ground	34
19	2's complement	Ground	34

Figure A-18 Connection Diagram, dbx A-D Converter Board and DSP Parallel Input

dbx D-A Converter Fixture

dbx Connector pin	dbx Connector function	System One Parallel Out Connector function	System One Parallel Out Connector pin
7	Ground	Ground	1
8	Ground	Ground	2
50	Inverted MSB (17)	MSB (bit 23)	3
49	Bit 16	Bit 22	4
48	Bit 15	Bit 21	5
47	Bit 14	Bit 20	6
46	Bit 13	Bit 19	7
45	Bit 12	Bit 18	8
44	Bit 11	Bit 17	9
43	Bit 10	Bit 16	10
42	Bit 9	Bit 15	11
41	Bit 8	Bit 14	12
40	Bit 7	Bit 13	13
39	Bit 6	Bit 12	14
38	Bit 5	Bit 11	15
37	Bit 4	Bit 10	16
36	Bit 3	Bit 9	17
35	Bit 2	Bit 8	18
34	Bit 1	Bit 7	19
33	LSB (Bit 0)	Bit 6	20
NC		Bit 5	21
NC		Bit 4	22
NC		Bit 3	23
NC		Bit 2	24
NC		Bit 1	25
NC		LSB (Bit 0)	26
21	Ground JUMPER	Ground	27
NC		Channel Select In	28
22	Ground	Ground	29
		Data Strobe In	30
NC		+5 Volts	31
NC		Sample ClockOut	32
5	Ground	Ground	33
6	Ground	Ground	34

Figure A-17 Connection Diagram, dbx D-A Converter Board and DSP Parallel Output tor and should be mounted on the connector itself. Proportionately lower resistances may be required when the cable impedance is significantly lower.

A.10.4. Parallel Port Connection Examples

Figure A-16 shows the wiring required to connect a Burr-Brown PCM 78 A/D converter evaluation board to the parallel input. This is a single channel device so it has been connected to the channel B input. The 16 most significant data lines of the DSP are connected to the 16 data lines of the evaluation board. The channel select line is connected to +5V and the lowest 8 DSP data input lines are connected to ground. This evaluation board requires a start conversion pulse which is generated from the DSP sample clock output. The converter requires that this pulse be a minimum of 50 ns but also will not begin conversion until the pulse returns high. The diode/resistor/capacitor network shown provides this one-shot action. When the conversion is complete, the data is latched into the input using the data strobe line. The converter data format must be set to two's complement using the jumper on the evaluation board.

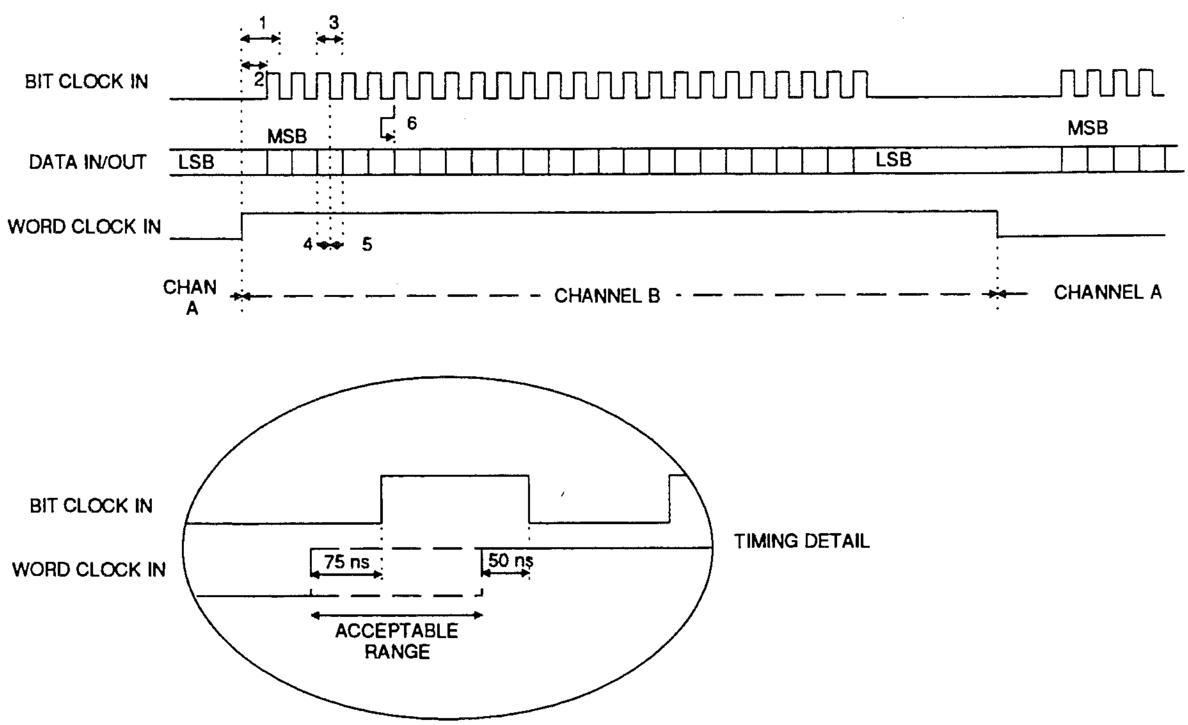
Figure A-15 shows the wiring required to connect a Crystal Semiconductor CDB 5326 A/D converter evaluation board to the parallel input. This is a dual channel device so it uses both channel A and B inputs. The 16 most significant data lines of the DSP are connected to the 16 data lines of the evaluation board. The lowest 8 DSP data input lines are connected to ground and the DSP channel select line is wired to the evaluation board's channel select output. The converter generates its own start conversion command. The conversion-complete pulse is used to latch data into the parallel inputs.

Figure A-18 illustrates the required connections between the DSP and the dbx A/D converter board. This is a single channel device so it has been connected to the channel B input by wiring the channel select line to +5V. The converter generates a 20-bit word which is connected to the 20 most significant bits of the parallel input. The lowest 4 input bits are tied to ground. The converter generates its own start conversion command. The conversion-complete pulse is used to latch data into the parallel inputs. A logic inverter is required between the word clock output (pin 25) of the dbx board and the data strobe input (pin 30) of the DSP connector.

Connecting the companion D/A converter board from dbx is equally easy, as shown in Figure A-17. This board consists of a Burr-Brown 18-bit D/A converter, a deglitcher, and a simple reconstruction filter. The board is ordinarily used to turn the A/D converter's output into an analog signal for evaluating the A/D. However, it is representative of connecting a parallel input D/A converter to the Audio Precision DSP parallel output. The most significant 18 bits of the parallel output are connected directly to the converter's 18 inputs. The unused bits from the parallel output are simply left unconnected. The channel select line is tied high to put the converter on the channel B output but either channel will work equally well. Since the D/A is straight binary format, not two's complement, a logic inverter must be used between the MSB (pin 3) output of the DSP unit and the MSB input (pin 1) of the dbx board.

A.11. Rear Panel Serial Connector

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



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

A-19 General Purpose Serial Input-Output Port, Timing Relationships

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

A.11.1. General Purpose Serial I/O

The information in this section is provided for the user who plans to construct his own logic interface circuitry in order to interconnect System One Dual Domain and digital devices with serial interfaces. Alternatively, the SIA-322 Serial Interface Adapter is an accessory product to System One Dual Domain which permits interconnection to serial interface devices without the necessity of designing or constructing special interface circuitry. Switches on the SIA-322 front panel permit selection of word length from 8 to 24 bits, single or dual-channel operation, data word length from 8 to 24 bits with MSB or LSB first, right or left justification, and 0 or 1 padding, selectable word clock transition within the data frame with bit-wide or word-wide clock, TTL or CMOS logic family support in either polarity and independent selection of bit clock and word clock source. For more information on the SIA-322, contact Audio Precision or your Audio Precision International Distributor.

The general purpose serial port consists of six signals, three for the input port and three for the output port. The serial port shares hardware with the AES/EBU-SPDIF/EIAJ interface, so the two may not be used simultaneously. The selection between the general purpose serial interface, the AES/EBU interface, and the SPDIF/EIAJ interface is made with the UTIL SERIAL-DSP command on the command line menu. The hardware word width of the general purpose serial port is always 24-bit. Unused bits fed to the DSP should be zeroed.

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

See Figure A-19 for timing diagrams of the general purpose serial port. Serial input data is clocked on the falling edge of the bit clock. The most significant bit is always first. The clock signal must be 48 times the data sample rate providing 24-bit data for each channel. This may be obtained with a gated clock at 64 times the sample rate which lies idle for 8 bits of the 32-bit frame as illustrated in the figure. The maximum word strobe frequency is 48 kHz, the minimum is 32 kHz. The maximum bit clock frequency is 3.072 MHz (64 times the 48 kHz sample rate).

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

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

The signals required to clock data at the serial interface may be created from the 64x master clock on the serial connector. However, since two 24-bit intervals are to be obtained from a 64x clock there will be idle periods in the resulting clocks. Care should be taken to insure noise-free and glitch-free signals on the cabling between the serial interface connector and the device being driven. The interface hardware is extremely fast and will respond to very short duration pulses.

A.11.2. AES/EBU Input Word Strobe Output

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

A.11.3. Sampling Clock Outputs

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

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

A.11.4. Auxiliary Inputs and Outputs

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

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

A.11.5. External Sync Input

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

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

For testing of interfaces at non-standard sample rates, an external square wave generator or the System One analog generator sync output may be routed to the external sync input. One effective setup is to use an external function generator with its VCF (voltage-controlled frequency) input driven from one of the DC variable outputs of the DCX-127 multifunction unit. Sweeping the DCX DCOUT as SOURCE-1 through the appropriate DC voltage range then results in sweeping the function generator squarewave frequency through a desired range, while measuring performance of the digital interface. If the System One analog generator is used, sharp and brief discontinuities may occur at 96 kHz and other integer values. This is due to the 16-bit frequency tuning DACs in the analog generator being programmed one byte (one-half word) at a time.

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

A.11.6. External Trigger Output

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

B. FFTGEN.DSP AND FFTSLIDE.DSP PROGRAMS

Two .DSP programs are furnished for general-purpose waveform acquisition, spectrum analysis via FFT (Fast Fourier Transform), and time domain waveform display (digital storage oscilloscope mode). These programs are FFTGEN.DSP and FFTSLIDE.DSP (FASTEST.DSP and MLS.DSP also use FFT analysis techniques). FFTGEN.DSP includes internal sinewave generation, the ability to generate arbitrary waveforms from downloaded .WAV files, plus spectrum averaging. FFTSLIDE.DSP lacks those features, but includes more powerful and flexible triggering (including pre-trigger), plus the ability to perform an FFT upon a specified portion of an acquired record. Acquired signals from either program may be saved to computer disk for later down-load to the DSP and further analysis.

Figure A-1 on page A-4 is an example of spectrum analysis via FFTGEN.DSP of the residual distortion and noise of the System One analog generator and analyzer at 1 kHz.

Figure A-2 on page A-4 is an example of waveform display of System One's analog sine burst signal.

The spectrum analysis and waveform display portions of both FFTGEN.DSP and FFTSLIDE.DSP are batch mode programs. This means that they first acquire an incoming signal (often in response to a trigger event), then perform the necessary processing and transmit the results to the computer for display. Once acquired, a signal is stored in the DSP memory until the next acquisition, until ac mains power is turned off, until another .DSP program is loaded, or until a previously-saved waveform is downloaded via LOAD WAVEFORM. An acquired signal may be saved to disk via SAVE WAVEFORM for later download and further analysis. The FFT parameters may be changed and another FFT performed without re-acquiring the signal, or the user may change at will between waveform display and spectral display by loading appropriate tests (which have the same .DSP program attached) without re-acquiring the signal. Waveform data may be passed between the two FFT programs by saving it to disk, then loading it into the other program.

> The expression "loading" a DSP program is used frequently throughout this manual. In fact, NAMES PROGRAM is the specific command required to download a .DSP program from computer disk to DSP unit. When a .TST file is saved to disk after using the NAMES PROGRAM command, the DSP program will also be automatically downloaded each time the .TST file is loaded thereafter (unless the DSP program is already in place from the previous test).

B.1. General Operation

Operation of either FFTGEN.DSP or FFTSLIDE.DSP involves the following main steps:

1. Select the appropriate signal source (DC-coupled BNC connectors, balanced and autoranged analyzer input, analyzer signal after all processing, etc.).

2. Select an appropriate trigger source, or select trigger OFF to trigger on <F9> (in effect, free-running trigger).

3. For waveform display, select DSP TIME as SOURCE-1 on the Sweep (F9) Definitions panel. Select start and stop times appropriate to display a few cycles of the signal period. Select DSP AMP1 or AMP2 as DATA-1 if only one channel is to be displayed. If two channels are desired, select the relationship between DSP channel and DATA-1 and DATA-2 as desired. Select DATA units such as Volts, with appropriate graph top and bottom values for the expected peak-to-peak signal amplitude and

a LIN vertical display. LOG display units such as dBV will result in distorted waveshapes. For typical ground-centered symmetrical signals, the graph top and bottom values will be normally numerically the same, with the graph bottom value negative.

4. For spectrum analysis, select DSP FREQ as SOURCE-1. Select start and stop frequencies for the spectrum to be displayed, not exceeding the folding frequency (one-half the sampling rate) for the upper frequency limit. Select DSP AMP1 and/or DSP AMP2 as DATA-1 and DATA-2 for one or two channel displays. Select DATA units such as Volts with a LOG display or a dB unit with LIN display.

5. Set an appropriate # STEPS value for display of the waveform or spectrum. This parameter determines how many selected points from the total waveform or spectrum will be sent by the DSP module to the computer for display. The typical S1.EXE default value of 30 steps is inadequate for most waveform or spectral displays. Values of 200 to 500 points are the typical range for display clarity, with graphing speed favoring the smaller numbers. If the # STEPS value is greater than the number of waveform samples or FFT spectral lines for the selected horizontal span in the DSP memory, some adjacent points in computer memory will have the same value. If the # STEPS is less than the number of FFT spectral lines for the frequency span displayed, the DSP will select the highest value in each group of frequencies as it sorts the larger number of FFT spectral data values into a smaller number to send to the computer, so as not to miss spectral peak values. In waveform display mode (SOURCE-1 DSP TIME), the DSP will round to the nearest time value as it sends points to the computer if NORMAL display mode is selected on the DSP panel. If INTERPOL is selected, the DSP performs an interpolation across 15 data values centered on the point currently requested. If Peak is selected, the DSP sends the largest positive or negative value since the last sample. If MAXIMUM is selected, the DSP sends the absolute value of the largest positive or negative peak. See below for more information on these several waveform display modes.

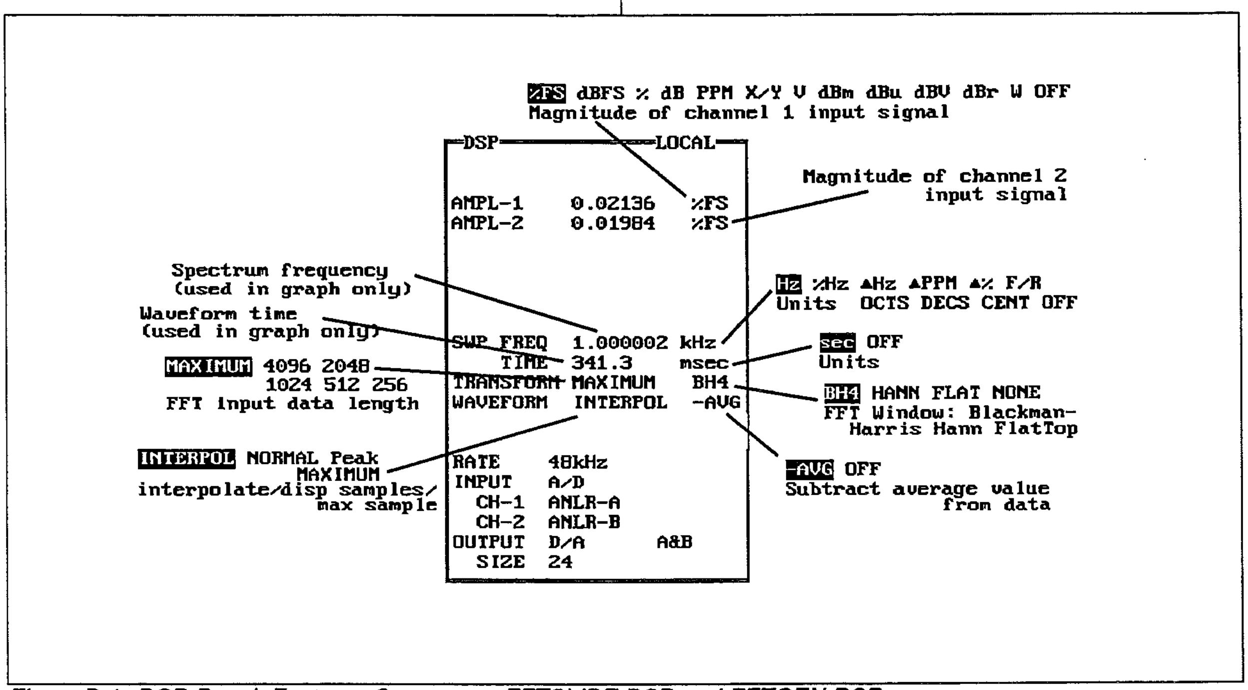


Figure B-1 DSP Panel, Features Common to FFTSLIDE.DSP and FFTGEN.DSP

6. Press <F9> to acquire, convert, and display the signal. The messages "Triggered and Acquiring" and "Transforming" should be seen at the top of the graph until the data is transmitted to the computer and displayed. If the message "Waiting for Trigger" is displayed during a period when you believe a trigger signal to be present, there is probably an error in selection of trigger source.

7. In general, you need not acquire again (<F9>) in order to change from waveform to spectral display or make other changes in the display of information. The acquired data remains in DSP memory and will be re-transformed according to the then-current DSP and SWEEP (F9) DEFINITIONS panel settings each time the <F6> key is pressed. Several .TST files can be set up with the various desired time and frequency display parameters, all with the same FFT program named. The user may then select among these several tests and press <F6> to cause a new display of the data according to the current test panel settings.

The following sections provide more detail on FFT program operation.

B.2. Features Common to Both Programs

Figure B-1 shows the DSP panel with only the fields visible which are common to both FFTGEN.DSP and FFTSLIDE.DSP.

B.2.1. FFT Real-Time Displays

The two displays at the top of the FFT panel are real-time displays of input amplitude of the two DSP channels. Their principal purpose is to help avoid overload of the input A/D converters. When signal is acquired via the analog analyzer and autoranging is used in the analog analyzer, overload will not normally be a problem.

When displaying signals acquired via the analog analyzer in analog units, the data will be scaled according to the ranges in use at the time of the last <F9> operation. If the signal amplitude or the signal itself changes, these real-time displays will be in error. There is no problem in graph displays. Therefore, only dBFS or %FS units should be used on the FFT real-time displays to avoid erroneous readings and possible overload. Zero dBFS or 100 %FS should never be exceeded. When analog signals are being acquired via the DSP BNC connectors, analog display units may be used on the FFT panel since no ranging is involved.

When "-AVG" is selected on the WAVEFORM line of the FFT panel, the real-time displays are based on one-half the peak-to-peak value. This makes them insensitive to dc voltages and allows readings of low level signals. When the -AVG mode is OFF, the display is based on the maximum (positive or negative) peak amplitude and dc values will affect the readings.

B.2.2. FFT Controls

Both FFTGEN.DSP and FFTSLIDE.DSP have fields to select record length (FFT input data length), FFT windows, waveform interpolation, and a subtract-the-average mode similar in effect to AC coupling.

B.2.2.1. Record Length and Resolution

Record length (FFT input data length) determines the number of waveform samples which will be used for the FFT computation. The resulting FFT will have half as many frequency points (bins) as the number of waveform samples from which it is computed. The equivalent frequency resolution is obtained by dividing the sample rate by the number of waveform samples. For example, at a 48 kHz sample rate with a 4096 sample record length, the resulting FFT will consist of 2048 (4096/2) spectral lines (bins) evenly spaced from zero Hz to the Nyquist frequency (1/2 sampling rate). The resulting bin width is approximately 11.7 Hz (48,000/4,096). With maximum memory (MEM option required in SYS-200 family), the MAXIMUM record length selection results in 16,384 samples with either FFT program. A spectrum will then be produced with 8,192 frequency bins and frequency resolution of

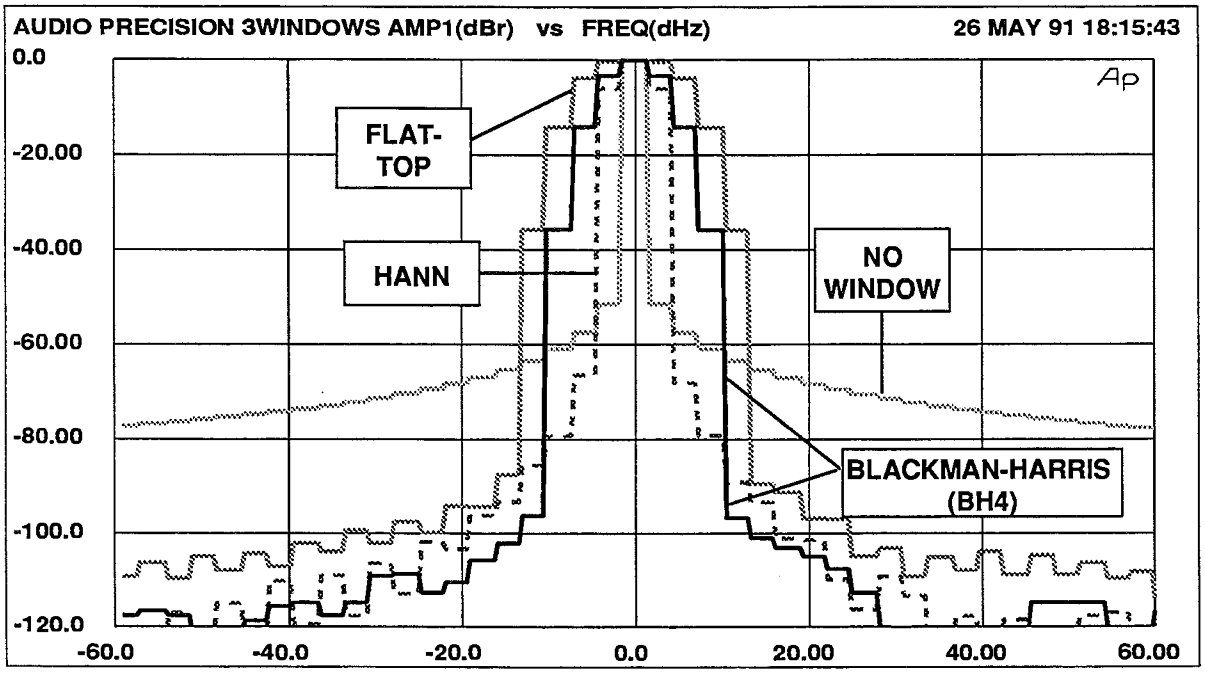


Figure B-2 Response of BH4, Hann, Flat, and No Windows. Horizontal Axis Calibration is Delta-Hz from Signal Frequency, Bin Width is 2.93 Hz

2.93 Hz (48,000/16,384). The actual number of samples acquired into DSP memory when <F9> is pressed depends upon which of the two DSP programs is loaded; see their individual descriptions below for discussion of the differences.

B.2.2.2. Window Functions and Selectivity

FFT algorithms process a series of signal samples called a data record. With FFTGEN and FFTSLIDE, this is always a power of 2 such as 512, 1024, etc. Since this data record starts at some point in time and ends a short time later, the FFT algorithm has no information about the signal outside this time region. The FFT algorithm inherently assumes that the data outside this data record is an infinite series of replicas of the data record. If the data record being transformed does not contain exactly an integral number of cycles of signal, the value at the end of the record will not match up with the beginning value of the next record. The resulting sharp discontinuities in the waveform appear as large amounts of high frequency energy in the transformed result.

To alleviate this problem, a "window" may be applied to the data. The idea behind a window is to gradually taper the data at both ends of the record toward zero so that it will always make a smooth transition with the following and preceding repetitions of the record. This is accomplished by multiplying each point in the data record by a mathematical function which is near unity (1.000) in the center of the data record and small at the ends of the record. The simplest such function is a cosine wave with an added DC offset so that its negative peaks just reach zero. After multiplication by the window function the data record goes to zero at the ends and so smoothly meets each data record on either side of the one being transformed. However, multiplying the data by the window function does alter the spectrum of the original sinewave. As might be expected by visualizing the envelope of the repeating windowed data record, the spectrum of the original sinewave has been broadened. However, the spurious high frequency components produced by the sharp discontinuities have been eliminated.

The generic term window comes about because it restricts the view of the FFT to the central portion of the data record in much the same way that a window restricts the view of a person looking through it. There have been an endless variety of windows developed which trade off the spread in the spectral peak versus the ultimate attenuation of the spurious energy created by the ends of the data record. The sharper the roll-off in the skirts, the wider the peak must be in the passband. The bandwidth of the peak will be a specific number of bins for any given window. Increasing the length of the data record will reduce the bin width and therefore reduce the bandwidth in Hertz.

The raised cosine window described above is called the HANN window after its inventor, Austrian meteorologist Julius von Hann. (It is often incorrectly called a Hanning window due to confusion with the "Hamming" window, named after its inventor Richard Hamming.) The Hann window is provided as one of the choices in FFTGEN and FFTSLIDE. The second window supplied is the -92 dB sidelobe 4 term Blackman-Harris window (BH4), named after its developers R. B. Blackman and F. J. Harris. It has much steeper skirts, resulting in the ultimate attenuation of sidelobes by 92 dB. It gives a moderately flat top approximately 3 bins wide. A third window, named FLAT on the panel and often called Flat-Top, has poorer skirt selectivity than either Hann or BH4 but provides an extremely flat top across several bins in order to produce essentially zero amplitude error regardless of the signal position in the bin. The final choice in the programs is labeled NONE and performs no windowing of the data; this is often called a rectangular window. Spectral analyses of a 1 kHz sinewave using these windows are shown in Figure B-2. The horizontal axis is delta-Hz (deviation from the exact signal frequency). Sample rate was 48 kHz and transform length MAXIMUM, producing bin width of approximately 2.92 Hz.

The selection of window affects the amplitude measurement accuracy with the FFT programs. Bin centers are at frequencies exactly determined by the sample rate and record length. A signal component being measured, however, may generally fall at any point within a bin. The amplitude measurement uncertainty is thus determined by how far from bin center the signal frequency falls and by the shape of the window function across the center bin. Worst-case errors (window attenuation at the edge of the bin) are essentially zero for the FLAT window, approximately 0.8 dB for the BH4 window, 1.5 dB for Hann, and about 4 dB for NONE (no window).

For an excellent technical discussion of windows and their characteristics see F. J. Harris, "On the use of Windows for Harmonic Analysis with the Discrete Fourier Transform", Proc. of the IEEE, Vol. 66, No. 1, Jan 1978, pp. 51-83.

B.2.2.3. Resolution and Display Steps

As noted above, the frequency resolution of the FFT depends upon the record length (number of samples) upon which the FFT was performed. This resolution determines the FFT's ability to separate the various spectral components of the signal. The graphically displayed resolution also depends upon the # STEPS parameter in the SOURCE-1 area of the SWEEP (F9) DEFINITIONS panel and upon the graphics display system in use. If 0 Hz is selected as START, the Nyquist frequency (1/2 the sample rate) is selected as STOP, and # STEPS is set to half the FFT input data length, the number of points sent by the DSP to the computer display will exactly match the intrinsic FFT resolution. For example, with a 48 kHz sample rate the Nyquist frequency is 24 kHz. If a 512 sample record length is chosen, the basic FFT spectrum computed will consist of 256 spectral lines or bins. Select 0 Hz START, 24 kHz STOP, and # STEPS 256. Each data point will then correspond to an FFT spectral line (bin).

With record lengths longer than 1024 points (FFTs with more than 512 lines) and correspondingly larger # STEPS, the horizontal resolution of the computer display system will also act as a limiting item. VGA, EGA, and CGA monochrome all have 640 pixel resolution across the full screen width. About 81% to 88% of this value is available for graphic data after space is taken up for borders and calibration at the left or both sides. Thus, on the order of 520 to 540 pixels are available horizontally. Greater FFT resolution and higher # STEPS will result in more than one FFT bin being plotted in the same pixel column. This may still be useful if the ultimate graph is to be reproduced on a high resolution laser printer or plotter via the PLOT or POST utilities, which are not limited by display system resolution.

If the # STEPS parameter is set to a smaller number than the actual number of FFT lines (and if no sweep table is in use), DSP program software will select the highest amplitude in each group of lines as it sorts them into the smaller number of points to be displayed (unless TABLE is ON on the SWEEP DEFINITIONS panel). This "peak picking" operation assures that no signals will be missed even when the displayed resolution is less than the FFT resolution. For example, if a 2048 sample record length is chosen, a 1024 line FFT will result. If the # STEPS parameter is set to 256, each set of 4 adjacent FFT lines will be examined and the highest amplitude value of each set transmitted to the computer for display.

It is possible to use a LOG frequency axis for FFT spectral analysis data, though the data itself is inherently linear on the frequency axis. For a LOG display, the START frequency must be a positive non-zero value.

It is often convenient to select the # STEPS parameter and STOP frequency to produce round numbers for convenient use of the graphic cursor feature. For example, with a 0 Hz START frequency, 20 kHz STOP, LIN display, and # STEPS 500, the displayed data points will spaced 40 Hz. The graphics cursor will then move in individual 40 Hz steps when the horizontal arrow keys are used alone, 200 Hz steps with <Ctrl> arrow (5 times magnification), and exact one kHz steps when the <Shift> key is held down while a horizontal arrow key is pressed (25 times magnification). If a mouse is used, rolling the mouse horizontally would move the cursor through every point. Holding down the left mouse button while rolling would move through every fifth point (200 Hz steps), and holding down the right button while moving would move the cursor through every 25th point (one kHz steps). Note that this multiplication effect of the <Shift> key is effective only via the horizontal arrow keys which are imbedded in the numeric key pad area of the keyboard, and not when using the separate horizontal arrow keys found on many 101-key keyboards.

B.2.2.4. Noise Amplitude

When a spread-spectrum signal such as noise is analyzed via FFT, the amplitude of the noise depends upon the effective bandwidth of an FFT line or bin. Using a longer record length (more samples) produces a higher resolution FFT (narrower bins) and lower noise amplitude within each bin. The common combination of 48 kHz sample rate and MAXIMUM record length produces 2.93 Hz resolution. If perfect white noise were being measured, the noise measurement bandwidth of any bin would thus be (2.93/22,000) less than the common 22 kHz audio bandwidth often specified for noise measurements. This bandwidth ratio will cause the noise voltage in each bin to be 38.8 dB lower (10 log bandwidth ratio) than an integrated 22 kHz bandwidth measurement would provide, if the noise were truly evenly distributed across the spectrum.

B.2.2.5. FFT Operating Speeds

The time required to acquire the signal may be calculated by dividing the memory length to be filled by the sample rate. Thus, if 8192 samples are to be stored at a 48 kHz sampling rate, the acquisition will take about 170 milliseconds following the trigger. Note, however, that FFTSLIDE.DSP and FFTGEN.DSP behave differently with respect to length of signal acquired. FFTSLIDE.DSP always fills the entire memory (except at the 1 kHz sample rate), so acquisition times with FFTSLIDE are independent of the record length selected. FFTGEN.DSP only acquires the number of samples necessary to fill the record length selected on the DSP panel when <F9> is pressed. The time required for the DSP chip to actually perform the FFT is typically quite small compared to the time to transmit the results to the computer. The transmission time from DSP modules to computer is proportional to the number of data points (# STEPS + 1) and depends strongly on the speed of the personal computer. The table below shows the approximate times for acquisition (assuming a 48 kHz sample rate and use of FFTGEN.DSP, which acquires only the record length selected) and FFT computation (windowing, transforming, magnitude and phase) for several different record lengths being transformed. The additional time for transmission to the computer plus units conversion and graphing is approximately 2.2 seconds for 512 points plotted on a 20 MHz 80386-based computer with 80387 math coprocessor and color VGA display system. All other speeds shown are independent of the computer type being used.

Record Size	Acquire Time	FFT Computation
16k	340 ms	600 ms
8k	170 ms	315 ms
4k	85 ms	165 ms
2k	43 ms	95 ms
1k	21 ms	50 ms

B.2.2.6. Dual Channel Operation

The DSP hardware, memory, and FFT programs are both capable of two-channel operation. Two independent A/D converters permit simultaneous acquisition of two analog signals, even at maximum sample rates. The digital interfaces are also of two-channel architecture. The input signal selection capability of the analog interface permits assigning a signal to one DSP channel and another related or completely independent signal to the other channel. The digital interface permits selecting the "A" digital signal channel as one signal and the "B" signal as the other. Both channels will be acquired, processed, and displayed when the <F9> key is pressed. Both will be re-processed and displayed when the <F6> key is pressed. See the Split Screen Effects section on page B-10 below for suggestions on how to display dual channel data.

It is also possible to acquire a signal into only one channel and then later acquire another signal into the other channel. If one of the two channel input selections near the bottom of the DSP panel is set to NONE, no signal will be acquired into that channel when <F9> is pressed. Conditions or devices may be changed to another desired test condition. Then, the originally-used input channel can be set to NONE and the desired signal source selected at the channel where no signal has yet been acquired. A second operation of the <F9> key will acquire data into that channel but not over-write the data in the channel with NONE selected for input. The data in both channels will then be processed, transmitted to the computer, and displayed.

Some conditions and precautions must be observed when acquiring signal into the two channels at different times:

- Any acquired signals may be lost when AC mains power to System One is turned off or when another DSP program is named (changing from FFTGEN.DSP to FFTSLIDE.DSP, for example). Thus, the two signals must normally either be acquired during the same testing session, or saved to disk using the SAVE WAVEFORM capability and later combined via LOAD WAVEFORM.

- Signal in both channels will be processed every time <F6> or <F9> is pressed, ***according to the present settings on the DSP panel.*** It is thus not possible to use different windows or different record lengths for the two channels. ***More important, both sets of data must be acquired at the same sample rate.*** Processing will be done according to the rate selected at the time of pressing (<F6> or <F9>). If the data in one channel had been acquired at a different rate, it will be erroneously transformed or displayed with a resulting frequency error directly proportional to the difference in the two rates.

B.2.3. Waveform Display

The waveform interpolation (INTERPOL, NORMAL, Peak, or MAXIMUM) and "AC coupling" (-AVG vs OFF) fields affect only waveform dis-

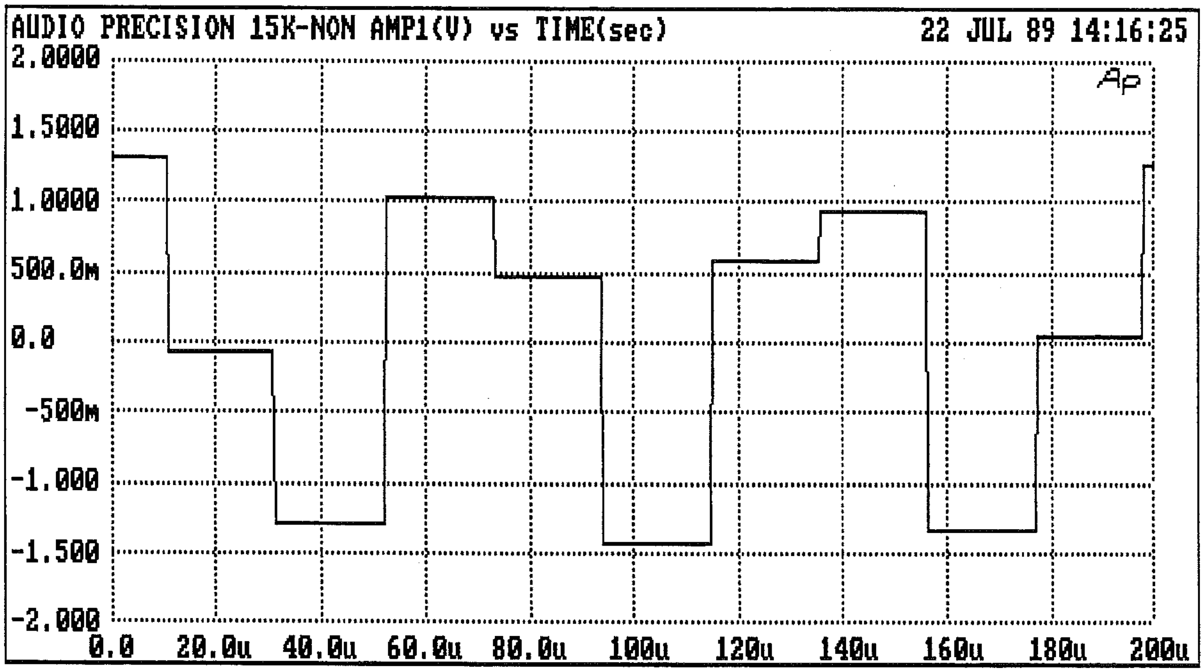


Figure B-3 15 kHz Sinewave Acquired at 48 kHz Rate, Waveform Interpolation OFF

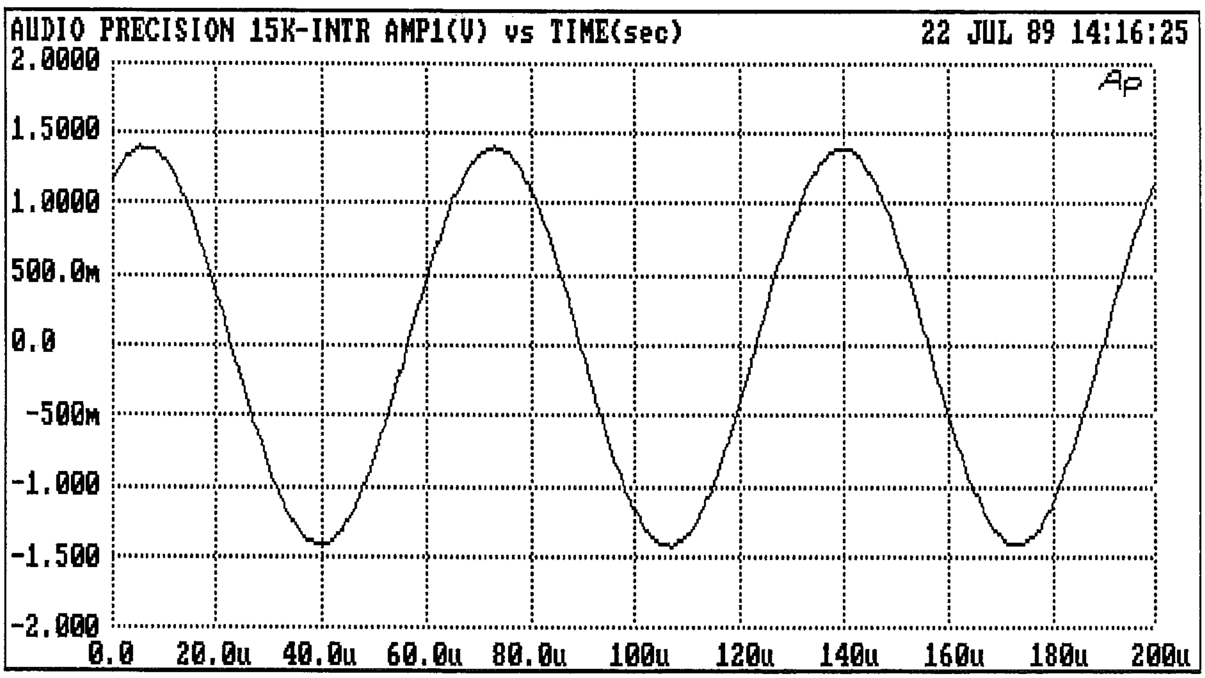


Figure B-4 15 kHz Sinewave Acquired at 48 kHz Rate, Waveform Interpolation ON play, not spectral displays (FFT). This contrasts with MLS.DSP where a similar field affects both time and frequency domain displays.

When waveform interpolation is in the NORMAL condition, the DSP will send to the computer the value of the sample closest to the time being graphed. This may produce adequately faithful reproduction of waveforms when the signal frequency being displayed is low relative to the sample rate. Each signal cycle is then represented by many samples and the display will be relatively smooth. When the signal frequency is high relative to the sample rate, however, each cycle is represented by only a small number of points. A 20 kHz sine wave acquired at a 48 kHz sample rate, for example, has an average of only 2.4 points per cycle. When S1.EXE software "connects the dots", the result does not look like a sine wave. See Figure B-3 for a graph of a 15 kHz sinewave displayed with a high # STEPS while using NORMAL mode.

When INTERPOL is selected instead of NORMAL, an additional software routine in the DSP unit computes what the signal waveform must have looked like to produce those samples. It assumes that the signal had been band-limited by an anti-alias (low-pass) filter before sampling. Given a sufficiently large # STEPS, the waveform will be faithfully reproduced. Figure B-4 shows the same waveform acquisition as Figure B-3, but with the INTERPOL selection.

The Peak choice on the WAVEFORM line selects a DSP routine which sends the largest positive or negative value since the previous plotted point to the computer, preserving the sign. The principal purpose of Peak display mode is to avoid the risk of completely missing a signal due to an unfortunate combination of signal frequency, sample rate, and the displayed # STEPS and start and stop times (graphic aliasing). See the graphic aliasing discussion below for more details.

The MAXIMUM selection on the WAVEFORM line returns the absolute value of the largest peak value since the previous plotted point. This permits dB units to be used for the vertical display, which is helpful is measuring the output of compressors and wide-dynamic-range signals.

The -AVG mode performs a computation whose results are similar to AC coupling on an oscilloscope. This function computes the average value of all the samples in the record and subtracts that average from each value in the record. The effect is to remove any DC offset present in the signal when the signal is displayed. Since this is not true AC coupling, care must be taken during the original acquisition of signals with a DC component that neither positive nor negative peak excursions exceed the full-scale range of the input A/D converters. When acquiring signals through the System One analyzer channel A or channel B inputs, their hardware AC coupling and autoranging should prevent overload. With a signal fed directly to the BNC connectors on the DSP unit, however, the user is responsible for control of the external signal amplitude. The real-time amplitude display fields at the top of the DSP panel indicate absolute input signal peak amplitude and should never be allowed to exceed 100%FS (0 dBFS).

B.2.4. Sweep (F9) Panel Fields

DSP measurement choices are available at both the DATA-1 and DATA-2 fields for plotting onto graphs. DSP horizontal axis choices are available at both SOURCE-1 and SOURCE-2.

B.2.4.1. Data-1 and Data-2 Choices

At DATA-1 and DATA-2, the choices are AMP1 and AMP2. These are the amplitudes for the two DSP channels (CH-1 and CH-2). Most amplitude units used elsewhere in S1.EXE software are available. Several are absolute: %FS, dBFS, V, dBV, dBu, dBm, and W. The two FS units are meaningful only when a digital input signal has been acquired (Dual Domain units only), where the FS refers to digital full scale. dBFS will normally be used for FFT spectral displays of digital input signals, and %FS (linear) for waveform display of digital signals.

earlier. Setting START and STOP frequencies with a span of less than a few hundred Hz will display the spectral lines as wide bars rather than lines.

B.2.5. <F9>, <F7>, <F6>, and <Alt><F6> Keys

With either FFT program in use, the <F9> key initiates a new cycle of signal acquisition, processing, transmission to computer, and display. If it is necessary to go to the panel or menu, the display may then be re-graphed with the <F7> key. Changes in GRAPH TOP and GRAPH BOTTOM values may be made and re-graphed via the <F7> key since all the displayed information is in computer memory.

If the display is to be changed from waveform to spectrum (or vice versa), it is not necessary to re-acquire data. The <F6> key will cause a new cycle of processing (FFT transform if frequency domain display is selected at SOURCE-1), transmission to computer, and display to take place from the acquired data which is still in DSP memory. If FFTGEN is in use and multiple acquisitions have been made and averaged in order to reduce noise (see below), the <F6> key will cause a re-transformation of only the last-acquired data, therefore destroying the noise reduction effect. If multiple acquisitions and averaging were done with FFTGEN, <Alt><F6> should be used for any necessary changes in display parameters since it will not cause a new FFT transformation and therefore will not destroy the noise-reduction effect of averaging.

Changes between interpolated versus non-interpolated waveform or turning the average-subtraction routine ("AC coupling") off and on may be effected with either the <Alt><F6> or <F6> keystrokes. Changes in the # STEPS value and SOURCE-1 START and STOP values on the Sweep (F9) Panel may also be effected with either the <F6> or <Alt><F6> keys in order to re-process the DSP-memory-stored data according to the new parameters.

B.3. Features Unique to FFTSLIDE.DSP

FFTSLIDE.DSP has several features not available in FFTGEN.DSP. See Figure B-6 for an illustration of the unique features on the DSP panel and Figure B-7 for the DSP HELP panel with FFTSLIDE.DSP loaded.

B.3.1. Record Length, FFTSLIDE

FFTSLIDE.DSP acquires a maximum record length (one memory-full) upon trigger except in the case of the 1 kHz sample rate selection. With the 1 kHz rate, the record length acquired is 1.5 times the length selected in the record length field (or the full memory length, whichever is less). The available data memory to FFTSLIDE is 8,192 samples per channel in the standard System One + DSP (SYS-222) and 30,720 samples per channel with the MEM option or in System One Dual Domain (SYS-300 series). The amount of memory available will be displayed as part of the HELP panel information.

Note that acquisition times with FFTSLIDE become quite long at lower sample rates; see the table below:

Rate	Time (sec.) Std Memory	Time (sec.) Max Memory
192k	0.043	0.160
48k	0.171	0.640
32k	0.256	0.960
8k	1.024	3.840
1k	6.144	24.576

B.3.2. FFT Start Point Sliding

The FFT START field of the FFTSLIDE DSP panel permits control over the starting point in the record for the FFT computation. The record length field ("FFT input data length") controls the number of samples, beginning at the FFT START point, which will be used in the FFT computation. As always, the number of frequency points in the spectrum will be 1/2 the number of samples in the selected record length.

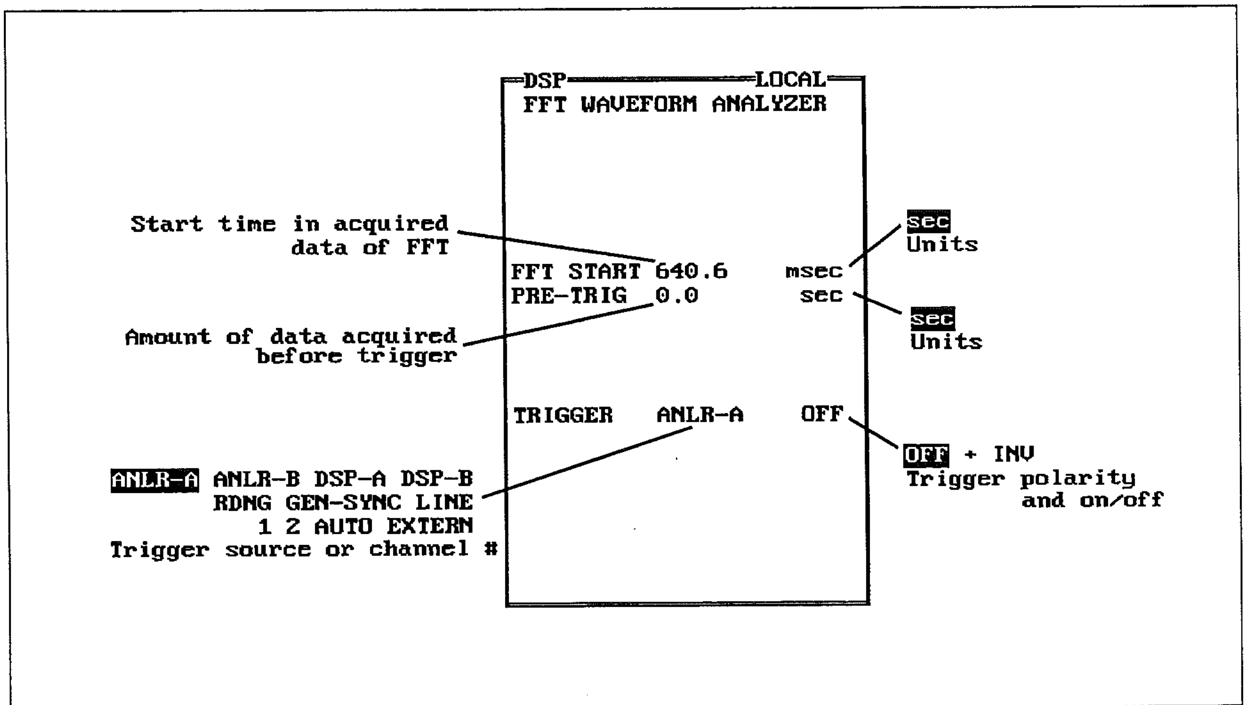


Figure B-6 DSP Panel Fields Unique to FFTSLIDE.DSP

DSP HELP SCREEN

One or two analog or digital signals may be acquired and displayed in either the time or frequency domains. DSP panel and bargraph readings display the real time peak input to allow setting of levels. When the DSP AMPLitude is graphed on the sweep panel vs DSP TIME an oscilloscope display of the waveform will result. A selected piece of the acquired signals may be transformed with an FFT to see the frequency spectrum. The beginning of the piece to be transformed is set with the FFT START time field on the panel. To display the frequency spectrum select a graph of DSP AMPL vs DSP FREQ.

The F9 function key causes a new data acquisition and transform. The F6 key causes previously acquired data to be transformed. The acquisition may be triggered by the positive or negative transition of any one of the input signals. Data may be acquired before the trigger event by selecting a PRE-TRIGger time less than zero. The maximum acquired record is 8192 points with standard memory, 30720 points with full memory. The maximum transform length is 4096 points with standard memory, 16384 points with full memory.

BH4 selects a Blackman-Harris 4 term minimum sidelobe window, HANN a raised cosine, FLAT has less than 0.02 dB rolloff, OFF gives no windowing.

CAUTION: DSP Panel readings must use %FS or dBFS. F7 does not allow units to be changed, use F6. DATA1 and DATA2 must not be the same (AMPL-1 or AMPL-2). At the 1k sample rate, acquired length equals 1.5 times the FFT input length.

Figure B-7 DSP HELP Screen with FFTSLIDE.DSP Loaded

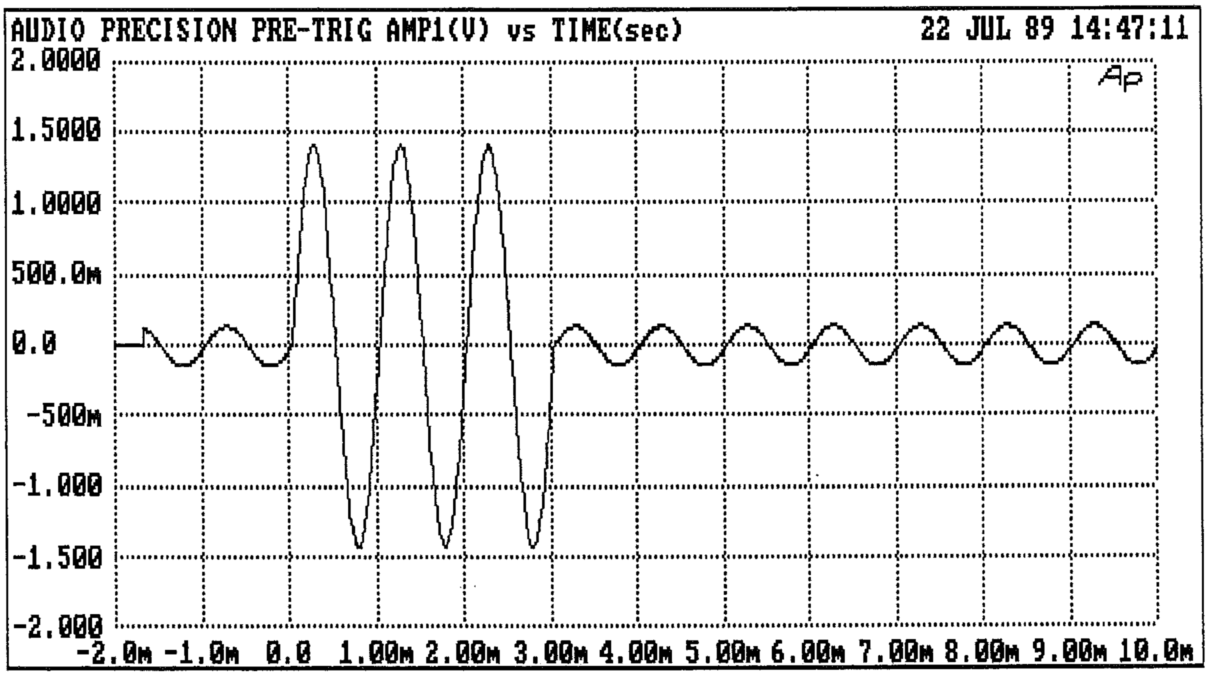


Figure B-8 Sine Burst Acquired With 1.5 Milliseconds Pre-Trigger

This sliding start point capability permits frequency analyses to be performed at various points of an acquired complex signal. If the signal is a swept or stepped frequency sweep, for example, spectral analysis can be done at different points through the acquired sweep to view the fundamental signal component plus distortion products and sidebands.

The initial trigger-and-acquire operation is controlled by the <F9> key. Once the signal is acquired and resident in the DSP module memory, further FFT transforms are accomplished via the <F6> key. Thus, each time the FFT START point or record length for FFT is changed, an <F6> operation will cause a new transform and display.

B.3.3. Triggering Capability, FFTSLIDE.DSP

FFTSLIDE has much more flexible triggering than FFTGEN, with a number of sources selectable into a hardware trigger generator in addition to software triggering. The TRIGGER source selection may be independent from the input selection fields for CH-1 and CH-2. The TRIGGER field includes as choices hardware triggering from both ANLR channels, both DSP BNC input connectors, the analyzer RDNG meter, GEN-SYNC, LINE, and EXTERN (pin 3 of DSP rear-panel 15-pin serial connector on Dual Domain units). Software triggering is also available from 1 (DSP channel 1 selected signal), 2 (DSP channel 2 selected signal), and AUTO (either DSP channel).

GEN-SYNC refers to the analog generator sync output signal. This signal is a squarewave at the generator frequency in sinewave and squarewave modes; the envelope of the burst signal in sine burst, trig, and gate modes; a squarewave at the lower IMD frequency in SMPTE IMD mode; a squarewave at one-half the spacing frequency in CCIF IMD mode; the squarewave IMD signal in DIM IMD mode; and a pulse at the pseudo-random repetition rate in PSEUDO noise modes. There is no signal at this point in RANDOM noise modes and therefore the DSP will not trigger (select OFF in the trigger slope field). LINE refers to the AC power mains line frequency. EXTERN selection monitors pin 3 of the general-purpose serial connector on the rear of Dual Domain (SYS-322) systems. The EXTERN trigger selection is thus not functional on SYS-200 series (System One + DSP) units. If pin 3 is high (or open circuit, since it is pulled high by an internal resistor), triggering occurs at the next sample. Pulling pin 3 low with external logic circuitry holds off triggering. The DSP will then trigger upon the first digital input sample after pin 3 goes high. EXTERN always requires a high level at pin 3 to trigger and is thus unaffected by the trigger slope selection described below.

When RDNG is selected as the CH-1 source for acquisition, the appropriate selection for proper triggering also depends upon the ANALYZER READING meter function. In THD+N mode with a low distortion sinewave signal, for example, the RDNG signal may consist only of signal-unrelated noise and very low amplitude distortion products which are undesirable or of insufficient amplitude to trigger the DSP. In such a case, select the ANLR channel which feeds the READING meter as the trigger source. Similarly, in SMPTE, CCIF, or DIM IMD modes the ANLR channel will produce more reliable triggering than the RDNG signal which consists only of demodulated IMD products. If a low sample rate such as 8 kHz or 1 kHz is being used for maximum resolution when examining demodulated CCIF IMD products at the RDNG meter output, triggering from the ANLR signal will not be possible when the fundamental signals are outside the currently-selected effective DSP bandwidth (for example, 13 kHz and 14 kHz).

The right-hand field on the TRIGGER line permits selection of trigger signal slope upon which the DSP will trigger an acquisition, except in EXTERN source. The + choice causes triggering at a positive-going zero crossing of the trigger source signal. INV causes triggering on the negative-going zero crossing. With the OFF selection, the DSP will acquire immediately upon pressing <F9>, equivalent to the "free running" trigger selection on oscilloscopes.

B.3.4. Pre-Triggering Capability, FFTSLIDE.DSP

In some cases it is desirable to view data which occurred just before the trigger signal. This is possible with FFTSLIDE since the DSP unit actually starts acquiring data into memory when the <F9> key is pressed, even before a trigger occurs. The newest samples overwrite the oldest samples.

Only negative time values can be entered into the PRE TRIG field. When <F9> is then pressed and a trigger event occurs, the specified amount of pre-trigger data samples is retained in addition to data sampled after the trigger to fill the remaining memory. Pre-trigger plus post-trigger samples will equal the acquired record length.

To view the waveform of the data including any PRE TRIG amount, select DSP TIME at SOURCE-1 and enter as START a negative time value equal to or greater than the value used in the PRE TRIG field of the DSP panel. See Figure B-8 for an illustration of a sine burst from System One's BURGEN option. The burst LOW LVL was set to -20 dB to produce the lower amplitude sinewave between the bursts. A value of -1.5 milliseconds was entered in the PRE-TRIG field of the DSP panel. The SOURCE-1 START value was set to -2 milliseconds to show that no signal was acquired before the pretrigger point. The actual "time zero" trigger point occurred when the burst envelope from the GEN-SYNC source went positive at the beginning of the burst. This results in 1.5 milliseconds of signal acquired before time zero. For FFT computations including the pretrigger samples, specify the same negative value as the FFT START time on the DSP panel.

B.3.5. Sweeping Pre-Trigger or Start Time

In addition to the FREQ and TIME selections at SOURCE-1 on the SWEEP (F9) DEFINITIONS panel, FFTSLIDE also offers PRET and STRT choices. PRET refers to the pre-trigger (PRE-TRIG) time value for signal acquisition, which must be zero or a negative value within the available re-

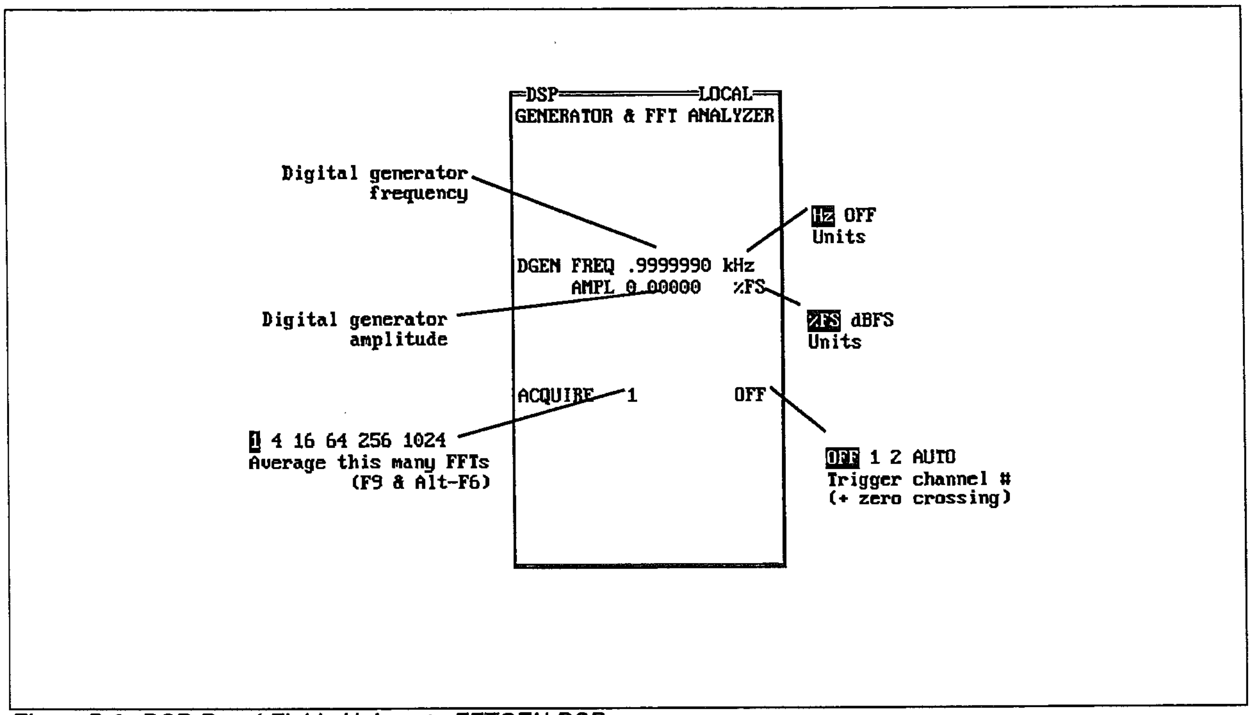


Figure B-9 DSP Panel Fields Unique to FFTGEN.DSP

DSP HELP SCREEN

Sinewave signals may be generated and one or two signals may be acquired and displayed in either the time or frequency domains. DSP panel and bargraph readings display the real time peak input to allow setting of levels. When the DSP AMPLitude is graphed on the sweep panel vs DSP TIME an oscilloscope display of the waveform will result. The acquired signals may be transformed with an FFT to see the frequency spectrum by selecting a graph of DSP AMPL vs DSP FREQ.

The F9 function key causes a new data acquisition and transform. The F6 key causes previously acquired data to be transformed. The acquisition may be triggered by the positive transition of either input signal. The maximum acquired and transformed record is 4096 points with standard memory, 16384 points with full memory. Multiple acquisitions may be averaged in the frequency domain to reduce variability. When averaging, the F6 key only re-transforms the last acquisition, Alt-F6 will re-display averaged data.

BH4 selects a Blackman-Harris 4 term minimum sidelobe window, HANN a raised cosine, FLAT has less than 0.02 dB rolloff, OFF gives no windowing.

CAUTION: DSP Panel readings must use %FS or dBFS. F7 does not allow units to be changed, use F6. DATA1 and DATA2 must not be the same (AMPL-1 or AMPL-2). Digital sinewave generation not allowed at 176.4kHz and 192kHz sample rates.

Figure B-10 DSP HELP Screen with FFTGEN.DSP Loaded cord length for the memory installed. STRT refers to the start time of the FFT computation (FFT START) within the acquired record, and can be negative if the data was acquired with a non-zero pre-trigger value.

B.4. Features Unique to FFTGEN.DSP

Certain features and characteristics are unique to the FFTGEN.DSP program. Figure B-9 shows the unique features of the DSP panel and Figure B-10 the DSP HELP panel with FFTGEN.DSP loaded.

B.4.1. Triggering Capability, FFTGEN.DSP

FFTGEN has no hardware triggering capability; only software triggering is available. FFTGEN triggers signal acquisition upon the positive-going zero crossing of the signal selected at "1" (CH-1; channel 1), "2" (CH-2; channel 2), AUTO, or DGEN. Selections "1" or "2" both have a fixed triggering threshold of 0.1%FS on the selected channel. AUTO selects whichever channel has the higher-amplitude signal and will trigger on any ampllitude greater than digital infinity zero. The DGEN selection functions only on System One Dual Domain (SYS-300 series) units. On those units, waveform acquisition is triggered at the beginning of each sinewave cycle or, if arbitrary waveforms are being generated from a downloaded .WAV file (see Arbitrary Waveform Generator section on page B-19), at the beginning of each "scan" through the generator waveform buffer.

B.4.2. Record Length, FFTGEN

FFTGEN acquires only the number of samples selected at the record length field, while FFTSLIDE normally fills the entire available memory. The HELP panel will display the amount of DSP memory available. When acquisition time (or disk storage space, if the waveform is to be stored) is not an issue, the MAXIMUM record length will typically

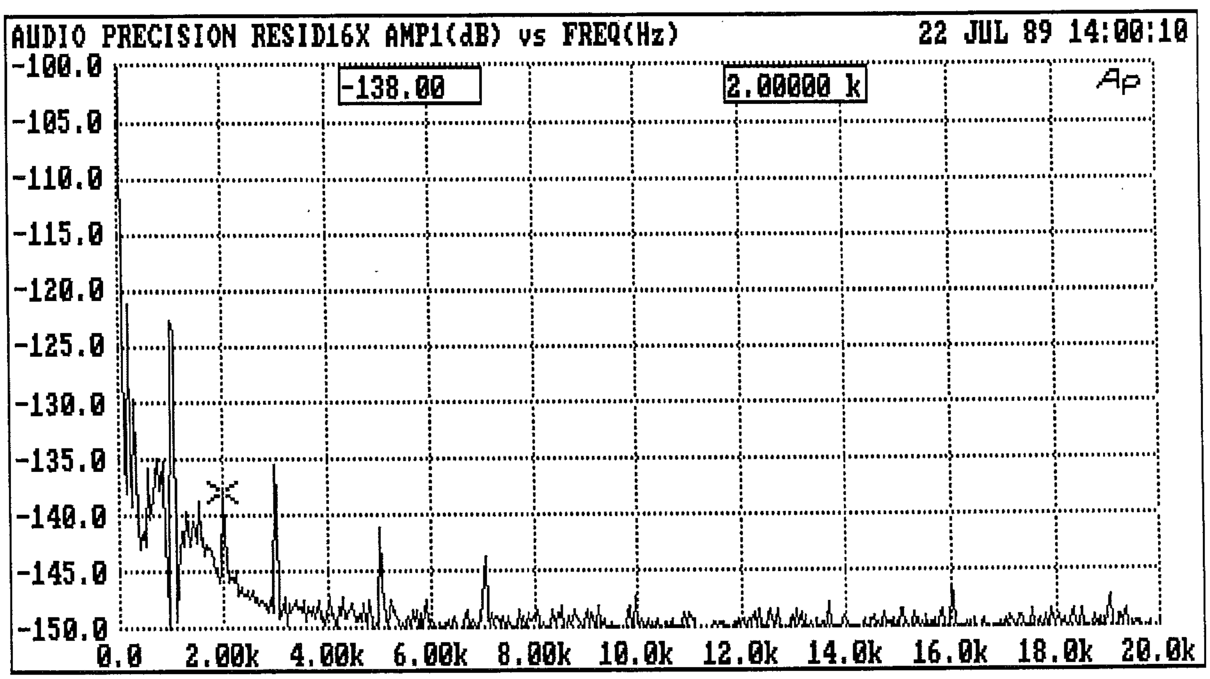


Figure B-11 Residual Distortion of Analog System One, FFT With 16X Averaging be selected before operation of the <F9> key. With an SYS-200 series instrument without MEM option, MAXIMUM corresponds to 4,096 samples (2,048 line frequency spectrum). With the MEM option or an SYS-300 series unit, MAXIMUM provides 16,384 samples (8,192 line frequency spectrum). If measurement speed is critical or the waveform is to be stored, trade-offs may be made of shorter record length for shorter acquisition time with consequently poorer spectral resolution available after the FFT computation. The speed issue is likely to become most critical when the lower sample rates (8 kHz and 1 kHz) are selected. The table below shows the acquisition time with the MAXIMUM selection for standard memory (SYS-200 series) and maximum memory (SYS-300 series or SYS-200 with MEM option):

Rate	Time (sec), Std Memory	Time (sec), Max Memory
192k	0.021	0.085
48k	0.085	0.341
32k	0.128	0.512
8k	0.512	2.048
1k	4.096	16.384

B.4.3. Spectrum Averaging, FFTGEN

FFTGEN permits 4 to 1024 repeated cycles of acquisition and FFT transform, and averages the resulting spectra automatically before display. The result is a significant reduction in the peak-to-peak magnitude of noise and thus an improvement in the ability to display a stable, coherent signal near the noise floor. This feature may also be used for averaging the spectral content of program material. Figure B-11 shows the result of 16 repeated acquisition-FFT computation cycles with the resulting spectra automatically averaged. The signal is the residual distortion of System One's analog generator and analyzer. Compare this graph with Figure A-1 on page A-4 to see the result of averaging. The coherent signals (suppressed fundamental and harmonics) are essentially unchanged by the averaging, while the peak-to-peak dispersion of the noise is reduced. This makes harmonic products visible which were obscured by noise in the single acquisition case. Note that this averaging is spectral magnitude averaging and not synchronous averaging of the acquired waveform.

The <F6> key causes a re-transform and display of the most recently-acquired signal on FFT programs. If the multiple-acquire-and-average mode was used to obtain the current display, re-transforming loses the noise-reduction advantage since it operates only on the last-acquired signal. To change display units, # STEPS, frequency span to be displayed, etc. without losing the advantage of averaging, use the <Alt><F6> keystroke. <Alt><F6> does not cause a re-transform, but will transfer data again from the DSP to the computer in accordance with the current SWEEP (F9) DEFINITIONS panel settings. This permits changing these other aspects of the display.

B.4.4. Digital Sinewave Generator

FFTGEN.DSP includes a digital sinewave generator program in addition to its acquisition and spectral analysis capabilities. The generator portion of the program functions at the 1 kHz, 8 kHz, 32 kHz, 44.1 kHz, and 48 kHz sample rates. In each case its maximum frequency is limited to 1/2 the sample frequency. The generator operates continuously and normally during panel mode, acquisition, and FFT computation. It is, however, shut down during SAVE WAVEFORM and LOAD WAVEFORM operations.

The dither feature described in the first chapter is fully operational with the FFTGEN digital generator.

The AMPLITUDE and FREQUENCY fields permit control of the signal generated. The sinewave is a full 24-bit amplitude resolution signal, with frequency resolution of (rate)/2^{24} where "rate" is the digital sample rate selected. With a 48 kHz sample rate, for example, the resolution is $\frac{48{,}000}{2^{24}}$ or approximately 0.0029 Hz.

Settling is disabled during ANLR vs DGEN sweeps. Therefore, the digital generator of FFTGEN should not be used for sweeps; the digital generator in the GENANLR.DSP program should be used instead.

B.4.5. Arbitrary Waveform Generator

FFTGEN.DSP can also function as an arbitrary waveform generator using downloaded waveforms (.WAV files). This powerful feature may be used for intermodulation distortion testing in the digital or analog domains, to produce squarewaves in the digital domain in order to view time domain aberrations, to create precise calibration signals for harmonic distortion analyzers, intermodulation distortion analyzers, and wow and flutter meters, and other applications. Sample waveform files are furnished for several of these applications. Custom waveforms may be created by the user with the MAKEWAVE.EXE utililty furnished with the DSP programs; see the "FASTEST.DSP Program for Fast Audio Testing" chapter of this DSP manual for full instructions on the use of MAKEWAVE.

This arbitrary waveform generation function works only at the 32 kHz, 44.1 kHz, and 48 kHz sample rates. FASTEST.DSP also shares this same arbitrary generation capability and adds unique analysis modes for noise, total distortion and noise, and interchannel phase, but lacks spectrum averaging capability. Either FASTEST.DSP or FFTGEN.DSP may be used to generate calibration signals for analog audio testing instruments. FFTGEN.DSP may be preferred for IMD testing in the digital domain due to its spectrum averaging feature.

The LOAD WAVEFORM command is used to download a .WAV file. The operator will then be prompted:

Enter buffers to be used "#[T or G] [,#[T or G]]":

The operator should normally respond "1G" to load an appropriate waveform file into the "G" (generator) buffer of FFTGEN. This waveform will be generated in both channels of FFTGEN. ***If the .WAV file is a multi-sinewave signal created with MAKEWAVE.EXE, the DGEN FREQ field on the DSP panel must then be set to 0.0 Hz.*** The DGEN AMPL field controls the digital output signal amplitude. A trigger signal is fed to the DSP module rear-panel BNC TRIG OUT connector at the beginning of each cycle through the buffer.

B.4.6. Intermodulation Distortion Testing

MAKEWAVE.EXE can generate waveform files consisting of essentially any number of sinewaves at any arbitrary amplitudes and starting phases. Downloading these waveform files to FFTGEN.DSP will cause this signal to be continuously generated in the digital domain. The signal may be used in the digital domain as stimulus to a digital device or D/A converter. It may also be converted to the analog domain by the internal D/A converter in System One + DSP and routed through the System One generator output stage via the WAVEFORM DGEN selection near the top of the GENERATOR panel. These digitally-generated IMD test waveforms are much more flexible than the analog-generated IMD waveforms furnished by the System One IMD Option. The digitally-generated IMD waveforms have performance superior in some respects and inferior in other respects to the analog IMD option. Analysis of the distortion products at the output of the device under test is performed with the FFT capability of FFTGEN.DSP.

B.4.6.1. Digital Domain IMD Test Signals

Standard and special IMD test signals are generated with MAKEWAVE.EXE simply by specifying in the input .DAT file the frequencies and amplitudes desired. Except for the small frequency round-off necessary to make all sinewaves exact integer multiples of 5.859375 Hz (see the FASTEST.DSP chapter), digital domain IMD signals can be created in exact compliance with SMPTE, DIN, CCIF, and DIM/TIM standards. Waveform files are furnished for these specific signals, and variations may be created by use of MAKEWAVE. The signals will be generated at the full 24-bit resolution of FFTGEN.DSP. The SIZE field should be used to

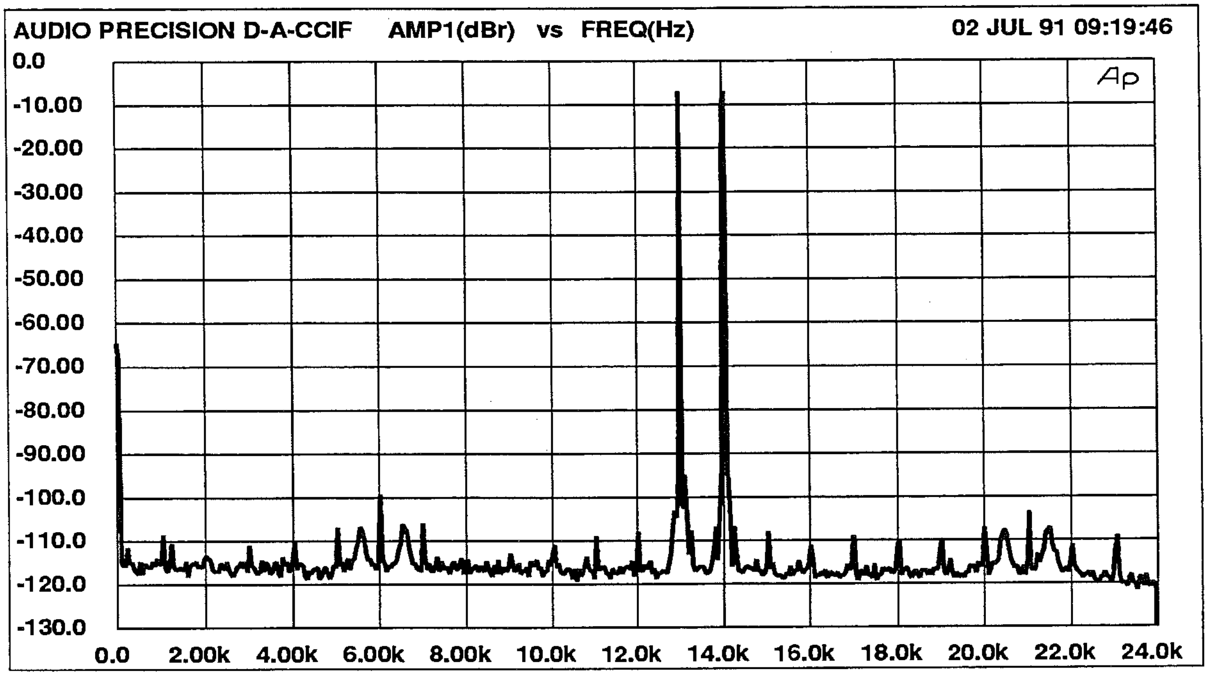


Figure B-12 IMD Products of D/A Converter Driven with Digital Domain CCIF Signal Generated from CCIF.WAV and FFTGEN.DSP.

round this 24-bit word down to the word length of the device being tested, and use of the selectable dither is recommended for most applications.

Analysis of the device-under-test output while being stimulated by these signals is via FFT. Signal to the DSP may be acquired from any of the digital inputs when testing D/D devices, or from the analog inputs when testing D/A converters. Note that, if the device under test does not shift frequency nor introduce wow and flutter, FFTGEN can operate with no window since the generated signal and acquired signal are synchronous. FFT selectivity is then essentially perfect with no energy spill-over into adjacent FFT bins. To turn the window function off, select NONE in the right-hand field of the TRANSFORM line on the DSP panel.

When measuring analog domain signals, performance will be limited by the specified -85 dB (relative to full scale) guaranteed linear range of System One's input A/D converters. See the Performance Factors section of the FASTEST chapter on page C-24. Figure B-12 shows the intermodulation distortion products of the D/A converter in an RDAT machine with digital domain stimulus from the CCIF.WAV file furnished with the DSP programs. The 64-times spectrum averaging feature of FFTGEN.DSP was used to reduce the peak-to-peak variation of noise and make low-amplitude products easier to see. The analog analyzer INPUT was fixed on the 1.2 Volt range, which placed the 2.00 Volt output of the RDAT machine at 89%FS of the A/D converter in the DSP unit, gaining approximately 6 dB in lower distortion and noise compared to the 2.5 Volt range which would have been chosen by AUTO ranging. Care must be exercised when fixing the INPUT range, however, to be sure that signal peaks do not exceed full-scale of the converters.

B.4.6.2. Improved Analog Domain IMD Testing

System One's analog IMD option generates test signals in accordance with three IMD standards; SMPTE/DIN, CCIF, and DIM/TIM. In addition, the IMD option permits certain flexibility of frequency

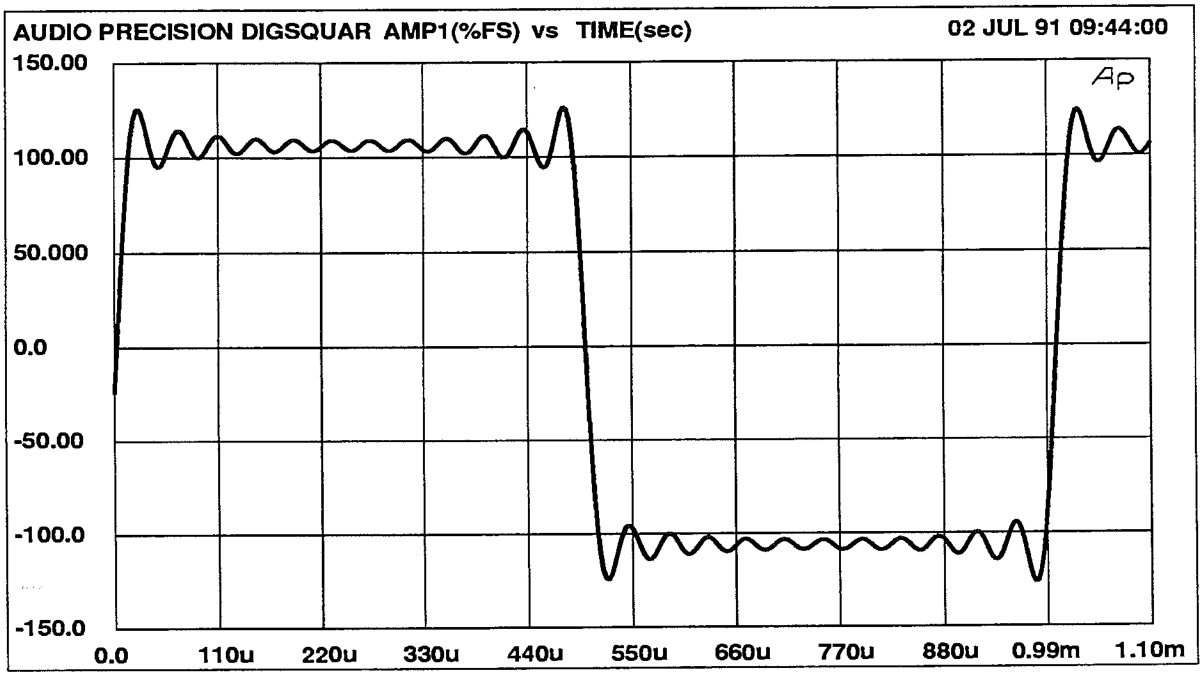


Figure B-13 Digital Domain 1 kHz Squarewave Generated by SQUARE.WAV with FFTGEN.DSP.

choices. But, the IMD option cannot generate (for example) a three-tone signal, nor a CCIF-like signal with more than 1 kHz spacing, nor IMD signals of arbitrary amplitudes, nor IMD signals with both tones below 2.5 kHz. MAKEWAVE.EXE and FFTGEN.DSP can do all those and can also generate standard CCIF signals with superior performance in certain respects to the analog IMD option generator.

The CCIF IMD analysis capability of analog System One is limited to measuring the amplitude of the second-order difference frequency falling at a low frequency (1 kHz in the case of a 13 kHz/14 kHz tone pair). This is also true of the few other analyzers available with CCIF capability. Some devices, notably analog magnetic tape recorders, generate principally odd-order distortion products and are thus not usefully measured by this test. A more general CCIF IMD test is made by also measuring the amplitude of the third-order products falling at 12 kHz and 15 kHz. This measurement is easily done with the FFT capability of System One DSP models.

The System One IMD option generator, however, was optimized only for use with the IMD option analyzer. Thus, its second-order difference product output is extremely low (below -106 dB). The technique used to generate this signal produces spurious signals at the center frequency between the two tones and at one-half the tone pair spacing near the two tones, and is not optimal for low third-order distortion. Instead, the multitone technique is a superior method for generation of a CCIF signal when the FFT analyzer will be used to measure IMD products near the two stimulus tones. In general with digital signal generation by the DSP unit, distortion products (due to non-linearity of the D/A converter) will be at least 85 dB below full-scale output of the D/A.

B.4.7. Time Domain Testing in the Digital Domain

None of the .DSP programs presently furnished have digital-domain squarewave generation capability as a standard feature. However, it is possible to create a waveform file of a squarewave, download it to FFTGEN.DSP, and view aberrations of the resulting device output signal in the time domain (oscilloscope mode).

The frequency-domain description of a squarewave consists of a fundamental sinewave component of reference amplitude, plus in-phase sinewaves at every odd harmonic. The amplitude of each harmonic component is 1/n of the fundamental amplitude, where n is the harmonic order. Thus, a 1 kHz squarewave can be synthesized by sinewaves as follows:

1 kHz	1.000 Volt
3 kHz	1/3 Volt
5 kHz	1/5 Volt
7 kHz	1/7 Volt

etc.

Generating these signals digitally requires that they will be band-limited at the Nyquist frequency of one-half the sampling rate. This band-limiting sets a finite limit on the rise and fall times of the squarewave. Figure B-13 is a time domain view of the digital-domain squarewave generated as described above with MAKEWAVE.EXE, using the 48 kHz sample rate and consisting of all odd harmonics through 23 kHz. This waveform is furnished as SQUARE.WAV on the DSP diskettes.

MAKEWAVE adjusts each user-requested frequency to an exact multiple of a 5.859375 Hz fundamental frequency (assuming 48 kHz sample rate) in order that the signal will be transient-free when the end and beginning of the record are spliced together as the DSP generator operates continuously. Synthesizing a squarewave also requires that each harmonic is an exact odd integer multiple of the fundamental. To simultaneously satisfy these two conditions requires some pre-calculations by the user. The harmonic of the basic 5.859375 Hz frequency nearest 1 kHz is the 171st harmonic, or approximately 1002 Hz. To calculate the 3rd harmonic, this 171 factor is then multiplied by three (171 * 3 = 513) and 513 multiplied by the base frequency to produce a 3006 Hz product which has the required relationship to both the 5.859375 Hz base sinewave and the 1002 Hz squarewave fundamental. This same technique was used to calculate the approximate frequency for each odd harmonic. Accuracy to the nearest Hz is sufficient, since MAKEWAVE will round from the user-requested value to the nearest exact multiple of the base frequency.

Note that if a squarewave generated in the digital domain is converted to analog by the DSP D/A converter and fed through the analog generator output stage via the DGEN selection, significant tilt will be produced by the ac coupling from D/A output to the input of the generator output stage. For most applications, analog-domain squarewaves are better generated by the BUR option.

B.4.8. Calibration Signals

Calibration of distortion analyzers and wow and flutter meters is not a simple task. Secondary standards, which exist for basic electrical parameters such as voltage and current, do not exist for these parameters. The usual calibration technique for harmonic and intermodulation distortion is to combine the outputs of several sinewave generators via a resistive network, after first carefully setting the individual amplitudes while connected to load resistances exactly equal to the network input resistance. For wow and flutter calibration, a typical technique involves frequency-modulating a function generator with the output of an audio oscillator. A low-frequency spectrum analyzer and the Bessel null technique are then used to set the frequency modulation index to the desired wow and flutter percentage value.

B.4.8.1. Distortion Calibration

The multi-tone technique makes it very simple to create signals for harmonic and intermodulation distortion analyzer calibration.

To create a 1.0% harmonic distortion signal requires only a fundamental sinewave plus a second sinewave of at least twice the fundamental frequency with an amplitude of 1% (-40 dB) of the fundamental. It is not necessary to create an exact 2:1 frequency ratio of "harmonic" to fundamental, since THD+N analyzers operate by filtering out the fundamental and measuring everything left over. It can be useful to have several waveforms with the "harmonic" at the same amplitude, but at different frequencies on the order of 2:1 to 3:1 above the fundamental. These calibration waveforms permit testing the analyzer's error with different harmonic orders. A major portion of the error of many THD+N analyzers is attenuation of the second harmonic by the notch filter; this attenuation is usually negligible by third and higher harmonics. Files are furnished with 1% second, third, and ninth harmonic of a one kHz fundamental signal (CALTHDK2.WAV, CALTHDK3.WAV, and CALTHDK9.WAV).

Intermodulation distortion calibration signals can be created for the three standards measured by the System One analog IMD option. Note that SMPTE IMD is calibrated as percentage amplitude modulation of the high-frequency signal. A 100% amplitude-modulated signal has two sidebands each 6.02 dB below the carrier amplitude. Thus, a 1% second-order SMPTE IMD calibration signal must have two sidebands 46.02 dB below the high-frequency "carrier". Since the SMPTE standard specifies the high-frequency signal amplitude to be one-fourth (-12.04 dB) of the low-frequency signal, the entire 1% distortion calibration signal consists of these products:

59 Hz	0 dB
6943 Hz	-58.06 dB
7002 Hz	-12.04 dB
7061 Hz	-58.06 dB

It is important that the two 1% distortion sidebands be spaced by exactly the same frequency difference above and below the high-frequency signal. This signal is furnished as CALSMPTE.WAV. A similar signal, but with a nominal 8 kHz "carrier" and nominal 250 Hz lower frequency and sideband spacing for DIN IMD analyzer calibration is furnished as CALDIN.WAV.

A 1% distortion CCIF signal for the System One IMD analyzer is furnished as CALCCIF.WAV. A low-frequency tone is located at 1.00195 kHz, the difference between the 13.00195 kHz and 14.00405 kHz tones. This low-frequency tone is 40 dB below the amplitude of either high-frequency signal in order to furnish a 1% CCIF IMD reference.

CALDIM.WAV is a 1% distortion calibration signal for the DIM mode of the System One IMD analyzer. A DIM signal consists of a squarewave and a high-frequency sinewave "probe tone". The squarewave signal is constructed from a sinewave fundamental at approximately 3.15 kHz plus the 3rd, 5th, and 7th harmonics of that exact fundamental as described above (see page B-21) under Time Domain Testing in the Digital Domain. A "probe tone" sinewave is also generated at 15 kHz with amplitude 14.14 dB below the sinewave at fundamental frequency. The DIM signal is defined to have the sinewave in a 1:4 (-12.04 dB) ratio to the peak value of the squarewave. The fundamental component of a squarewave can be computed to be ($\pi/4$) below the peak of the composite squarewave. Therefore the probe tone sinewave must be $\pi/16$, or 14.14 dB below the fundamental. The 1% distortion product is 40 dB below the amplitude of the probe tone, and may be located in frequency anywhere between about 700 Hz and 2.4 kHz to fall within the bandpass of the IMD option DIM analyzer.

B.4.8.2. Wow and Flutter Calibration

Wow and flutter consists of frequency modulation of a signal, caused in practice by imperfect mechanical rotating parts or imperfect speed control of a tape recorder or turntable. IEC/DIN wow and flutter is numerically defined in terms of peak frequency deviation of the fm as a percentage of the "carrier" frequency. Thus, at a 3.000 kHz test frequency ("carrier"), 1.0% wow and flutter corresponds to 30 Hz peak deviation. The bandwidth defined for unweighted wow and flutter measurements is from 0.2 Hz to 200 Hz. The 5.859375 Hz fundamental frequency of the multitone generation technique (assuming 48 kHz sample rate and 8,192 sample generator record) is a convenient modulating frequency (flutter rate).

To construct the spectrum of a frequency-modulated signal with the multitone method requires use of the Bessel functions which mathematically describe any frequency modulation process. Tables of Bessel function values can be found in many mathematical and electronics reference texts. The tables are most convenient to use for specific integer values of modulation index. The modulation index of a frequency-modulated signal is the ratio of the peak deviation to the modulating frequency. A modulation index of 4.0 with a 5.859375 Hz modulating frequency results in peak deviation of 4.0 * 5.859375 or 23.44 Hz, which corresponds to 0.781% IEC/DIN unweighted wow and flutter (23.44 Hz/3000 Hz = 0.00781).

The phase of the carrier and each sideband must be specified when describing or constructing an fm signal. The multitone signal generation technique permits specifying in the third column of the .DAT file the phase of each sinewave generated (see the "Using MAKEWAVE.EXE" section of the FASTEST.DSP chapter). For an fm signal, the first order sidebands lag the carrier by 90 degrees, the second order sidebands lag the carrier by 180 degrees, etc. The table below is the .DAT file of a 3000 Hz frequency-modulated carrier with 5.859 Hz modulating frequency and a modulation index of four. Sidebands out through the 8th order were specified; 9th order and higher sidebands are of negligible amplitude for a modulation index of four. The negative sign for the amplitudes of the carrier and first-order sidebands comes directly from the Bessel tables, and results in a phase inversion of those components when MAKEWAVE.EXE processes this .DAT file into a .WAV file. Frequencies could have been entered with only one Hz resolution, since MAKEWAVE.EXE rounds the requested frequencies to the nearest integer multiple of the 5.859375 Hz fundamental.

Hz	V deg1
2953.125,	0.00400000019,0
2958.98438,	0.0152000003,90
2964.84375,	0.0491000004,180
2970.70313,	0.132100001,-90
2976.5625,	0.281100005,0
2982.42188,	0.430200011,90
2988.28125,	0.364100009,180
2994.14063,	-0.0659999996,-90
3000,	-0.397100002,0
3005.85938,	-0.0659999996,-90
3011.71875,	0.364100009,180
3017.57813,	0.430200011,90
3023.4375,	0.281100005,0

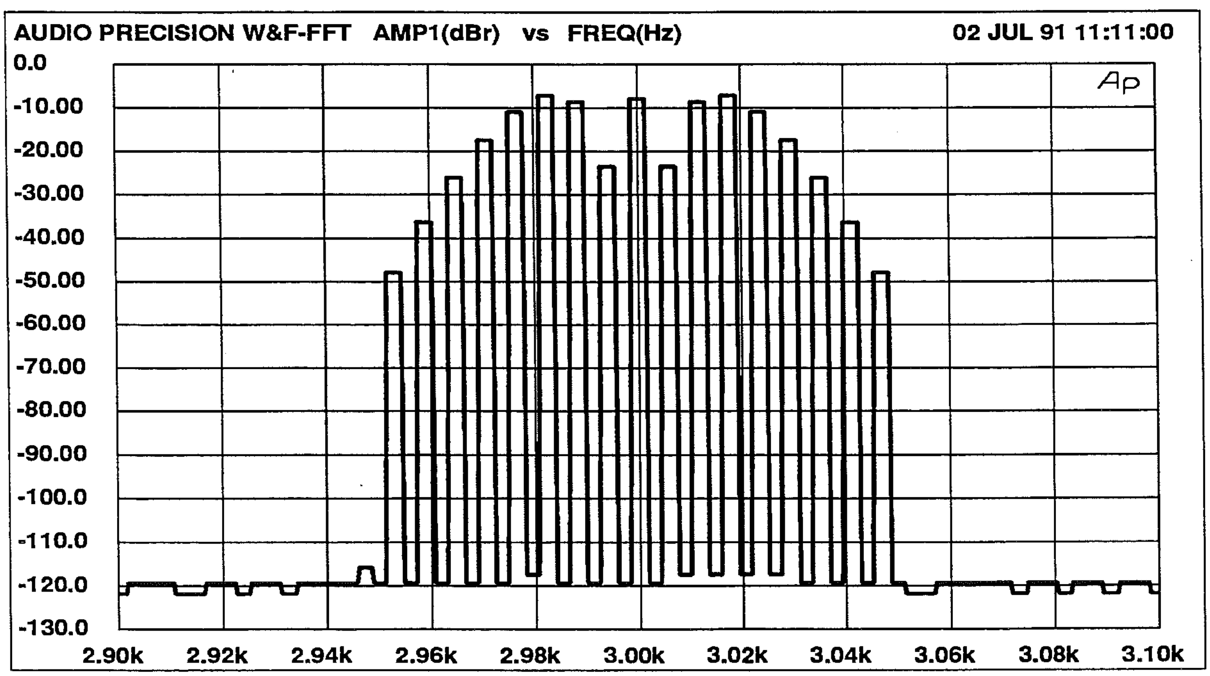


Figure B-14 Spectrum Analysis of Wow & Flutter Calibration Signal Generated by CALW&F.WAV with FFTGEN.DSP

3029.29688,	0.132100001,-90
3035.15625,	0.0491000004,180
3041.01563,	0.0152000003,90
3046.875,	0.00400000019,0

Figure B-14 shows the spectrum of this signal generated by the FFTGEN.DSP. Applying it to a wow and flutter analyzer in IEC/DIN unweighted mode should produce an indication of 0.781% wow and flutter. NAB and JIS unweighted modes should read 0.552%, since they are essentially RMS-indicating and thus read 3 dB lower than the peak-reading IEC/DIN mode. Selecting weighted measurements should produce a measurement of 0.95 times the unweighted value, since the wow & flutter weighting filter specified by all three standards is 0.45 dB down (0.95 voltage ratio) at 5.86 Hz from its 4 Hz unity-gain value.

B.5. Saving Waveforms to Disk

The SAVE WAVEFORM and LOAD WAVEFORM commands permit saving acquired waveforms to disk and later loading those waveforms back into the DSP module for further analysis. The .WAV file extension is automatically supplied by S1.EXE software during SAVE operations, and is expected by the LOAD command in order to display a menu of available waveforms in the current directory. Optional implementations of these commands permit saving waveforms from either or both channels of the DSP and saving only selected portions of the waveforms instead of the entire acquired waveform. The first 128 characters of the Edit Comments buffer (or text up to the first carriage return, which ever comes first) will be saved into the .WAV file and can be examined by some text editors or programs such as the DEBUG utility furnished with DOS.

B.5.1. Save Waveform Command

The SAVE WAVEFORM command may be used whenever acquired data is in the DSP module, resulting from either an <F9> key operation or a previous LOAD WAVEFORM command. SAVE WAVEFORM causes the actual waveform samples (not the FFT or the interpolated-for-display values) to be transferred from data memory within the DSP modules to the computer and stored in a disk file. SAVE WAVEFORM must be invoked with one or two numerical arguments, and may optionally also be invoked with one or two alphabetical arguments. The general form of the command is:

SAVE WAVEFORM filename <Enter>

followed by

#[T] [,#[T]] <Enter>

where # may be 1 or 2 and refers to CHAN-1 and CHAN-2 waveform data in the DSP modules.

Thus, selecting SAVE WAVEFORM results in the user being asked to supply a legal DOS file name into which the data will be saved. The .WAV file extension is automatically furnished by the system. Upon furnishing the file name and pressing <Enter>, the system will say:

Enter buffers to be used "#[T or G] [,#[T or G]]":

These characters are codes which control what data will be saved into which portions of the disk file. The first numeric code entered defines whether CHAN-1 or CHAN-2 data from DSP memory is to be stored in the first of two possible sections within the disk file. The second (optional) numeric code permits specification of the alternate DSP channel data to be stored into a second section within the disk file. For example:

SAVE WAVEFORM filename <Enter> 1,2 <Enter>

will cause data from CHAN-1 of the DSP to be saved into the first portion of the disk file and data from CHAN-2 to save into the second portion of the file. This is also the default operation if no codes are entered.

The command:

SAVE WAVEFORM filename <Enter> 2 <Enter>

will cause only the CHAN-2 DSP data to save into the disk file.

SAVE WAVEFORM filename <Enter> 2,1 <Enter>

causes CHAN-2 data to save into the first portion of the disk file and CHAN-1 DSP data to save into the second part of the file.

It is possible to save less than the full DSP memory contents by using the T option plus the "FFT input data length" field (on the TRANSFORM line of the DSP panel) and the FFT start time field of FFTSLIDE. The T option causes only the data in the "to be transformed" portion of DSP memory to transfer to the computer and be saved. If, for example, 2048 is selected as the "FFT input data length" value on the DSP panel of FFTSLIDE and the command is given:

SAVE WAVEFORM filename <Enter> 1T,2 <Enter>

the first portion of the disk file will contain only the first 2,048 samples of CHAN-1 data in the DSP, while the remainder of the file will contain the entire length of CHAN-2 DSP data.

With data present in FFTSLIDE, the "FFT start time" field and the "FFT input data length" field work together to define the "to be transformed" section of DSP memory which will be transferred to the computer when the T option is used. For example, if 27 milliseconds is entered as FFT start time and 4096 is selected as FFT input data length, the command:

SAVE WAVEFORM filename <Enter> 2T,1T <Enter>

will cause the 4,096 waveform samples starting at 27 milliseconds after trigger to be transferred to the computer and saved. CHAN-2 data will save into the first portion of the file and CHAN-1 data into the second portion.

In all cases, the necessary range information and sample rate information is transferred with the data and saved into the file. In the case of FFTSLIDE, the value of pre-trigger time is also saved.

Note that when data is acquired in the spectrum-averaging mode of FFTGEN.DSP, only the final data acquisition resides in memory even though the screen displays the averaged spectrum with consequent noise reduction. If SAVE WAVEFORM is used and the file later down-loaded and re-analyzed, no noise dispersion reduction will be visible since only the final acquisition was saved.

B.5.1.1. Waveform File Size

The disk files created by SAVE WAVEFORM can be very large compared to normal System One .TST files. With a maximum-memory unit and the FFTSLIDE.DSP program in use, saving one channel results in an approximately 92 kbyte file and both channels will result in a file of almost 185 kbytes. The actual file size will consist of 3 bytes per sample per channel (since the DSP acquires 24-bit data words and disk files bytes are 8 bits wide) plus 256 bytes of header per channel. Thus, a full 30,720 sample acquisition with FFTSLIDE will occupy 92,160 bytes per channel, plus 256 header bytes for a total 92,416 bytes per channel. A dual-channel file is thus 184,832 bytes. An arbitrary waveform generation file is 8,192 24-bit samples long plus header, for an exact length of 24,832 bytes.

The length of time required to transmit this data to the computer is considerable. A hard disk and a computer with at least an 80286 processor are virtual requirements for saving and loading waveforms. To conserve disk space when saving waveforms, save only the minimum record length and minimum number of channels required for the application. Shorter record lengths can be saved either by using FFTGEN.DSP, which acquires only the record length specified in the FFT input data length field of the DSP panel, or by using the "T" option and data length field of FFTSLIDE.DSP.

B.5.2. Load Waveform Command

The LOAD WAVEFORM command may be used to down-load the previously-acquired and stored information from disk files back into the DSP modules for further analysis and display or to down-load arbitrary waveform files into the generator buffer. Waveforms saved from one FFT program may be later loaded into the other. LOAD WAVEFORM is also invoked with one or two numeric arguments, which can be 0, 1, or 2. The position of these numbers in the command statement defines whether they refer to the first or second portion of data in the disk file. The value of the argument determines whether that data should simply be discarded (0), loaded into DSP CHAN-1 analysis buffer (1), or loaded into DSP CHAN-2 analysis buffer (2). Thus, the command:

LOAD WAVEFORM filename <Enter> 1,2 <Enter>

causes the first portion of the disk file to load into the CHAN-1 analysis buffer and the second portion to load into the CHAN-2 analysis buffer. This is the default operation if no numbers are entered. The command:

LOAD WAVEFORM filename <Enter> 0,1 <Enter>

causes the first portion of a two-record disk file to not load into the DSP, while the second portion loads into the CHAN-1 analysis buffer. The command:

LOAD WAVEFORM filename <Enter> 1G <Enter>

causes an arbitrary waveform file to load into the generator buffer, where it will create the signal for both output channels of the digital generator.

When a waveform file is loaded, the input source settings, input channel settings, and sample rate are changed to the settings in effect when the file was saved.

B.5.3. Using Down-Loaded Waveforms

When a stored waveform has been downloaded into the DSP analysis buffers, the <F6> function key must be used to transform and "bin" the data and send it to the computer for graphing. As with a freshly-acquired waveform, the SWEEP PANEL parameters may be changed for time domain or frequency domain display, individual sections of the time or frequency axis may be expanded, interpolation and "ac coupling" turned on and off, and a re-display made each time with the <F6> key.

B.5.4. Combining Waveforms Acquired at Different Times

The LOAD and SAVE capabilities may be used to combine into the dual channel analysis buffers of the DSP waveforms acquired at different times or places. Proper use of the numeric arguments during the LOAD operation allows any saved waveform to be placed into either the CH-1 or CH-2 analysis buffer of the DSP. However, it is critical that both these waveforms must have been acquired at the same sample rate. ***If waveform data files acquired at two different sample rates are loaded into DSP memory and the <F6> key pressed, both channels will be transformed and displayed according to the sample rate selected on the DSP panel when <F6> is pressed. The result will be directly proportional frequency errors in the record acquired at a different rate.***

B.6. Typical Applications and Tests

The DSP program diskette contains several typical tests already set up for use with the FFT programs. This section describes the use of these tests.

B.6.1. Waveform Display Applications

Typical time-domain applications of the FFT programs include audio waveform examination for artifacts such as clipping or crossover distortion, display of distortion products following the notch filter during THD+N measurements, measurement of delay time through devices such as a three-head tape recorders, satellite links, and signal processors, and general digital storage oscilloscope applications where a bandwidth of 80 kHz is adequate.

WAVE22K.TST and WAVE80K.TST on the DSP diskette are set up for dual-channel signal acquisition through the analog analyzer balanced inputs. Both are set up to trigger from a positive-going transition of the channel 1 signal, but can be changed for negative polarity or free-running triggering. They will perform time domain (waveform) display with bandwidths of 22 kHz (48 kHz sample rate) and 80 kHz (192 kHz sample rate), respectively. The DATA-1 and DATA-2 graph top and bottom values are selected for optimum dual-channel display of two signals of approximately one Volt. To display signals of significantly different amplitudes, estimate the peak-to-peak span of each signal. Set the DATA-1 graph top and bottom for approximately twice the expected span, with the GRAPH TOP at approximately the expected positive peak value. Set DATA-2 graph top and bottom for about twice the expected channel 2 span, with the GRAPH BOTTOM at about the expected negative peak value. For single-channel display, set DATA-2 to NONE and adjust the DATA-1 graph bottom value.

The SOURCE-1 START and STOP values may be set to examine any desired portion of the record with any desired horizontal time span.

B.6.1.1. Time Delay Measurements

DELAYDUT.TST is set up to measure the time delay through a device such as a satellite link, signal processor, or three-head tape recorder. It requires the BUR option to function. The System One analog generator channel A output must be connected to the input of the device under test. The test setup is stored with a one Volt output amplitude from the generator, but other amplitudes may be substituted as desired. The signal is a short burst of 1 kHz sinewave, repeated every 0.5 second. If the expected delay through the device under test is greater than 0.5 second, the BURST INTERVAL should be lengthened until longer than the greatest expected delay.

The analyzer acquires signal through System One's analog channel A balanced input. The setup is stored with a fixed 1.2 Volt range selected at channel A. If the expected device output amplitude is significantly different, change the fixed input range value accordingly; ***do not operate with autoranging with a tone burst signal.*** Acquisition is triggered each time a burst is sent from the generator, via the GEN-SYNC trigger selection with positive polarity. The received burst will be displayed on the screen and delay through the device measured by moving the graphics cursor to the first portion of the displayed pulse.

RES1K-T.TST is set up to display both the original signal and the noise and distortion products following the notch filter during a THD+N self-test of System One's analog generator and analyzer. By changing the analyzer inputs from GEN-MON to INPUT, it may also be used in stimulus-response testing of other devices. System One's analog generator may be used as the stimulus, or another generator or recorded test tone (CD test disc, for example) may be used. DATA-2 displays the original signal while DATA-1 displays the distortion products. The DATA-2 GRAPH TOP and BOTTOM values are selected assuming a signal of about one Volt; they can be changed for any other actual value. The DATA-1 GRAPH TOP and BOTTOM are set assuming very low distortion (100 microvolts peak-to-peak), but can be changed to any other desired value.

The SOURCE-1 START and STOP times are predicated on a signal frequency of about 1 kHz and may be changed to best display any other value. No analog bandwidth limiting is set up in the test, but may be selected. The test is stored with the 48 kHz sample rate, limiting the analysis bandwidth for noise and distortion products (DATA-1) to about 22 kHz. To increase this bandwidth to 80 kHz, change the sample rate to 192 kHz.

B.6.2. Spectral Analysis Applications

FFT22K.TST and FFT80K.TST are set up for dual-channel signal acquisition through the analog analyzer balanced inputs. Both are set up to trigger from a positive-going transition of the channel 1 signal, but can be changed for negative polarity or free-running triggering. They will perform frequency domain (spectrum analysis) display with bandwidths of 22 kHz (48 kHz sample rate) and 80 kHz (192 kHz sample rate), respectively. The DATA-1 and DATA-2 graph top and bottom values are selected for optimum dual-channel spectral display of two signals of about one Volt with dynamic ranges of approximately 120 dB. To display signals of significantly different amplitudes, set the DATA-1 graph top to approximately the expected channel 1 signal amplitude (in dBV or another selected unit) and graph bottom to about 240 dB below that value. Set DATA-2 graph bottom to a value about 120 dB below the expected channel 2 amplitude, and the graph top 240 dB above that. For single-channel display, set DATA-2 to NONE and adjust the DATA-1 graph top and bottom values for about a 120 dB range with graph top at the signal amplitude. The vertical display may also be changed from dB to Volts with a LOG axis.

SOURCE-1 start and stop frequencies may be changed to other values within the bandwidth (zero to one-half the sample rate) to "zoom in" on a selected frequency range. Assuming maximum memory (MEM option or System One Dual Domain), the maximum frequency resolution is approximately 3 Hz in FFT22K.TST and 12 Hz in FFT80K.TST, determined by the record length.

If SOURCE-1 START is changed from zero to a positive value, LOG may be selected instead of LIN for the horizontal axis. An FFT is inherently a frequency-linear process, however, and graphing the results logarithmically will accent the limited resolution at lower frequencies.

RES1K-F.TST is set up for a spectral display of the noise and distortion products following the notch filter during a THD+N self-test of System One's analog generator and analyzer. Since the fundamental frequency component has been removed by System One's analog notch filter, the limited dynamic range (85 dB distortion-free) of the DSP input circuitry is no longer a significant limitation and distortion products may be resolved below -130 dB. By changing the analyzer input from GEN-MON to INPUT, it may also be used in stimulus-response testing of other devices. System One's analog generator may be used as the stimulus, or another generator or recorded test tone (CD test disc, for example) may be used. The DATA-1 GRAPH TOP and BOTTOM are set assuming very low distortion (more than 100 dB below fundamental) but can be changed to any other desired value.

The SOURCE-1 START and STOP frequencies may be changed to display any desired range. The test is stored with the 48 kHz sample rate, limiting the bandwidth to about 22 kHz. To increase this bandwidth to 80 kHz, change the sample rate to 192 kHz.

B.6.2.1. Wow and Flutter Spectral Analysis

FFT-W&F.TST is set up to make a 0-200 Hz spectrum analysis of the wow and flutter components of a tape recorder or turntable playing a 3 kHz or 3.15 kHz reference tape or disk. The wow and flutter option must be installed in System One for this test to function.

B.6.3. Changing Between Waveform Display and Spectrum Analysis

Waveform display and spectrum display tests may be loaded without destroying the waveform data in DSP memory if both tests have the same DSP file attached. WAVE22K.TST, WAVE80K.TST, FFT22K.TST, and FFT80K.TST all use FFTSLIDE.DSP. Thus, you may acquire a signal with any one of these tests and then easily switch analysis domains by simply loading another of these tests and pressing <F6>. This is generally significantly easier than making all the necessary DATA-1, DATA-2, and SOURCE-1 changes to go between time and frequency domain display.

B.7. Furnished Files

A number of .TST and .WAV files are furnished on the General DSP diskette in addition to FFTGEN.DSP and FFTSLIDE.DSP. Their names and intended usage are listed below:

CCIF.WAV 13kHz/14 kHz CCIF IMD waveform generated by MAKEWAVE

D-A-CCIF.TST for IMD testing of D/A converters. Uses FFTGEN, generates signal in digital domain with CCIF.WAV, does 2-channel FFT from analog inputs. CCIF-WAV must be downloaded with the 1G option before this test will function

D-ASQUAR.TST for squarewave testing of D/A converters. Uses FFTGEN, generates signal in digital domain with SQUARE.WAV, displays 2-channel time domain signal from analog inputs. SQUARE.WAV must be downloaded with the 1G option before this test will function

D-D-CCIF.TST for IMD testing of digital domain devices. Same as D-A-CCIF.TST except serial digital outputs and inputs. CCIF-WAV must be downloaded with the 1G option for this test to function

D-DSQUAR.TST for squarewave testing of digital domain devices. Same as D-ASQUAR.TST except serial digital outputs and inputs. SQUARE.WAV must be downloaded with the 1G option for this test to function

DELAYDUT.TST test for measuring delay time through 3-head tape recorder, satellite transmission link, or other similar audio path. Requires BUR option, uses FFTSLIDE.DSP. Sends 1 kHz burst to device input every 500 milliseconds, triggers DSP acquisition from burst envelope, acquires signal from device output and displays in envelope mode

FFT-W&F.TST Requires W&F option in analyzer. Acquires unweighted signal from W&F discriminator using 1 kHz sample rate, performs high-resolution FFT analysis of flutter components

FFT22K.TST General-purpose two-channel acqusition and FFT analysis test setup, 48 kHz sample rate

FFT80K.TST General-purpose two-channel acqusition and FFT analysis test setup, 192 kHz sample rate

FFTGEN.DSP version 2.10 revised program

FFTSLIDE.DSP version 2.10 revised program

RES1K-F.TST Measures residual THD+N of System One via GEN-MONITOR path, DSP acquires signal following analog notch filter, performs FFT and displays spectrum analysis of residual distortion

RES1K-T.TST Measures residual THD+N of System One via GEN-MONITOR path, DSP acquires signal following analog notch filter, displays residual distortion in time domain

SQUARE.WAV 1 kHz square wave generated by MAKEWAVE, for use by FFTGEN.DSP

SWEEP1.TST Uses FFTSLIDE.DSP, to be used by loading waveform SWEEP1.WAV (stepped sinewave sweep) with the "1" option, pressing <F6> to perform FFT spectrum analysis at several different FFT START times by use of SOURCE-2 "nested sweep"

SWEEP1.WAV Stored acquired signal of sinewave sweeping through several frequencies from 2 kHz-10 kHz. For use with SWEEP1.TST and <F6>

WAVE22K.TST General-purpose dual-channel waveform display test (storage oscilloscope mode), 48 kHz sample rate for 22 kHz bandwidth

WAVE80K.TST General-purpose dual-channel waveform display test (storage oscilloscope mode), 192 kHz sample rate for 80 kHz bandwidth

B.7.1. Calibration Waveforms

CALCCIF.WAV 13k/14kHz 1% distortion reference CCIF IMD signal with -40 dB 1 kHz product

CALDIM.WAV standard DIM (dynamic-transient IMD) 1% distortion reference signal with -40 dB lf product

CALDIN.WAV standard DIN IMD 250 Hz/8 kHz reference signal with 1% distortion 2nd order products

CALSMPTE.WAV standard SMPTE IMD 60 Hz/7 kHz reference signal with 1% distortion 2nd order products

CALTHDK2.WAV 1 kHz 1% THD distortion reference signal with -40 dB @ 2 kHz

CALTHDK3.WAV 1 kHz 1% THD distortion reference signal with -40 dB @ 3 kHz

CALTHDK9.WAV 1 kHz 1% THD distortion reference signal with -40 dB @ 9 kHz

CALW&F.WAV 0.781% IEC unweighted W&F reference signal with 5.8XXX Hz flutter rate

CCIF.TST test file for System One IMD analyzer CCIF mode calibration. Download CALCCIF.WAV with the "1G" option and press <F9>. Results will be evaluated agains IMD limits

DIM.TST test file for System One IMD analyzer DIM mode calibration. Download CALDIM.WAV with the "1G" option and press <F9> to evaluate against IMD limits

DIN.TST test file for System One IMD analyzer SMPTE/DIN mode calibration. Download CALDIN.WAV with "1G" option, press <F9> to evaluate results against IMD limits. Identical to SMPTE.TST

IMD-LOW.LIM -41 dB IMD lower limit (target -40 dB, tolerance æ1 dB)

IMD-UP.LIM -39 dB IMD upper limit (target -40 dB, tolerance æ1 dB)

SMPTE.TST test file for System One IMD analyzer SMPTE mode calibration. Download CALSMPTE.WAV with "1G" option, press <F9> to evaluate results against IMD limits. Identical to DIN.TST

THD-LOW.LIM -40.5 dB THD lower limit (target -40 dB, tolerance æ0.5 dB)

THD-UP.LIM -39.5 dB THD upper limit (target -40 dB, tolerance æ0.5 dB)

THD.TST test file for System One THD+N analyzer calibration. Download any of the three waveforms CALTHDK2.WAV, CALTHDK3.WAV, or CALTHDK9.WAV with the "1G" option and press <F9> to evaluate measurement against THD limits

W&F-LOW.LIM 0.781% -5% lower limit for W&F.TST

W&F-UP.LIM 0.781% +5% upper limit for W&F.TST

W&F.TST test file for System One W&F analyzer IEC unweighted mode calibration. Download CALW&F.WAV with "1G" option, wait five seconds for stabilitztion of W&F analozer, and press <F9> to evaluate measurement against W&F limits

Note that MAKEWAVE.EXE and MAKEDIST.EXE are distributed on the "Utilities and Equalization" diskette furnished with all System Ones.

C. FASTTEST.DSP AND FASTTRIG.DSP PROGRAMS FOR FAST AUDIO TESTING

C.1. Overview

FASTTEST.DSP and FASTTRIG.DSP were developed for very rapid frequency response, distortion, noise, inter-channel phase, and stereo separation testing of audio systems and equipment. Both programs operate by digitally generating a multi-sinewave signal as stimulus, then performing an FFT analysis of the resulting signal at the output of the device or system under test. The multi-sinewave signal is defined by waveform (.WAV) files which must be downloaded to the generator buffers of FASTTEST.DSP or FASTTRIG.DSP. A number of standard equal-amplitude multitone waveforms with from five to sixty sinewaves are furnished with the programs. These furnished waveforms have the sinewaves distributed approximately logarithmically across the audio spectrum. A set of utility programs is also included which permit the user to create custom multitone signals with any desired set of user-specified frequencies at user-chosen amplitudes and phase relationships. Multitone signals can be used in the digital domain for testing digital devices. More commonly they are converted to the analog domain by the D/A converter in the DSP module and fed out through the analog generator output stage for testing analog devices. ***FASTTEST and FASTTRIG have no "built-in" default waveform or signal generation capability. An appropriate .WAV file must be downloaded into the generator memory buffers before either program can generate any signal.***

The analog output signal from the device under test is applied to System One, typically via the analog analyzer input circuits and signal conditioning. It is converted to digital domain in the DSP A/D converters, and an FFT is performed. The spectrum

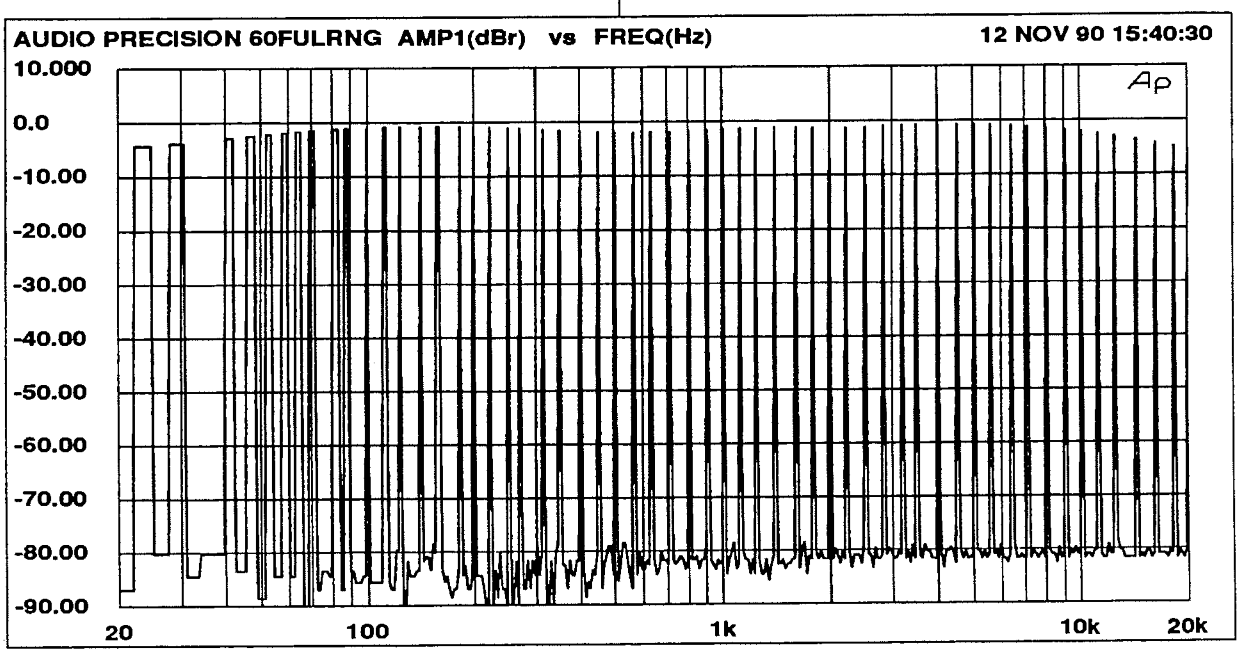


Figure C-1 60-Sinewave Signal Fed Through Equalizer, High-Resolution FFT of Output

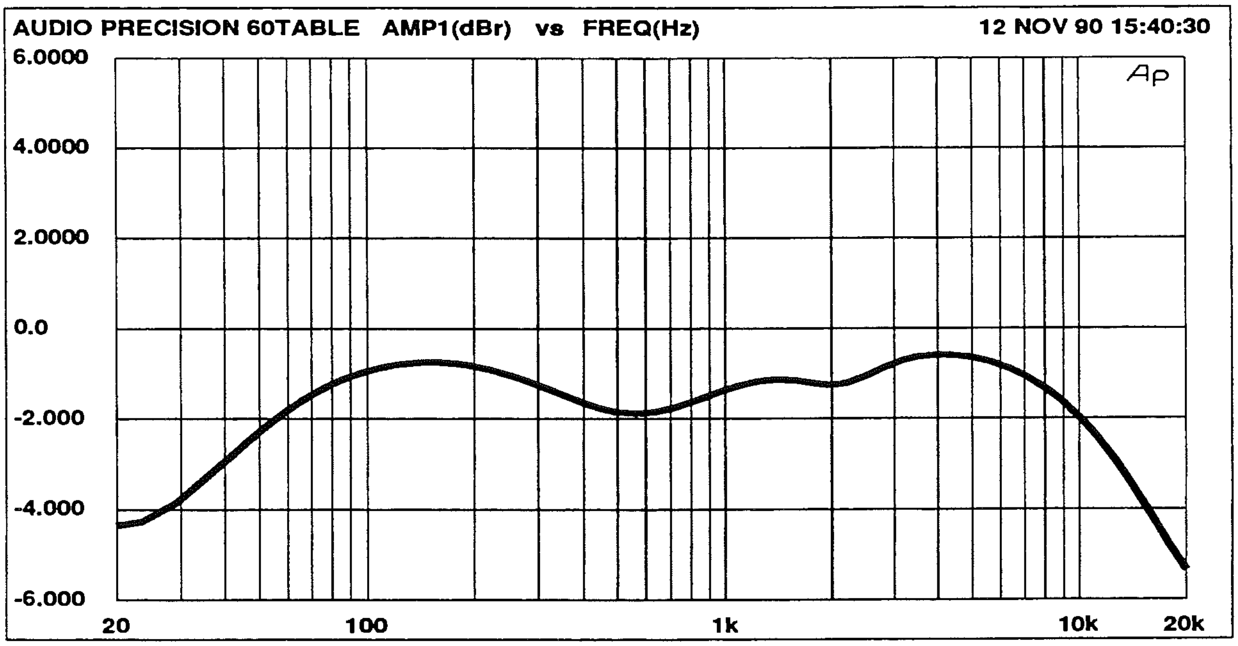


Figure C-2 60-Sinewave Signal Fed Through Equalizer, Sweep Table ON to Plot Frequency Response Only. Vertical Scale Expanded.

of a 60-tone signal created with this technique is shown in Figure C-1, after passing through an equalizer under test. Depending upon how the FFT results are processed and displayed, the final result may be frequency response, distortion, noise, phase, or stereo separation (crosstalk). For example, by plotting only the amplitudes of FFT bins which contain the exact fundamental frequencies of the sinewaves in the signal, the resulting display becomes a conventional frequency response graph as shown in Figure C-2. The vertical scale has been expanded for better display of response. This graph is exactly equivalent to a line connecting the spectral peaks of the Figure C-1 graph. Conventional upper and lower limit files can be applied to such a test for go/no-go frequency response tests. The .SWP file capability of S1.EXE is used to specify exactly which bins should be interrogated.

The principal advantage of this multitone technique is speed. A 60-point (1/6 octave resolution) frequency response graph can be obtained in about two seconds using a 386-based computer, compared to a typical 15-20 seconds required for a 60-point sinewave sweep. A 16-point (1/2 octave) resolution measurement can take less than one second. For testing broadcast transmissions, tape recording quality, and similar applications, a burst of multitone signal as short as 0.25 seconds can be captured and will provide all the performance measurements listed above during the following several seconds.

Total distortion and noise can be measured by selecting the FASTTEST or FASTTRIG DISTORT mode in which only the energy in FFT bins between the fundamental signals is measured and graphed. If necessary, it is also possible to test for harmonic distortion and intermodulation distortion individually. For distortion testing, the original signal should be properly constructed so that sinewave fundamentals are not generated at low integer multiples of lower frequency sinewaves. Such signals would obscure harmonics generated by the lower frequency signals. Noise evaluations can be made by measurements of FFT bins where no fundamental or distortion products fall. Inter-channel phase of stereo signals is measured by the DSP subtracting the left channel FFT phase measurements from the right channel data. Stereo separation can be measured by sending left and right channel waveforms with some frequencies different on the two channels and measuring the amplitude in the opposite channel at the unique frequencies. Stereo separation measurements on analog domain devices require external dual D/A converters, since System One contains only one D/A converter and only one analog generator output stage.

The signal, once acquired and stored in DSP memory, can be repeatedly evaluated (usually in a procedure) by a succession of .TST files, each with its own panel settings and .SWP and .LIM files. Thus, for example, a broadcast system could be tested with a short burst of as little as 0.25 seconds of multitone signal inserted during programming and acquired at many measurement locations. Normal program material would follow immediately. Meanwhile, several different parameters of the signal could be automatically measured at each location from the acquired signal during the following few seconds, displayed if desired, compared to limits, and possibly printed. The complete acquired signal could even be stored to disk for further evaluation later or at another location.

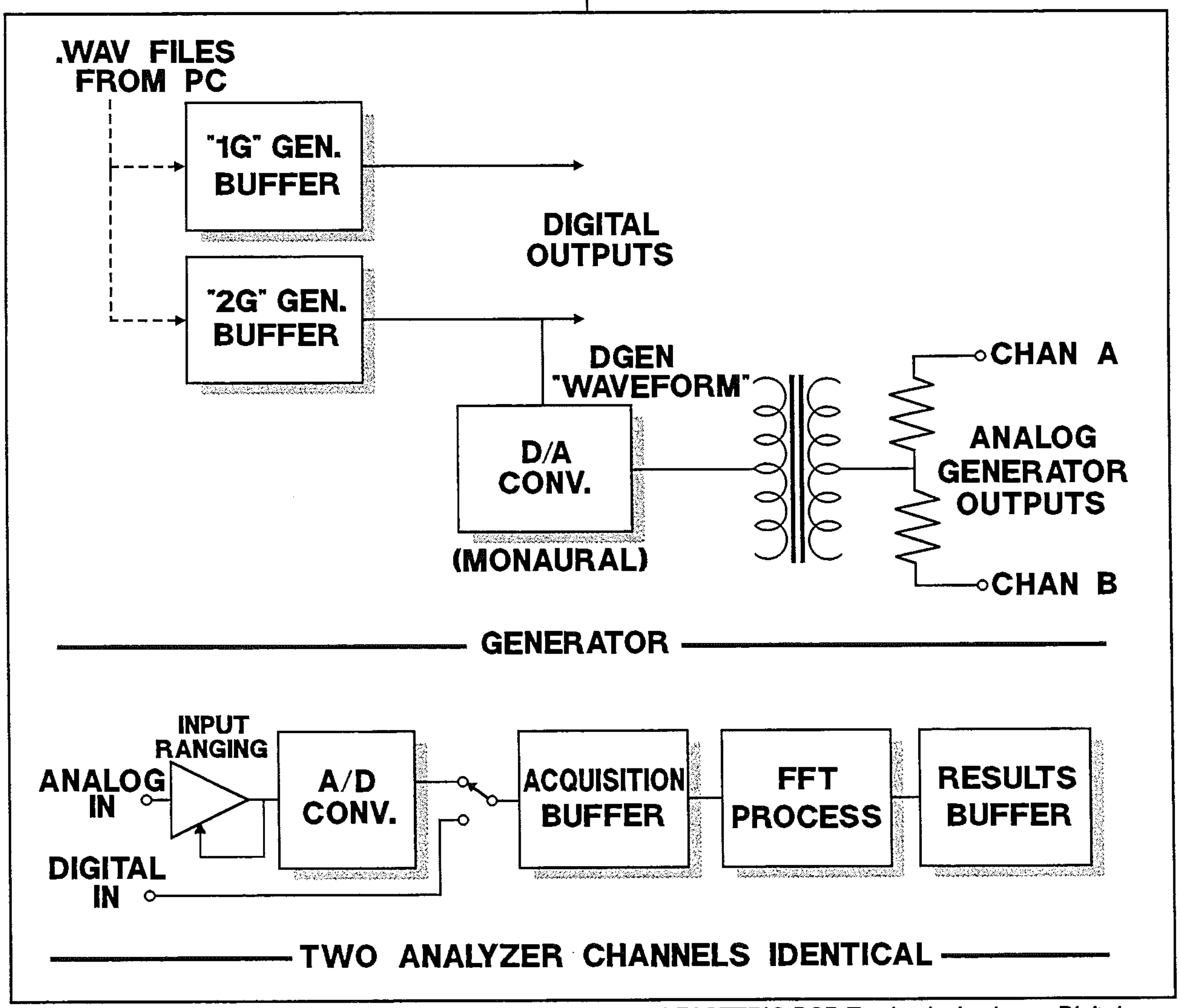


Figure C-3 Simplified Functional Diagram, FASTTEST.DSP and FASTTRIG.DSP Testing in Analog or Digital Domains

C.2. FASTTEST vs FASTTRIG; Which Program to Use

FASTTEST and FASTTRIG are similar in general concept and share many features. FASTTEST is intended principally for real-time stimulus-response testing of electronic devices. Such devices typically do not shift the frequency of the signal passing through them, have little or no time delay between input and output, and have input and output within convenient cable connection distance from a single System One.

FASTTRIG is intended for applications when the multitone signal is being transmitted from a distant point or has been previously recorded and is to be measured when played back. To accomplish testing under these situations, FASTTRIG has the ability to trigger signal acquisition and a set of measurements only when it recognizes that the incoming signal corresponds to the signal stored in its internal generator buffers. FASTTRIG also corrects for any shift in frequency such as commonly occurs between recording and playback, or when the "transmitting" System One or digital recorder and the "receiving" System One have slightly different quartz crystal reference frequencies.

FASTTRIG requires full DSP memory; an SYS-222 with MEM option or an SYS-322. FASTTEST will run at reduced frequency resolution on an SYS-222 (System One + DSP) without MEM option but requires either the MEM option or an SYS-322 (System One Dual Domain) for full resolution.

In the remainder of this chapter, most information will refer to both programs. Differences will be clearly identified.

C.3. Synchronization and Selectivity

By making the DSP generator waveform buffer length exactly equal to the acquired signal buffer to be FFT transformed, or an exact integral sub-multiple such as 1/2, 1/3, 1/4, etc., the generated signal is synchronous with the FFT analysis. With synchronous generation and acquisition, no window function is required for the FFT and no energy "spillover" occurs to adjacent FFT bins. Thus, a fundamental signal or distortion product amplitude may be measured in one bin with no interference from another signal in an adjacent bin. Figure C-4 illustrates sinewaves at 1001 and 1007 Hz. Several distortion products are visible in nearby bins (bin spacing is 2.93 Hz in this example), but no energy from the fundamentals spills into adjacent bins since no window is used. Actual analyzer frequency resolution and generator frequency-setting resolution will depend upon the generator waveform file record length and corresponding analyzer acquisition buffer length. Maximum available resolutions occur with an 8192-sample generated signal and 16,384-sample analyzer buffer. In this case, analyzer resolution is approximately 2.93 Hz and the generated signal resolution is approximately 5.86 Hz. Faster testing results can be obtained by using shorter buffers, with a corresponding loss of frequency resolution.

C.4. Sweep Tables

A sweep table (.SWP file) consists of a list of SOURCE-1 values which will be used by the .TST file. With FASTTEST and FASTTRIG, these SOURCE-1 values will always be frequency values. The NAMES SWEEP command is used to establish a linkage between the .TST file and the .SWP file which it should use. Selecting TABLE ON below SOURCE-1 in the .TST file instructs S1.EXE to use the frequencies in the sweep table rather than computed intermediate points between the START and STOP frequencies entered on the SWEEP DEFINITIONS panel. When the sweep table consists of the exact sinewave fundamental frequencies of the multitone signal, the computer will then request amplitude data from the DSP only at those exact frequencies.

Sweep tables are furnished for the fundamental frequencies of all the furnished waveforms. These fundamental-frequency sweep tables are used for frequency response, noise, phase, and total distortion measurements by selecting the proper modes on the FASTTEST panel and SWEEP DEFINITION panel. Users can create custom sweep files for custom

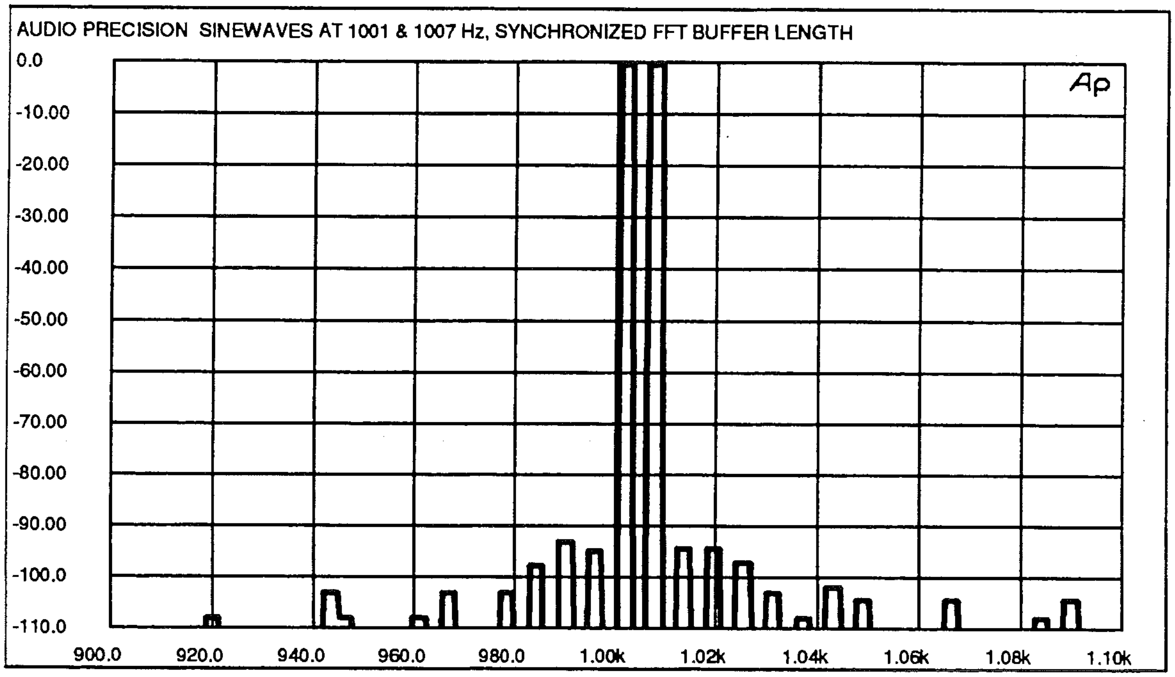


Figure C-4 High-Resolution FFT of FASTTEST.DSP-Generated Sinewaves at 1001 and 1007 Hz, Illustrating Zero Energy Spillover Due to Synchronous Relationship of Generation and Transform

waveforms or for applications such as measuring harmonics or intermodulation products rather than total distortion and noise.

C.5. Features Common to Both FASTTEST.DSP and FASTTRIG.DSP

Figure C-5 shows the DSP panel fields common to both FASTTEST and FASTTRIG. As with the other FFT programs, the two top display fields are indications of the real-time peak input level to verify the presence of signal. They also help the user avoid overload when the DSP BNC input connectors are used or analog analyzer input ranges are fixed.

The two phase fields are not active in panel mode, but FFT phase information can be graphed at DATA-1 or DATA-2.

The FREQ RES field sets the effective measurement resolution in RESPwW+F and DISTORT modes (see below). This is used when testing analog tape recorders or other devices which may produce flutter, frequency modulation, or small amounts of frequency shift. The FREQ RES field also determines how far away from nominal frequencies FASTTRIG's triggering mode will search when comparing acquired signals with the stored waveform.

The DGEN AMPL field controls the amplitude of signal presented to the digital outputs in a System One Dual Domain when DSP OUTPUT is SERIAL or PARALLEL. When the DSP panel OUTPUT selection is D/A and WAVEFORM is ***not*** DGEN on the analog generator panel, the DGEN AMPL field controls the amplitude of the analog signal at the BNC connector labeled DSP OUTPUT D/A. DGEN AMPL has no effect when DGEN is the selected waveform and the DSP OUTPUT choice near the bottom of the panel is D/A. In this latter case, amplitude from the analog generator output connectors is controlled only by the AMPLITUDE field on the analog generator panel.

The left-hand field on the MEASURE line permits selection among four post-processing measurement modes. With NORMAL selected, the DSP will furnish to the computer only the amplitude of the single FFT bin (in both channels) which contains the frequency requested by S1.EXE. Typically, this requested frequency will be a sinewave fundamental frequency from the .SWP file attached to the test file, and the graphed result will be frequency response. No "peak picking" or searching of bins between requested frequencies takes place in NORMAL mode.

With RESPwW+F (Response with Wow and Flutter) selected, the DSP will examine the amplitude of all FFT bins within ±(FREQ RES) percent of the frequency requested by S1.EXE. It computes the RSS (root sum square) of these bin amplitudes, and send the result to the computer. This process compensates for the fundamental energy spread out into near-by bins due to the frequency modulation sidebands caused by wow and flutter in the device under test. This mode should be used for measuring frequency response of tape recorders and turntables. If a FREQ RES value of zero is used the results will be identical to NORMAL mode except for a 0.01% minimum FREQ RES value not reflected in the panel field.

The DISTORT mode causes the DSP to perform an RSS summation of the amplitudes of all FFT bins between the previously-requested frequency and the currently-requested frequency, but not including those two frequencies. Thus, DISTORT mode sums all harmonic distortion products, intermodulation distortion products, and noise over the span between each adjacent pair of fundamental frequencies. The result is furnished to the computer to be plotted as the value of total distortion and noise at the currently-requested frequency. If a value greater than zero is entered in the FREQ RES field, the DSP ignores the bins within ±(FREQ RES) per-

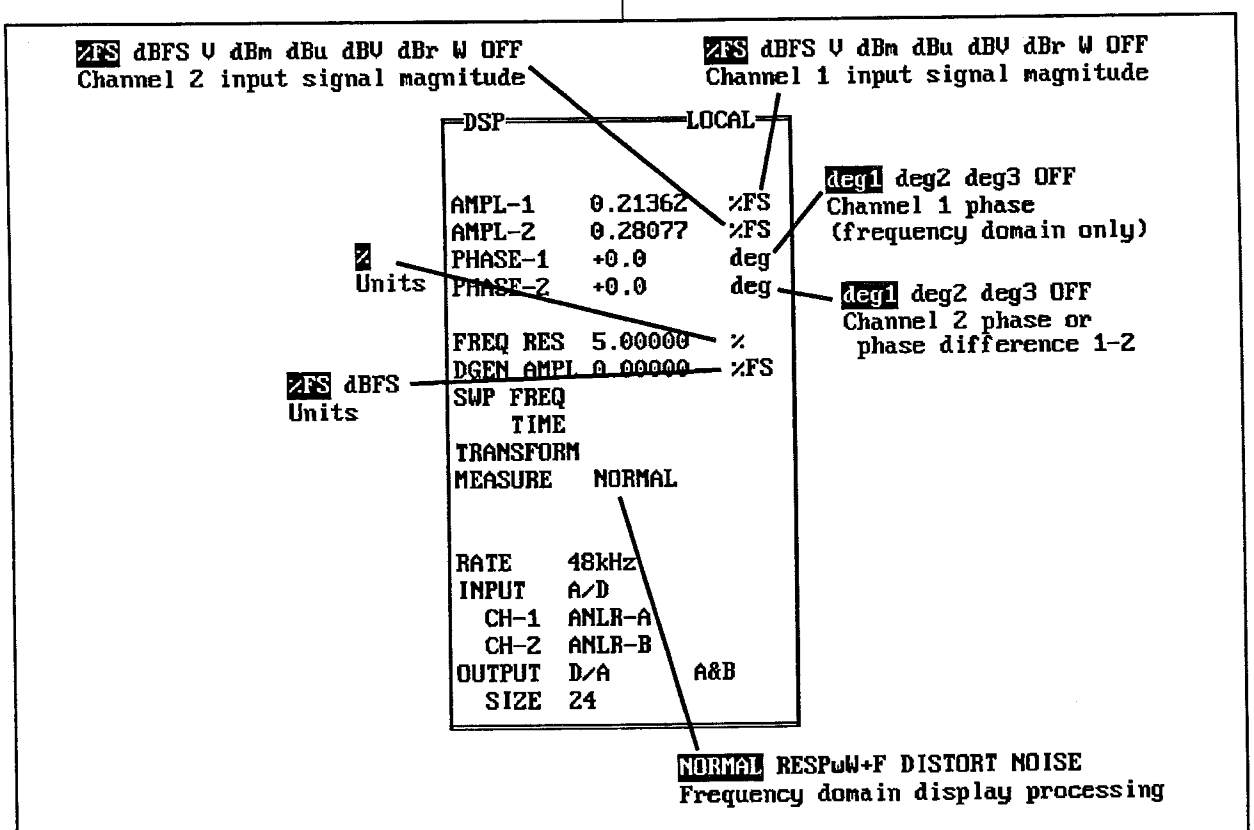


Figure C-5 DSP Panel Features Common to Both FASTEST.DSP and FASTTRIG.DSP cent of each requested frequency so as not to include flutter sideband energy into the distortion and noise summation.

NOISE mode also performs an RSS summation of bin amplitudes between the previously-requested and presently-requested frequencies. However, NOISE mode only sums the bins into which harmonic and intermodulation products cannot fall, thus creating a spectrum analysis of noise in the presence of signal. These "empty bins" exist when the analyzer record length is twice the generator record length. Distortion products can only fall at multiples of the generator's waveform repetition rate. Since the analyzer's repetition rate is half that of the generator, it will have twice the resolution. Thus, every alternate analyzer bin cannot contain generator-related products. These alternate bins are the ones measured in NOISE mode.

FASTTRIG automatically selects an analyzer record length twice the downloaded generator record length. With FASTTEST, the operator is responsible for selecting a TRANSFORM record length twice as long as the generator signal used if NOISE mode is to function properly.

The NORMAL, RESPwW+F, DISTORT, and NOISE modes are discussed in more detail below.

C.5.1. Frequency Response Testing

C.5.1.1. Normal Response Mode

NORMAL should be chosen in the first field following the MEASURE label on the panel for most frequency response measurements, particularly when measuring electronic audio devices such as amplifiers, equalizers, consoles, transmitters, and most other devices which neither shift frequency nor add wow and flutter. In NORMAL mode, the DSP module simply sends to S1.EXE the FFT bin amplitude of one or both channels, as selected, at each frequency requested by S1.EXE. With a sweep table which contains the fundamental frequencies attached and turned on, the resulting graph is a frequency response measurement. Figure C-6 is a frequency response measurement of the encoder section of a noise reduction system for tape recording, using a 31-tone signal (ISO31.WAV).

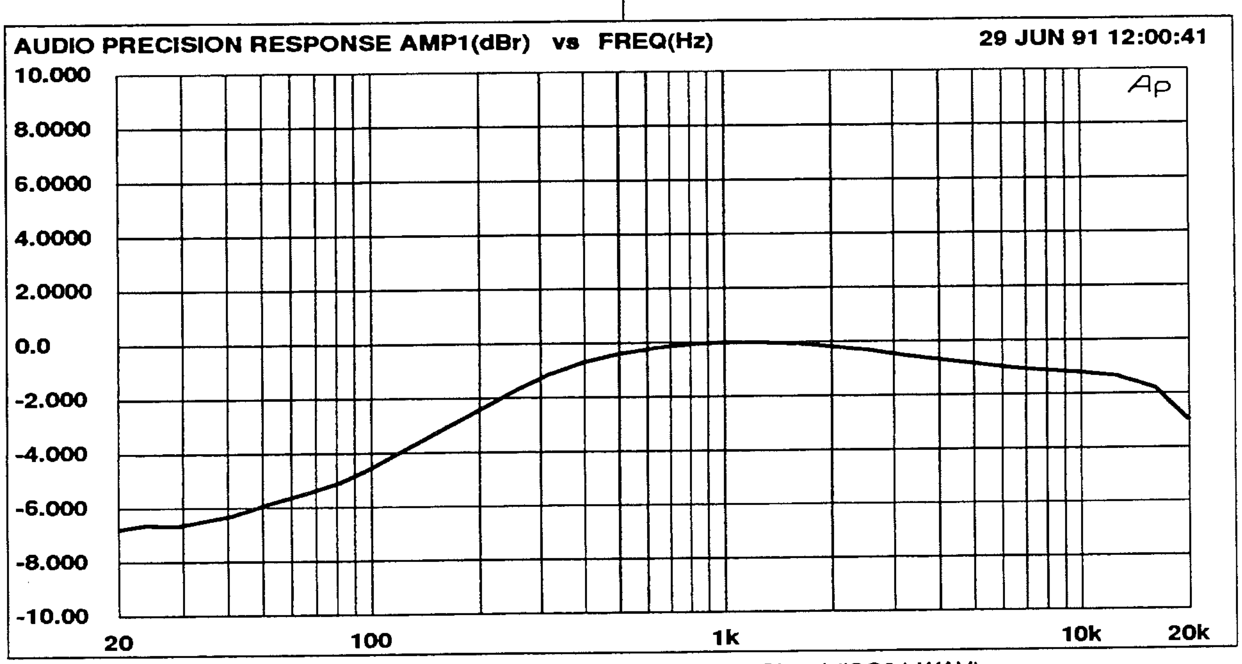


Figure C-6 Frequency Response, Noise Reduction System, 31-Tone Signal (ISO31.WAV)

Note that FASTTEST.DSP and FASTTRIG.DSP, when used ***with the sweep table off***, do a "peak picking" operation. This insures that the highest amplitude signal components are graphed whether they fall exactly at a computer-requested frequency or between requested frequencies. With TABLE OFF, the DSP searches all FFT bins between the last-requested and the currently-requested frequency and sends the amplitude of the highest value FFT bin to the computer. For FASTTEST and FASTTRIG with the step table ON, this "peak-picking" is defeated and only the amplitude of the requested bin is sent to the computer.

C.5.1.2. Response in the Presence of Wow & Flutter

Analog tape recorders and turntables add wow and flutter to recorded and reproduced signals. Small imperfections in the transport mechanisms cause frequency modulation of each signal frequency component. Frequency modulation creates families of sidebands around each original signal. All the energy in these sidebands is taken from the original signal amplitude "carrier", reducing its amplitude. If the sideband spacing is such that significant sideband energy falls outside the FFT bin containing the fundamental (original) frequency component, the amplitude of that fundamental will be reduced by the amount of energy falling outside the bin. In fact, certain combinations of wow and flutter frequency and magnitude could result in ***no*** energy being present in the target bin. This is identical to the Bessel null property familiar in frequency modulated radio.

The FFT analysis bin width is only about 2.93 Hz with the 48 kHz sample rate and the MAXIMUM transform length of FASTTEST or with an 8k generator waveform downloaded into FASTTRIG (assuming full memory). Therefore, typical flutter amounts and flutter frequencies produce sidebands which spread through many bins, especially at high audio frequencies. If the amount of flutter is significant, the effect (in NORMAL mode) is to cause an apparent amplitude reduction when only the bin amplitude containing the original frequency component is evaluated. This is typically seen as a high-frequency roll-off, since the sidebands (assuming low flutter frequencies) occupy approximately a constant percentage bandwidth around each funda-

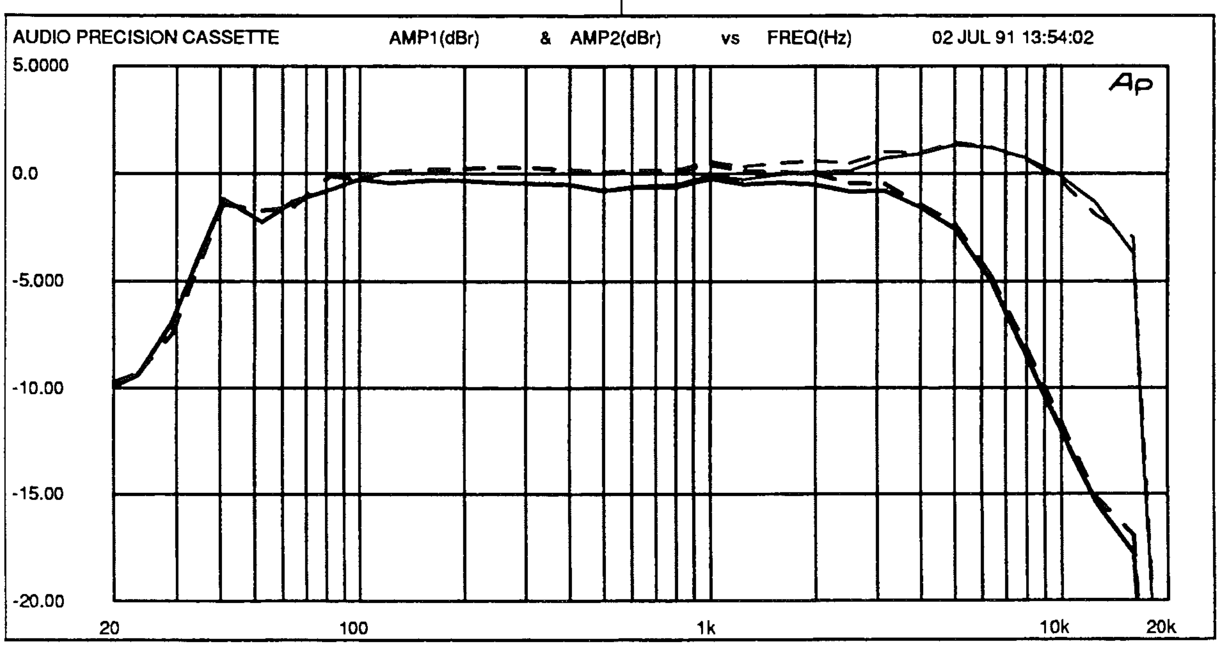


Figure C-7 Frequency Response, Cassette Stereo Tape Recorder. Solid Traces Left Channel, Dashed Traces Right Channel. Darker Traces NORMAL Mode, Lighter Traces RESPwW+F Mode.

mental frequency. They therefore spread through more and more of the linearly-spaced FFT bins at high frequencies.

In order to provide accurate frequency response measurements in the presence of wow and flutter, these programs incorporate the MEASURE RESPw-W+F mode (Response with Wow and Flutter) and the FREQ RES field. At each fundamental frequency requested by the computer, the DSP computes the root-sum-square (RSS) of all FFT bin amplitudes within ±(FREQ RES) percent of that frequency. This computed value is sent to the computer to be graphed. The FREQ RES field is located below the phase display fields in the upper part of the panel. In effect, the DSP adds back to each original frequency component the energy which wow and flutter subtracted from it and placed into FM sidebands. For wow (which is low-frequency by definition) and lower-frequency flutter, the frequency modulation percentage deviation at each frequency is approximately equal to the wow and flutter percentage of the tape machine or turntable. This is a minimum value for FREQ RES.

In most practical applications with a reasonable signal-to-noise ratio, the only negative effect of using larger values for FREQ RES during frequency response measurements is an upwards shift of each measured point by 0.1 dB or less due to adding the small amounts of noise from nearby FFT bins to each signal component. Thus, values of 1% or 2% are appropriate "all-purpose" selections for FREQ RES even with tape machines known to have less than 0.1% wow and flutter. Figure C-7 shows frequency response of a cassette tape recorder with both NORMAL (darker trace) and RESPwW+F modes. The apparent high-frequency roll-off in NORMAL mode is clearly visible as more and more flutter components fall outside the bin containing the fundamental frequency.

C.5.2. Total Distortion Testing

When a multitone signal is passed through a device which is truly linear and noise-free, the output signal would consist of exactly those same frequencies and no others. In practice, non-linearities in the device will produce harmonics of each input signal and intermodulation products of each combination of two or more input signals. Noise generated in the device under test produces energy across the spectrum, falling into all FFT bins. Interfering signals such as power-mains-related hum or the magnetic field from a nearby video or computer monitor may also be coupled into the device under test at specific frequencies. A measurement of all the energy falling into bins ***other than at the original sinewave signal frequencies*** is thus a measurement of the total distortion and noise produced in the device.

The MEASURE DISTORT (distortion) mode of FASTTEST and FASTTRIG calculates the RSS (root-sum-square) amplitude of all the bins ***between each pair*** of original sinewave frequencies. This computed value is sent to the computer to be plotted as the "distortion plus noise" at the second frequency. For example, assume a sweep from low to high frequency and original signals of 100 Hz, 250 Hz, and 450 Hz. With MEASURE DISTORT mode selected, when the computer requests a measurement at the 100 Hz frequency, the DSP will calculate the RSS amplitude of all bins above the 0 Hz (DC) bin to the bin just below 100 Hz. This value is sent to the computer to be plotted as total distortion and noise at 100 Hz. The computer then requests a measurement at 250 Hz and the DSP computes the RSS amplitude starting with the first bin above 100 Hz up through the bin just below 250 Hz. This value is sent to the computer as total distortion and noise at 250 Hz, etc.

The resulting plot is thus representative of the total energy across each portion of the spectrum, excluding the energy at the original input frequencies. Conceptually, this technique is similar to the THD+N method with a single sinewave, in which the fundamental is removed by a bandreject filter and all remaining energy measured. In DISTORT mode, the fundamentals are not removed; those bins are simply ignored by the DSP during its calculations. If the highest frequency in the signal is not at or near half the sample rate and if distortion and noise measurements are desired above the top fundamental frequency, half the sample rate can be added as the highest entry in the sweep table used in DIS-

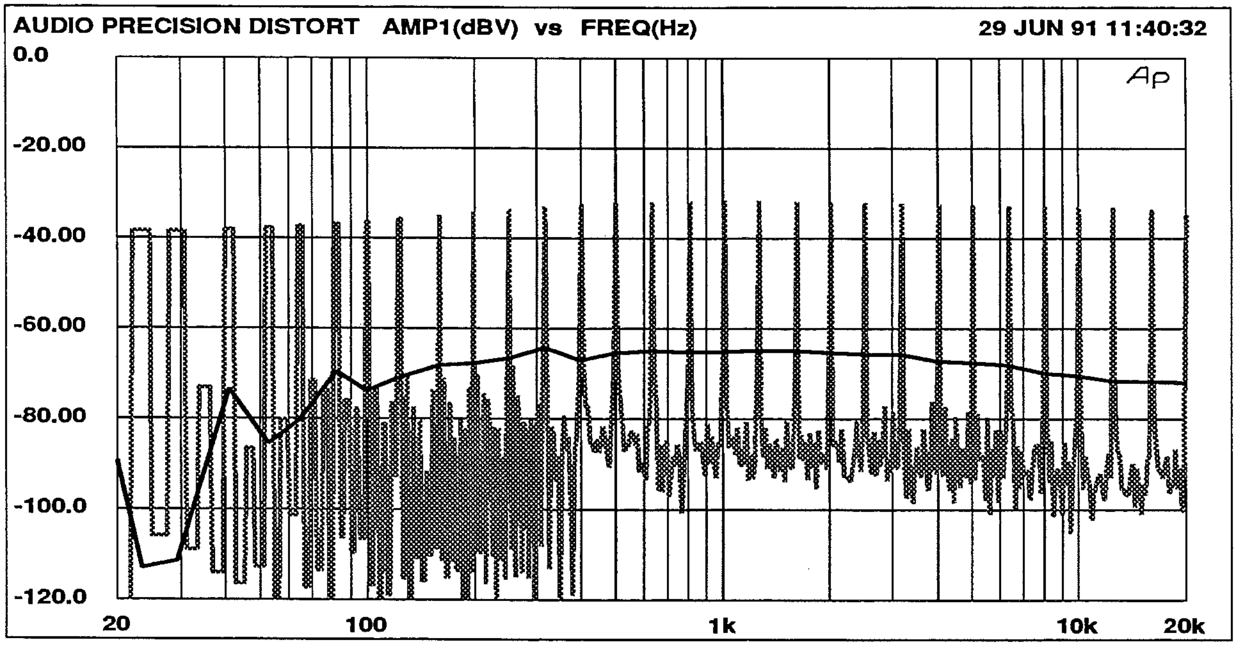


Figure C-8 Total Distortion and Noise (Darker Trace), Noise Reduction Unit, 31-Tone Waveform. Lighter Trace is High-Resolution FFT of Same Acquired Signal

TORT mode. The DSP will then return at this value the RSS value of amplitude in the bins above the highest fundamental frequency up to half the sample rate.

Figure C-8 shows (darker trace) a total distortion and noise measurement of the encoder section of a noise reduction unit. The lighter trace is a high-resolution FFT (TABLE OFF) from the same acquired signal. Note that the darker trace represents, at each fundamental frequency, the root-sum-square amplitude of all the bins above the next lower fundamental and below the fundamental at which it is plotted (assuming that the sweep table runs from low to high frequency).

If any value greater than zero is entered into the FREQ RES field when MEASURE DISTORT mode is used, the RSS calculations of distortion and noise will exclude the band of bins within ±(FREQ RES) percent of each input frequency. This permits use of the MEASURE DISTORT mode with tape recorders without including normal flutter sidebands into the distortion and noise calculations and graphs.

C.5.3. Noise Testing

When the generated waveform repeats at the 8192 sample length period, assuming the 48 kHz sample rate, all generated signals must fall at exact integer multiples of a basic 5.8593772 Hz frequency. All harmonic and intermodulation distortion products due to generated signals also fall at exact integer multiples of that same basic frequency. The analyzer, however, has a maximum acquisition buffer length of 16,384 samples. It therefore performs FFT analysis to a resolution twice that of the generator, with bin centers at multiples of 2.9296875 Hz. Generator-related products can only fall into the half of these bins whose center frequency is evenly divisible by 5.8593772 Hz. Every alternate bin (not divisible by the basic generator frequency) thus remains empty of generator-related signal products and can be used to measure noise in the presence of the multitone signal. This effect can be noted in Figure C-4, where the 1001 Hz and 1007 Hz fundamentals and their near-by intermodulation distortion products all fall into alternate analyzer bins. The remaining half of the analyzer bins only show lower-amplitude noise generated in the device under test. Measuring noise in the pres-

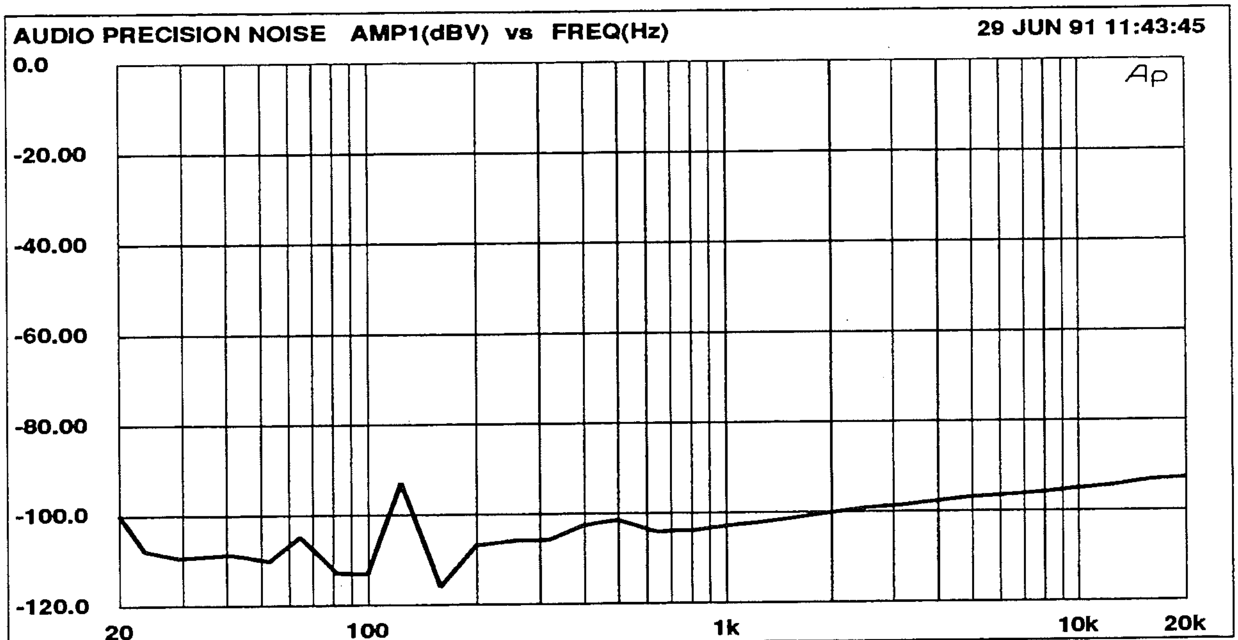


Figure C-9 Noise Measurement, RSS of Empty Bins Between Fundamentals of 31-Tone Waveform, Noise Reduction Unit

ence of signal is not only fast, but yields measurements otherwise impossible to make on compressors, expanders, and other non-linear devices whose gain is different with and without signal. ***Noise mode is not recommended when testing devices such as analog tape recorders which spread the signals into adjacent bins due to flutter.***

NOISE mode is selectable on the MEASURE line of the panels. In NOISE mode, the DSP module computes the root-sum-square amplitude of all these "empty" alternate bins between each pair of frequencies requested by the computer. For NOISE mode operation, the computer uses the same sweep table used for frequency response measurements, containing the list of exact fundamental frequencies which make up the multitone signal. When the computer asks for an amplitude value at the starting frequency (lowest frequency if sweep table runs from low to high), the DSP computes the RSS amplitude of all "empty" bins above DC up to this first fundamental. This is sent to the computer as the value of noise at that frequency. The computer next requests an amplitude value at the second fundamental. The DSP computes the RSS value of all empty bins above the first but below the second fundamental and sends that as the noise value at the second fundamental, etc.

Figure C-9 shows a measurement in NOISE mode of the same noise reduction system encoder shown in several earlier figures. ***NOISE mode can only be used when the analyzer acquisition buffer length is twice the generated signal buffer length in order to achieve "empty" analyzer bins. This ratio of buffer lengths is created automatically by FASTTRIG when the generator buffers are loaded, but must be deliberately achieved by users of FASTTEST by selecting the analyzer buffer to be twice the length of the generator buffer. With maximum memory, the MAXIMUM transform length of the FASTTEST analyzer is 16k, twice the 8k generator buffer length.***

C.5.4. Stereo Phase Testing

FASTTEST and FASTTRIG are also capable of measuring inter-channel phase in stereo audio systems. When DIFF is selected in the right-hand field on the MEASURE line, the PHAS-2 measurement becomes the phase difference between the two channels. Thus, selection of PHA2 at DATA-1 or DATA-2 with DIFF selected on the DSP panel will result in graphs of stereo interchannel phase. This is particularly useful in azimuth alignment of analog tape recorder heads from a multitone signal. Only the fixed-scale degree units are available (deg1 for ±180 °, deg2 for +270/-90 °, and deg3 for 0 to +360 °. Channel 2 is the reference channel for phase measurements. Figure C-10 shows the interchannel phase characteristics of the noise reduction system encoder.

C.5.5. Stereo Separation-Crosstalk Testing

The DSP with FASTTEST or FASTTRIG loaded contains two generator buffers, each 8,192 samples long. Different .WAV files can be downloaded into the two buffers by use of the "1G" and "2G" arguments in response to the query during LOAD WAVEFORM. However, the DSP module contains only one D/A converter (driven from the waveform in the "2G" buffer) and the analog generator contains only one power amplifier and output transformer; the two analog generator outputs are simply "split" from a common signal. Therefore, ***System One cannot generate independent signals in the analog domain at the analog channel A and B outputs.***

FASTTEST and FASTTRIG can simultaneously generate two different signals in the digital domain from the two waveform buffers. External dual D/A converters can be driven from the System One Dual Domain digital outputs to create stereo analog signals. Alternatively, a digital tape can be recorded with different signals on the two channels by using a recorder such as an RDAT. The analog outputs of the RDAT then provide two different signals which can be used for stereo separation and crosstalk testing. Many RDATs provide real-time analog output while being driven with a digital-domain signal and can be used as dual D/A converters.

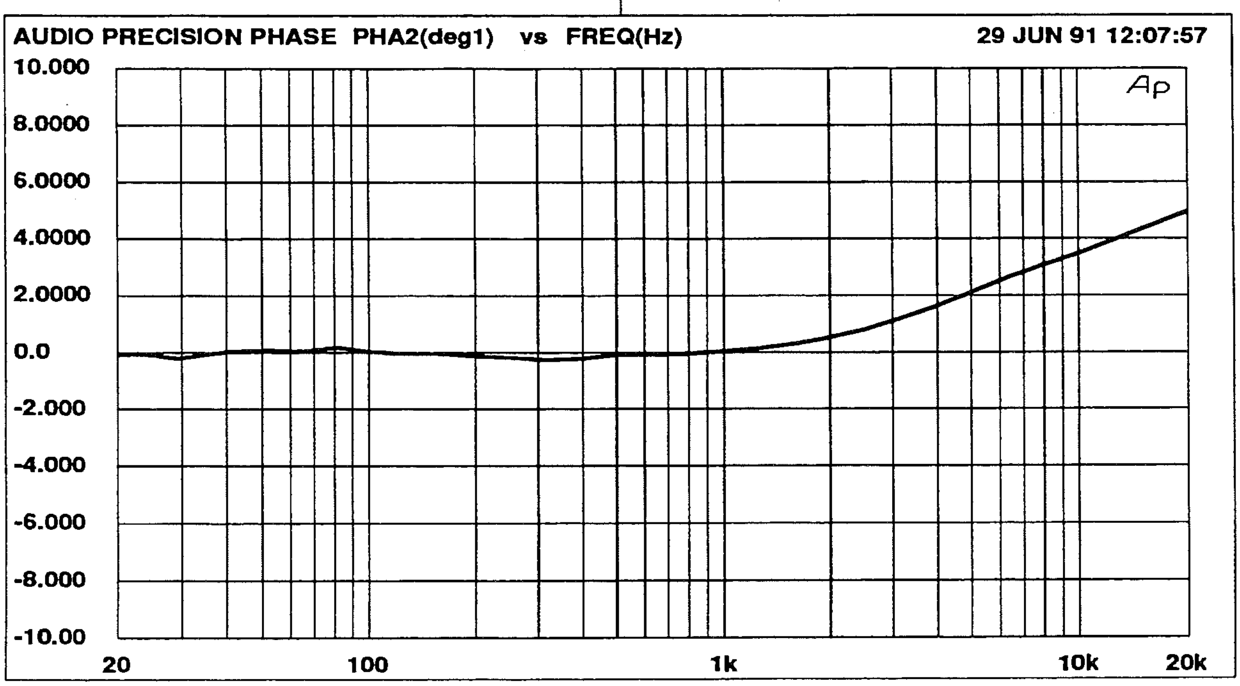


Figure C-10 Stereo Inter-Channel Phase Measurement, Noise Reduction Unit, 31-Tone Waveform. Obtained by Plotting PHAS-2 with DIFF Mode Selected

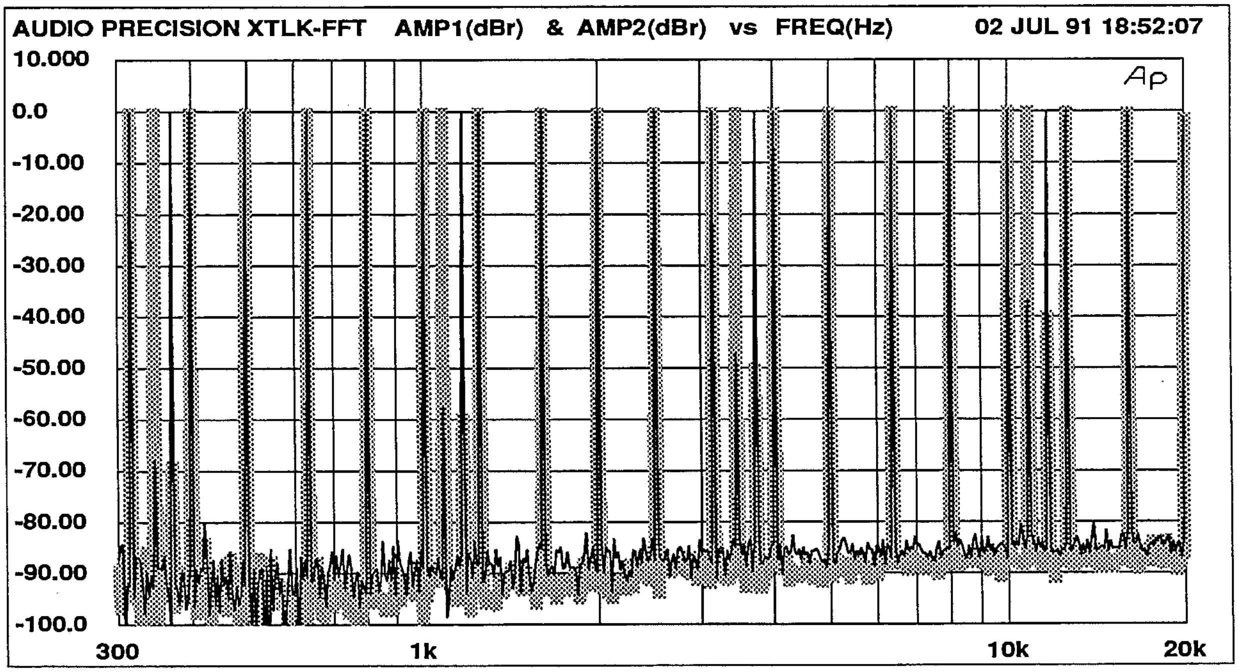


Figure C-11 High-Resolution FFT of Both Channels of XTLK.WAV. Channel One is Dark Narrow Line and Channel Two is Lighter Wide Line. Driving Analog Path of System One via AES/EBU Output of System One Dual Domain and D/A Converters of RDAT.

XTLK.PRO, furnished with FASTTEST, loads the two-channel waveform file XTLK.WAV into the channel one and two generator buffers by use of the "1G,2G" argument following selection of the file name. All the "sweep naming, waveform file loading" procedures furnished with FASTTRIG which have an "S" in their file name also load two-channel ("S") waveforms. XTLK.PRO (for FASTTEST) and TR-STER2.PRO and TR-STER4.PRO (for FASTTRIG) test a stereo device for left-to-right separation and right-to-left separation in addition to the other parameters. In all the two-channel waveforms furnished, the channel one and channel two waveforms are based on regular spacing (usually 1/3 octave) on both channels. In addition to the identical frequencies on both channels, each channel adds additional sinewaves unique to that channel for stereo separation tests.

Figure C-11 shows a high resolution two-channel FFT of the 300-20,000 Hz portion of the two signals of XTLK.WAV, passing through only the D/A converters of an RDAT recorder plus the analog input channels of System One. Both the left channel (narrow dark trace) and right channel (wider, lighter trace) superimpose at all the one-third-octave-spaced frequencies of ISO31. At the channel-unique frequencies, the inter-channel crosstalk of the RDAT can be seen by the difference in height between the dark and light traces. Figures C-12 and C-13 show the L-to-R crosstalk and R-to-L crosstalk measurements obtained by XTLK.PRO. The test device was a stereo noise-reduction unit driven by the D/A outputs of an RDAT machine.

In tests using waveforms which are different on the two channels, the original sweep table (not including any of the added signals for crosstalk testing) is used for the stereo response, phase, and noise testing. A new sweep table with the original fundamental frequencies common to both channels, plus the new crosstalk-stimulation frequencies unique to left and right channels, is used with the total distortion test since all fundamental frequencies on both channels must be excluded from these measurements. Note that the plotted values of total distortion and noise dip slightly in the frequency regions where the additional tones were added. The fre-

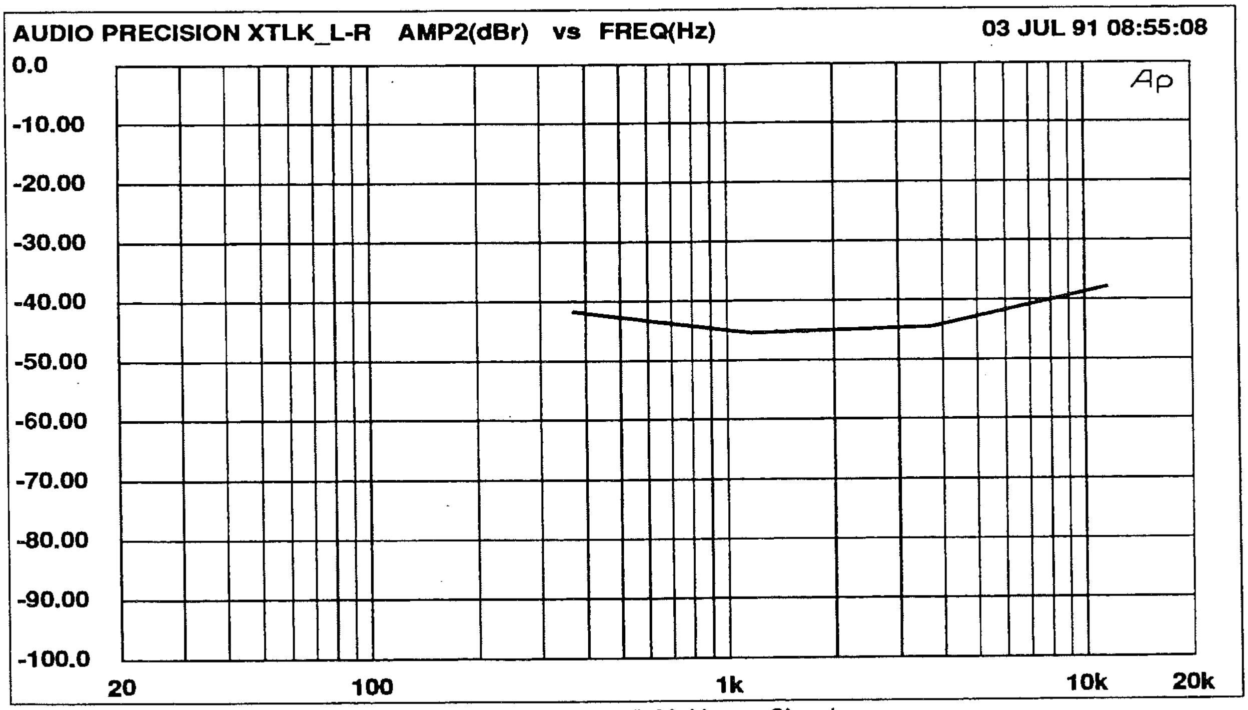


Figure C-12 Stereo Separation Left-to-Right with Crosstalk Multi-tone Signal

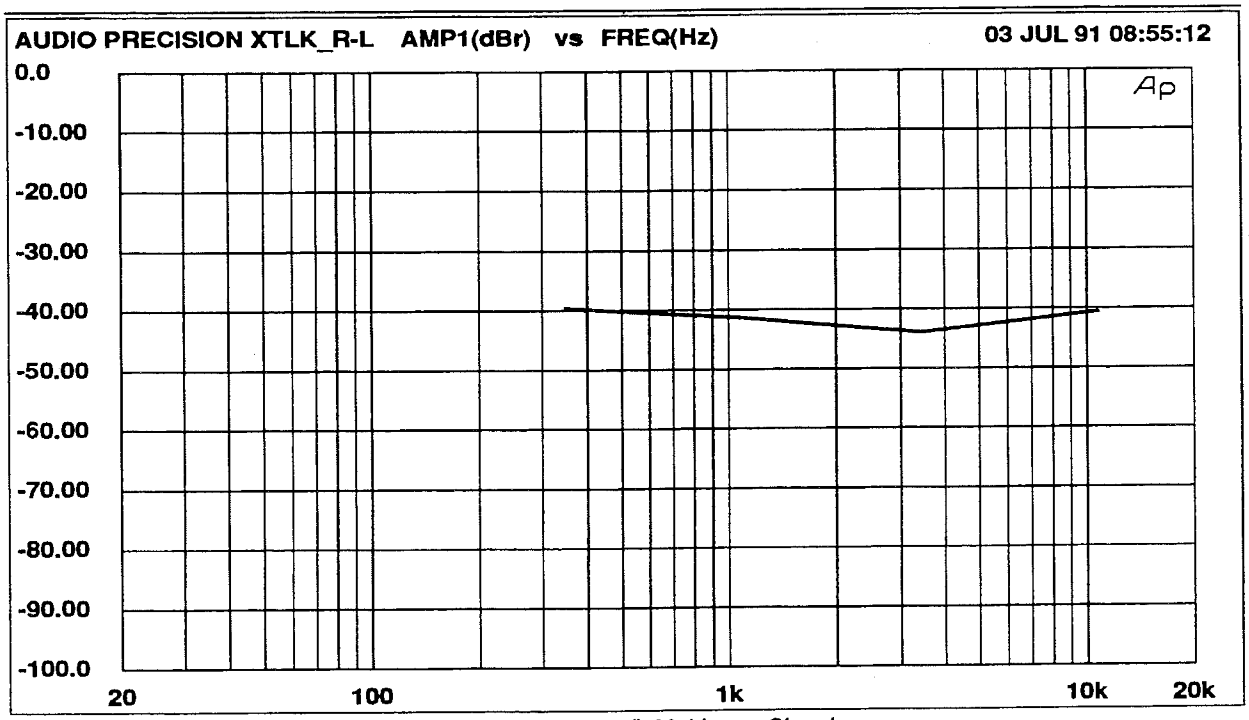


Figure C-13 Stereo Separation Right-to-Left with Crosstalk Multi-tone Signal quency spans summed in these spectral areas are smaller, since the original span has been further subdivided into three sections by adding two near-by crosstalk-testing frequencies. Therefore, the root-sum-square summation of distortion and noise will be lower in these areas since fewer bins are being summed.

To measure stereo separation, different .SWP files must be used from those used for frequency response, phase, total distortion, or noise. A new sweep table listing only the unique left channel tones was used for the L-to-R crosstalk measurement. Another new table listing the unique right channel tones was used for the R-to-L measurement.

To reiterate, ***crosstalk testing with the FAST-TEST-FASTTRIG technique can only be done with digital domain output from System One Dual Domain*** since the system lacks independent analog output channels. Analog domain crosstalk testing requires external dual D/A converters such as those in an RDAT recorder.

C.5.6. Harmonic and Intermodulation Distortion Testing

The DISTORT mode discussed above is recommended for most normal distortion testing. DISTORT mode has the advantage of sensitivity to non-fundamental energy at all frequencies, is fast, and operates with the same .SWP file used for response, noise, and phase. However, harmonic distortion and intermodulation distortion may be separately evaluated if necessary with the panel set in NORMAL mode. It is merely necessary to use a sweep table containing the frequencies at which harmonic products or intermodulation products will fall.

MAKEDIST.EXE, described later in this chapter, can generate ASCII lists of all the harmonic products up through a specified order, or all the intermodulation products of two-tone pairs up through a specified order. These files can be imported into S1.EXE via the LOAD DATA command, then saved as a .SWP file. Naming that .SWP file and using the NORMAL mode of FASTTEST-FASTTRIG will result in graphs of only the specific type of distortion products.

To evaluate harmonic distortion and intermodulation distortion, different .SWP files must be used from those used for frequency response, phase, total distortion, or noise. For example, one .SWP file would be used for harmonic distortion and would contain only the frequencies at which harmonics of the fundamentals will fall (NORMAL mode). A second .SWP file would be used for IMD evaluation and consists of the frequencies where IMD products fall (NORMAL mode). A third .SWP file would contain only fundamental frequencies and would be used for frequency response (NORMAL mode), phase (NORMAL MODE, DIFF, measuring PHAS-2), total distortion and noise (DISTORT mode), and noise (NOISE mode).

C.5.7. Individual Parameter Evaluation

Evaluation of several parameters such as response, distortion, noise, and phase is normally done by a succession of .TST files, each with the appropriate panel mode settings and the appropriate .SWP file attached. This produces a series of conventional graphs, each showing one parameter.

If automatic PASS/FAIL decisions are desired, separate limit (.LIM) files will also be required for each parameter evaluated. If the desired limits are the same at all frequencies, the limit files can be simple one or two-point files as described in the ACCEPTANCE TEST LIMITS chapter of the main System One User's Manual. More complex limit files may be required, such as the case when the original waveform was created with different amplitudes at each different frequency. Such .LIM files should be created starting with the .SWP files with which they will be used. In this manner, both the .SWP and the .LIM files have the exactly-correct frequency values in the first (SOURCE-1) column. Frequency response and inter-channel phase evaluation require both upper and lower limit files. Noise and distortion tests use only upper limit files.

C.5.8. Combined Evaluation

To further reduce data analysis time following acquisition and transform, it is possible to evaluate several amplitude parameters (not including phase) such as frequency response, distortion, noise, and stereo separation all at one time instead of with successive .TST files and <Alt><F6> operations. This is normally practical only when a procedure makes pass/fail decisions and when graphic display or hard copy is not desired, since the resulting signal display and limit files are quite complex and difficult to interpret visually.

For this combined parameter evaluation, DSP panel mode must be NORMAL and the .SWP table used must contain all the frequencies at which measurements are desired. The utility program MAKEDIST.EXE (described later in this chapter) can be run with the appropriate command-line options to create a list including fundamental frequencies, harmonic frequencies up to a specified order, and intermod frequencies up to a specified order, all in one file. Frequencies for noise measurements must be manually inserted.

More complex limit files must then be constructed for combined analysis. Both upper and lower limit files will be required if frequency response is one of the parameters to be measured. These limit files must be made from the .SWP file as a starting point. At each of the fundamental frequencies in the signal, the upper and lower limit files will have amplitude values spaced above and below the expected amplitude value at the desired tolerance. At each of the harmonic, intermodulation, stereo separation, or noise bin frequencies, the upper limit file will be at a low value chosen to ensure adequate quality (40 dB below the fundamental signal amplitudes, for example). The lower limit file will have essentially zero values (-999 dBV) at the distortion, stereo separation, and noise bin frequencies.

C.5.9. Test Setup Files

Several settings and conditions are required for .TST files used with either of these programs. As noted above, it is typical to use a sequence of .TST files in a procedure (.PRO) to extract the response, distortion, noise, etc. data following an initial signal acquisition and Fourier transform.

1. All files used in the same procedure must have either FASTTRIG.DSP or FASTTEST.DSP "attached" by the NAMES PROGRAM command. FASTTRIG and FASTTEST cannot be mixed in the same procedure, since changing between these programs destroys previously downloaded generator waveforms.

2. All files must have an appropriate .SWP file attached by the NAMES SWEEP command and TABLE ON must be selected.

3. To stimulate analog input real-time devices with FASTTEST.DSP, the DSP output setting must be D/A and the DSP OUTPUT must be on (A, B, or A&B are all acceptable). The WAVEFORM DGEN selection must be made on the generator panel. This converts the digitally-generated multitone signal into the analog domain and routes it through the generator's output stage, transformer, and attenuators. Note that, if the <F1> key is used to abort a procedure or turn off the generator instantly, it turns off both the analog generator and DSP generator. ***Both*** must be manually turned on again in order to obtain digitally-generated signal via the DGEN "waveform" and generator analog outputs. Either or both analog output A and B channels may be on or off. In any case, only the signal from DSP generator buffer channel two will be present at the analog outputs. This is because System One contains only one D/A converter and only one analog generator output stage.

4. If stereo separation (crosstalk) testing is required, external stereo D/A converters such as those in an RDAT are required. The DSP panel will then typically be set for SERIAL output rather than D/A, with the AES/EBU or SPDIF digital interface to the converters used.

5. Analog output signal amplitude is controlled by the analog generator AMPLITUDE field. The DGEN AMPL field on the FASTTEST or FASTTRIG DSP panel has no effect on analog amplitude via the analog generator. For waveforms created with the default one dB headroom value of MAKEWAVE, the peak-to-peak amplitude of the output signal will be one dB less than the generator amplitude shown if Vpp units are used. If other (RMS) units are used, the output peak-to-peak value will be one dB less than the p-p amplitude of a sinewave with the indicated amplitude.

6. To measure analog input devices, the DSP panel is normally set for ANLR-A at CH-1 and ANLR-B at CH-2. Analog signal applied to the analyzer CHANNEL A and CHANNEL B inputs then passes through the balanced inputs, ranging amplifiers and attenuators, and is fed to dual A/D converters. Fixing the analog input range is absolutely required for FASTTRIG to capture a short burst signal, and is strongly recommended for optimum speed with FASTTEST.

7. DSP FREQ is selected at SOURCE-1 for all tests, usually with the LOG selection. Frequency response, distortion, noise, and stereo separation tests have DSP AMP1 as DATA-1 and DSP AMP2 as DATA-2; monaural device tests use only DATA-1. Stereo interchannel phase tests have DSP PHA2 as DATA-1 and nothing at DATA-2.

C.5.10. Saving and Loading Acquired Waveforms

Signals which have been acquired into DSP memory for measurement may be saved to disk from FASTTEST.DSP or FASTTRIG.DSP by the SAVE WAVEFORM command. After furnishing a file name into which the data will be saved, the arguments "1", "2", or "1,2" must be furnished in response to the next query. These arguments specify whether the contents of analyzer acquisition buffer one, two, or both are to be saved to disk. If frequency error correction was used when the signal was acquired, the corrected result is saved to disk.

Previously-saved waveforms may later be downloaded into the acquisition buffer of FASTTEST or FASTTRIG. FFTs may be performed for further analysis with the <F6> key, exactly as with the other FFT programs. The LOAD WAVEFORM command is used, the desired .WAV file name selected, and the "1" or "2" argument supplied. The argument specifies into which acquisition buffer (channel one or channel two) the waveform will be loaded. If the disk file contains two channels of waveform, the "1,2" or "2,1" arguments can also be supplied. "1,2" loads the first waveform in the disk file into channel one and the second waveform into channel two. "2,1" reverses the positions of the two waveforms, loading the first into channel two and the second into channel one. ***Note that frequency error correction only takes place during initial acquisition and cannot be executed on a waveform loaded from disk.***

C.5.11. Performance Factors

C.5.11.1. Generated Signal Resolution

All frequencies generated must have an integral number of complete cycles in the DSP waveform generation memory buffer. This ensures that there are no discontinuities as the DSP generator continuously "splices" from the end to the beginning of that buffer. This requires that every frequency component be an exact multiple of the basic frequency whose period equals the length of the generated waveform buffer (see Figure C-14). With the 8192 sample generator waveform buffer available in FASTTEST and FASTTRIG and using a 48 kHz sample rate, for example, the waveform length is 8192/48000 = 170.66666 milliseconds. The basic frequency is thus 5.8593772 Hz. MAKEWAVE.EXE assures that all frequencies generated will be exact integer multiples of the basic frequency. This frequency is based on either the default 8192 sample buffer or the shorter buffer specified in the /L option of MAKEWAVE.

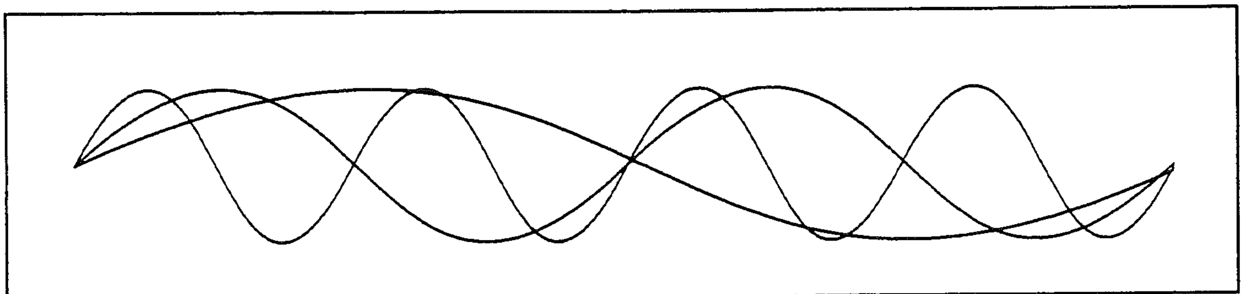


Figure C-14 All Waveforms Must Have an Exact Integral Number of Cycles in the Generator Buffer for Transient-Free Signal Generation. The Signal With a Period Equal to the Buffer Length is the Fundamental Frequency.

C.5.11.2. Complex Signal Amplitude, Crest Factor, and Phase Relationships

As each additional sinewave component is added to a multitone signal, evaluation of the amplitude of the total signal becomes more complex. Depending upon the phase relationships of the individual sinewaves which make up the complex signal, it can have very high peak amplitudes. In the worst case, if all the fundamental components should reach a positive or negative peak simultaneously, the complex signal amplitude would be equal to the sum of the peak amplitudes of all the individual sinewave components. The rms value of the complex signal is the root-mean-square of all the component amplitudes, independent of their phase relationships.

Crest factor is the ratio of the peak to rms values. For frequency response testing of linear devices, it is only important that signal peaks stay below the device clipping level. For distortion testing of devices whose distortion-generating characteristics are a function of power, it is usually desirable for the test signal to have as low a crest factor as practical. This will stress the device with maximum power (rms value) while staying within the device's peak-handling capability (below clipping). Using higher crest factor signals may be desirable in an effort to simulate the speech or music.

C.5.11.3. Waveforms with Lower Crest Factors

One way to reduce the crest factor of the complex waveform is to establish one sinewave component at a much higher amplitude than all the remaining sinewaves. LOWCREST.WAV, furnished with the FASTTEST files, is a 32-tone waveform consisting of the signals in ISO31.WAV plus an additional 1125 Hz sinewave at an amplitude 30.0 dB higher than the other 31 tones. LOWCREST.WAV thus has a crest factor of approximately six dB, only about 3 dB above the crest factor of a single sinewave. The procedure STEREOLC.PRO (where LC stands for "low crest") uses a series of tests to measure response, total distortion and noise, empty-bin noise, and interchannel phase, plus a separate measurement of the 2nd through 9th harmonics of the 1125 Hz tone. Since the complex signal is dominated by the 1125 Hz component, distortion measurements of that tone will correlate more closely to conventional single-sinewave distortion measurements. See the "Furnished Files" section at the end of this chapter for more information on this procedure.

C.5.11.4. Dynamic Range

When testing in the analog domain, it is desirable that the signal peaks closely approach full scale of the D/A and A/D converters. This minimizes the distortion products of the converters (specified to be at least 85 dB below converter full scale). Most of the furnished single-channel waveforms, and waveforms generated with the normal defaults of MAKEWAVE.EXE (described earlier), have their peak amplitudes set one dB below D/A converter full scale. Dynamic range of the D/A is less likely to be a limitation in most applications than dynamic range of the A/D converters.

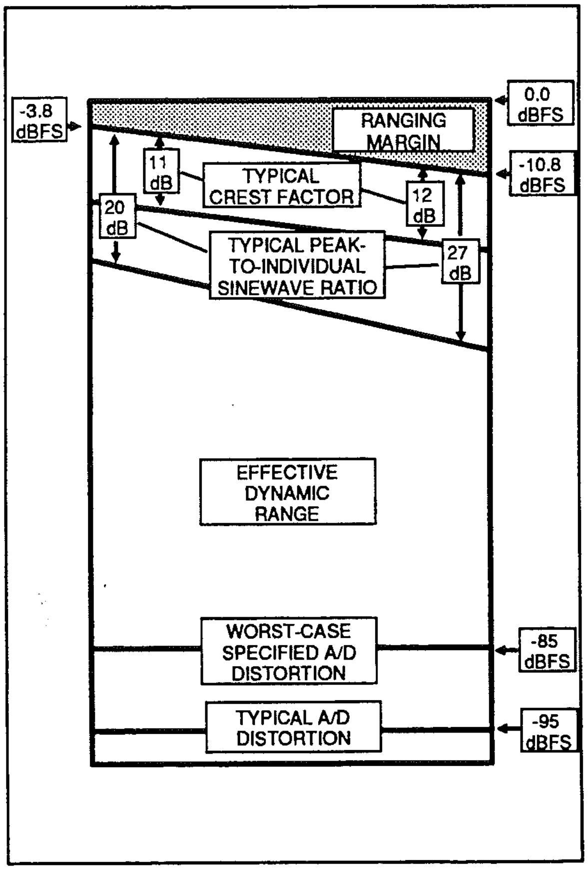


Figure C-15 Measurement Dynamic Range Factors, FASTEST with Furnished Waveforms

When the signal is acquired via the System One analog Channel A and Channel B connectors with input AUTO ranging in use, the peak-sensitive autorange circuits will normally place the signal peaks within the range of about -3.8 dB to -10.8 dB below full scale of the A/D converter. If the signal amplitude falls below the approximate -10.8 dBFS point, the analog analyzer automatically switches in another 6.02 dB gain range (down to the 80 mV most-sensitive input range). If the signal amplitude increases to larger than about -3.8 dBFS, gain will automatically be reduced by 6.02 dB.

The RMS value of a multitone signal will be lower than the peak by the crest factor of the signal. The crest factors of the 15FAST, ISO31, and ISO60 waveforms furnished are in the 11 to 12 dB range. The amplitudes of the individual sinewaves which make up these complex signals are still further below the complex signal peak value. The amount varies among FASTTEST waveforms from about 20 dB for 15FAST.WAV to 27 dB for ISO60.WAV. Thus, distortion products generated in the DSP unit A/D converters (if at the worst-case specified -85 dBFS level) may be 81 dB to 74 dB below the complex signal peak, 70 dB to 62 dB below the complex signal RMS value, and 61 dB to 47 dB below the RMS amplitude of each individual sinewave. See Figure C-15 for a graphic presentation of the dynamic range for complex multitone signals.

C.5.11.5. Measuring Amplitudes

RMS amplitude of the complex signal can be measured in the analog domain with either the LEVEL or READING meter (AMPLITUDE function, RMS detector) of System One's analyzer. Peak amplitude can be measured in the AMPLITUDE function of the READING meter with Peak detector selected. Crest factor may be directly read in the 2-CHANNEL function with the multitone signal connected into both analyzer inputs. When measuring the internal generator this is easily done by using the GEN-MONITOR path at both channels. By selecting the Peak detector for the READING meter and using X/Y (or dB) units, crest factor is displayed since the LEVEL meter always uses RMS detection. The measured crest factor may not be as accurate as the crest factor predicted by MAKEWAVE. This is especially true if the signal has significant energy at very low frequencies, since the Peak detector loses accuracy at these frequencies. The amplitude of any single sinewave must be measured selectively, normally via the FFT analysis capability of FASTTEST-FASTTRIG.DSP.

C.5.11.6. Calibrating Generator Amplitude

The analog GENERATOR AMPLITUDE parameter is designed so that all analog-generated signals have the same peak values. It is calibrated (except for Vpp units) in terms of the RMS value of a sinewave. A DSP-generated waveform with peaks reaching digital full scale and sent through the analog output via the WAVEFORM DGEN selection would thus have equal peak amplitude to a sinewave of the value entered in the AMPLITUDE field (for instance, 2.828 Volts peak-to-peak with a 1.000 Volt RMS GENERATOR AMPLITUDE value). MAKEWAVE normally creates multitone waveforms with the peak instantaneous amplitude one dB below full-scale digital output. This default headroom allowance will cause the peak amplitude of the complex signal to be one dB (about 12.2%) further below that calibration value.

Establishing the absolute analog output amplitude for any individual sinewave component of any multitone waveform requires a selective measurement. It cannot be deduced from the AMPLITUDE field value or from RMS or Peak amplitude measurements. Selecting the analyzer GEN MONITOR path (with no external loading on the generator outputs) and acquiring and performing an FFT with any absolute units will provide the amplitude of each of the sinewaves in the output signal to approximately ±0.25 dB accuracy. The ratio of the GENERATOR AMPLITUDE field value to the selectively-measured RMS amplitude of any sinewave will remain constant for any .WAV file, but varies from waveform to waveform. For the furnished equal-amplitude multitone .WAV files with FASTTEST, the calibration factors are shown below. The individual sinewave amplitudes for all the .WAV files furnished for FASTTRIG are shown in the tables at the end of this chapter.

	GEN. PANEL AMPLITUDE	EACH SINEWAVE AMPLITUDE
15FAST	0.0 dBV	-20.8 dBV
ISO31	0.0 dBV	-25.2 dBV
ISO60	0.0 dBV	-27.8 dBV

C.5.11.7. Hardware Modification for Improved Low-Frequency Amplitude Accuracy

Units built before serial number SYS1-03329 have a low-frequency roll-off in the coupling circuitry between the DSP D/A output and the analog generator input. This causes an error reaching about -0.6 dB at 20 Hz, with significant phase shift beginning as high as 100 Hz. A one-resistor change on the DSP module corrects the amplitude roll-off (but does nothing for the phase shift) on units below S/N SYS1-03329. This modification may be made at Audio Precision or any Audio Precision Distributor.

C.6. Features Unique to FASTTEST

C.6.1. FASTTEST.DSP Panel

Figure C-16 shows the DSP panel and Figure C-17 shows the HELP DSP screen when FASTTEST.DSP has been loaded.

The TRANSFORM line has a field with the selections MAXIMUM, 4096, 2048, 1024, 512, and 256. This field functions identically to FFTGEN.DSP. It determines the record length which will be acquired into DSP acquisition buffers when <F9> is pressed, and the length of record (starting at the trigger point at time zero) which will be transformed upon <F9> or <F6>. For an SYS-222 without the MEM option, MAXIMUM and 4096 produce the same 4096-sample acquisition buffer and transform length. With full memory units (SYS-222 with MEM or SYS-322), MAXIMUM produces a 16,384-sample acquisition and transform. For NOISE mode to function properly, the following relationship to the generated signal repetition length must be established by the user. See the description of the "/L" option of MAKEWAVE.EXE later in this chapter for how to control the repetition length of the generated signal.

Generator Buffer (/L option of MAKEWAVE.EXE)	Analyzer Buffer
8192	16384
2048	4096
1024	2048
512	1024
256	512

On the TRANSFORM line, the right hand field allows selection of the window function applied before the FFT. The four choices are NONE, HANN, FLAT, and BH4. Note that these choices are not arranged in the same sequence in FASTTEST as in FFTSLIDE and FFTGEN. NONE is normally the selection with FASTTEST, since the

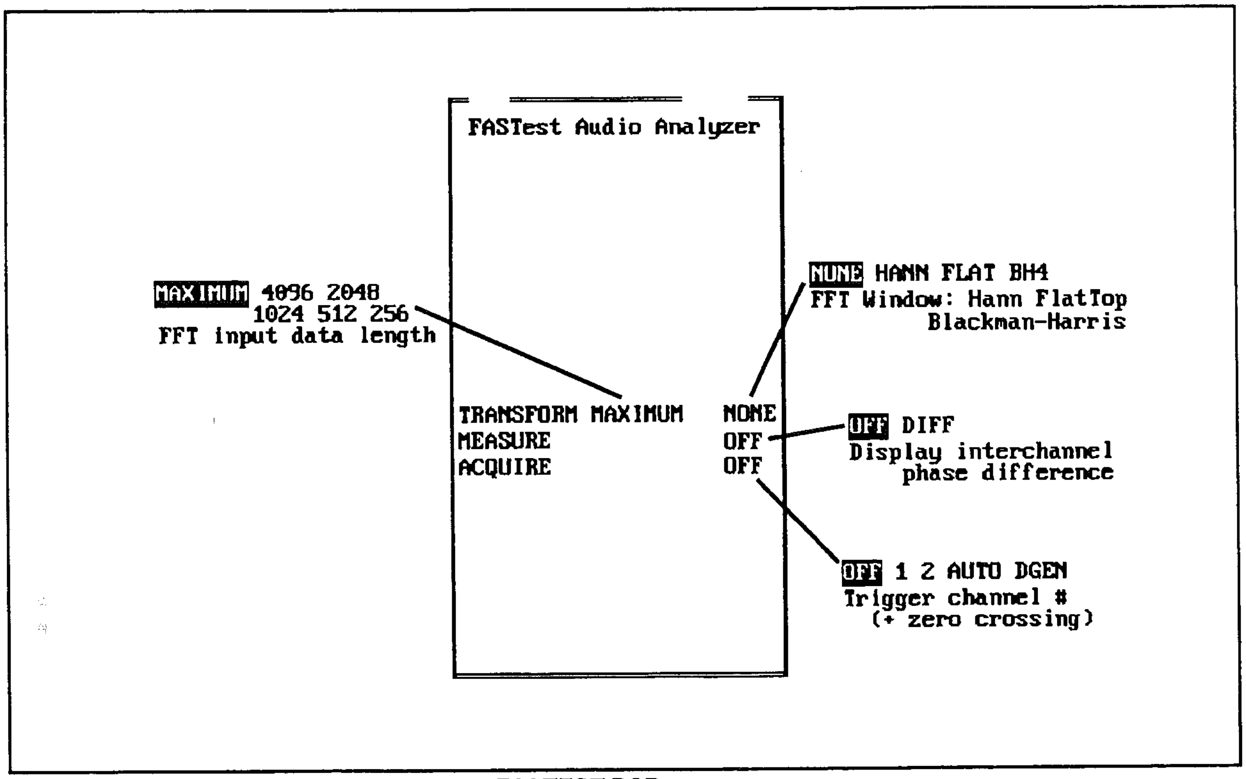


Figure C-16 DSP Panel Features Unique to FASTEST.DSP

DSP HELP SCREEN

Multiple sinewave test signals are generated and analyzed for high speed audio system testing. These tests include response, distortion, noise, phase response, and interchannel phase.

The panel readings are real time peak amplitude displays to assist in setting levels.

The F9 function key causes a new data acquisition and measurement. The F6 key causes previously acquired data to be transformed.

The minimum frequency resolution of the generated signal at 48 kHz sample rate is 5.86 Hz with full memory, 23.4 Hz with standard memory.

Window choices are HANN, FLATtop and BH4. NONE gives no windowing. NONE is normally used since the generated and acquired signals are synchronous unless the device under test shifts frequencies or time compresses.

CAUTION: DSP Panel readings must use %FS or dBFS. F7 does not allow units to be changed, use F6. DATA1 and DATA2 must not be the same (AMPL-1 or AMPL-2).

Figure C-17 DSP HELP Panel with FASTEST.DSP Loaded signal is generated from a buffer exactly equal in length to or an exact integer sub-multiple of the analyzer acquisition buffer. Thus, no window is required. For unique applications where the generated signal repetition cycle length is not synchronous with the acquisition buffer length, one of the windows may be selected. For highly accurate amplitude measurements, the FLAT (Flattop) window should be chosen. BH4 and HANN provide better selectivity than FLAT, but introduce significant amplitude error if the actual signal frequency is not exactly at the center of an FFT bin. See the "Window Functions and Selectivity" section of the FFT PROGRAMS chapter of this manual for more information on these windows.

The right-hand field on the MEASURE line offers the choices of DIFF or OFF. When DIFF is selected, the PHAS-2 variable can be selected at either DATA-1 or DATA-2 and the resulting curve will be phase difference between the two DSP channels. If OFF is selected, PHAS-2 is the absolute phase of the DSP channel two signal.

C.6.2. Triggering Signal Acquisitions, FASTTEST

FASTTEST has limited triggering capability. It is designed principally for measurement of devices with little or no time delay between their input and output, and the input and output both within practical cable length of the System One used. Signal is normally generated continuously in FASTTEST, applied to the device input, and an acquisition and measurement made immediately at the device output under the assumption that the continuous multitone signal is present there. No checking is done by FASTTEST to see that the acquired signal is correlated to the signal being generated.

The right-hand field on the ACQUIRE line controls acquisition triggering with choices of OFF, 1, 2, AUTO, and DGEN. When OFF is selected, the DSP immediately acquires signal whenever <F9> is pressed. With the 1 selection, acquisition will be triggered when the DSP channel one signal first exceeds -60 dBFS (0.1%FS) after the <F9> key is pressed. The 2 selection is identical except driven by the DSP channel two signal. The AUTO selection causes acquisition upon any signal activity, even at the LSB level, on either DSP channel one or channel two after <F9> is pressed. The DGEN selection triggers an acquisition at the first "time zero" sample (waveform beginning) of the generator buffer after <F9> is pressed. This trigger selection is intended for measuring phase shift (time delay) through a device with FASTTEST.

C.6.3. Loading Waveforms and Running Tests, FASTTEST

The desired generator waveform must be downloaded to the DSP module before testing can begin. The waveform will remain in the DSP module while different .TST files all using the same DSP program are consecutively loaded. ***Loading a test with a different .DSP program attached, including going from FASTTEST to FASTTRIG or vice-versa, destroys any previously-loaded waveform in the DSP.***

Generator waveforms for FASTTEST with full memory must be full 8192-sample waveforms. Without full memory, FASTTEST waveforms must be 2048 samples. See the descriptions of the "/L" and "/S" options of MAKEWAVE.EXE later in this chapter for how to generate a 2048-sample waveform. To load a waveform, select the LOAD WAVEFORM command, use the cursor to select the desired .WAV filename and press <Enter>. Then, specify which generator channel the waveform loads into by supplying the correct argument in response to the prompt which follows. When a generator waveform is downloaded to FASTTEST with the "1G" argument supplied, the waveform automatically loads into both the channel one and channel two generator buffers. A second download with the "2G" argument is not required unless it is desired to have separate waveforms in the two channels. For separate waveforms, the channel two waveform must be downloaded last with the "2G" argument and will over-write any waveform already present in that channel. Alternately, if a dual waveform .WAV file with different waveforms in channel one and two has been created by the technique described later, it will load into both buffers in a single step by a LOAD WAVEFORM command and the "1G,2G" argument. Note that, if different waveforms are loaded into the two generator buffers, they can only be sent out via the digital interface of a System One Dual Domain.

The only checking of downloaded generator waveforms done by FASTTEST.DSP is to verify that they are of the correct length to fill the generator buffer (8k with full memory, 2k for a System One + DSP without MEM option). No testing is done for the number of tones or for tone spacing.

For optimum speed, fixing the input ranges is recommended. If AUTO ranging is selected, the analyzer must search for the optimum range before signal is acquired and processed. If the signal amplitude is known, the input ranges can be fixed on the appropriate range and faster operation will result.

C.6.4. Furnished Files, FASTTEST

Many waveform, sweep, and other files are furnished for use with FASTTEST. These files are discussed at the end of this chapter.

C.7. Features Unique to FASTTRIG

C.7.1. FASTTRIG.DSP Panel

Figure C-18 shows the DSP panel and Figure C-19 shows the HELP DSP screen when FASTTRIG.DSP has been loaded.

The left field on the TRANSFORM line gives three choices of triggering criteria, NORMAL, LOOSE, and TIGHT. These choices give different trade-offs between the amount of signal quality degradation acceptable through which FASTTRIG can still recognize the multitione signal, versus the probability of falsely triggering on music or other normal program material.

On the TRANSFORM line, the right hand field allows selection of the window function applied before the FFT. The two choices are NONE and HANN. NONE is normally the selection with FASTTRIG, since the signal is generated from a buffer exactly equal in length to or an exact integer sub-multiple of the analyzer acquisition buffer. Thus, no window is required. For unique applications where the generated signal repetition cycle length is not synchronous with the acquisition buffer length, the HANN (raised cosine) window may be selected.

Proper operation of both the triggering-upon-recognition function (TRIG+n) and the frequency error correction function require the multitone signal to have certain characteristics. There must be a sufficient number of tones in the middle and upper frequency band (typically 300 Hz to 10 kHz). The tones in this frequency range must comply with a minimum frequency spacing criterion of six bins (about 35 Hz with a maximum resolution signal, 140 Hz with a 2048-sample signal for 250 millisecond burst testing). FASTTRIG.DSP checks downloaded generator waveforms to verify that they meet these criteria. Warning messages will be displayed if there are too few sinewaves or if the spacing between any pair of sinewaves in the middle and upper frequency band is too close. A waveform may still be useful for certain applications even if it does not comply with these criteria. Since the warnings have the effect of halting an automatic procedure, a field is provided at the right end of the MEASURE line of the FASTTRIG panel to turn off the warnings. ON leaves the warnings intact; OFF disables the warnings so that procedures can run even if there are deficiencies in the waveform.

C.7.2. Generator Waveforms

Generator waveforms for FASTTRIG may be full 8192-sample waveforms for maximum frequency resolution, or may be shorter by binary multiples (4096, 2048, 1024, 512, or 256) if greater speed is required and poorer frequency resolution is acceptable. FASTTRIG requires waveform files to be downloaded into both the channel one and channel two generator buffers. To load the same waveform into both channels, select the LOAD WAVEFORM command, use the cursor to select the desired .WAV filename and press <Enter>, and then supply

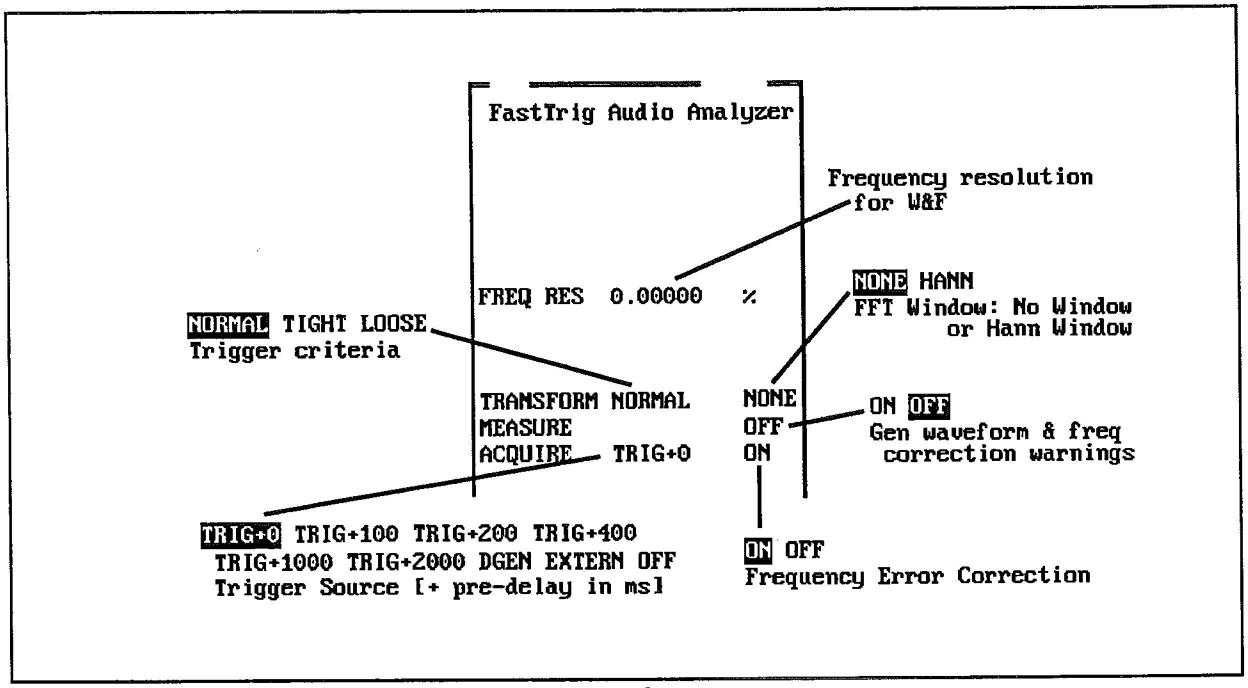


Figure C-18 DSP Panel Features Unique to FASTTRIG.DSP

DSP HELP SCREEN

Multiple sinewave test signals are generated and analyzed for high speed audio system testing. These tests include frequency response, distortion, noise, interchannel phase, and stereo separation.

The top two panel display fields are real time peak amplitude displays to assist in setting levels.

FASTTRIG can recognize and capture short bursts of signal by comparing incoming test signals to downloaded generator waveforms. The expected generator waveform must be downloaded to both 1G and 2G buffers in order to capture signals. 1G must be loaded before 2G, and both signals must be the same length. The analyzer acquisition length will be twice the generator buffer length. The F9 function key causes FASTTRIG to begin comparing input signals. Once acquired and transformed, data can be re-analyzed for different parameters with the Alt-F6 key.

External dual D/A converters driven from a digital output of System One Dual Domain are necessary for stereo separation testing. A and B analog outputs both come from the 2G buffer signal.

Window NONE is normally used for acquiring signals originally generated by FASTTRIG. The Frequency Error Correction function accepts signals with up to 3% frequency error.

Figure C-19 DSP HELP Panel with FASTTRIG.DSP Loaded the "1G" argument. Select LOAD WAVEFORM again (the cursor will still have the same waveform selected), press <Enter> again, and supply the "2G" argument. If separate waveform files are to be loaded into the two channels, the "1G" argument must be furnished for the first and the "2G" argument for the second. The channel one waveform must be loaded before the channel two waveform. Both waveforms must be the same length. If a dual-channel .WAV file for stereo separation testing has been created by the technique described later, it will load into both buffers in a single step by a LOAD WAVEFORM command and the "1G,2G" argument. Furnished dual-channel waveforms are distinguished by the character "S" as the fifth character of the file name.

C.7.3. Acquisition Buffers

The TRANSFORM length field of FASTTEST, FFTGEN, and FFTSLIDE is not present on FASTTRIG. When a generator waveform is downloaded to FASTTRIG, the program determines the length of the waveform and automatically sets the analyzer acquisition buffer length to exactly twice that value.

FASTTRIG cannot acquire signals into the two channels at different times by setting the CH1 or CH2 input selection to NONE; both channels must be acquired simultaneously.

C.7.4. FASTTRIG Triggering

FASTTRIG has very powerful and flexible signal recognition and triggering capability. This allows it to capture brief bursts of a specific multitone signal inserted into a broadcast program, or at the beginning or end of a tape recording.

The signal recognition capability is based upon comparison of the incoming signal to the waveforms stored in the generator buffer. Therefore, FASTTRIG must have the expected multitone signals downloaded into both channels of the generator buffer even if the generator is not being used. This will be the case when signal is expected from a recording or a distant System One. FASTTRIG has two generator channels and will potentially be examining two input channels when stereo channels or devices are being tested. FASTTRIG tolerates stereo channel transposition or one missing channel; it compares each generator buffer signal with each channel of the received signal and will trigger if any of the four combinations match.

The signal to be recognized and acquired may have been seriously degraded by non-flat frequency response through the device or system being measured, and by the addition of noise, distortion, and interfering signals. For this reason, the recognition technique has built into it certain tolerances for differences in frequency response and addition of noise, distortion, and interference in the acquired signal with respect to the "pure" reference signal in the generator buffers. Three choices are available on the TRANSFORM line for these tolerances; NORMAL, LOOSE, and TIGHT. LOOSE tolerances may be required in order to trigger on signals which have passed through channels with large amounts of response variation and noise or distortion. Broadcast signals which have passed through modulation processors will frequently require the LOOSE selection.

As the tolerances are made wider, the probability increases that FASTTRIG may at some time falsely identify music or other program material as matching the generator waveform. Use of the shorter waveforms (required for shorter burst lengths) also increases the probability of falsing, since the wider FFT bins produced from shorter records make it more difficult to discriminate between normal program material and the multitone signal. Adequate protection against falsing with very short records may require TIGHT triggering, which may then not recognize signals passed through systems with substantial amounts of distortion and response variation.

When one of the triggered modes is selected, FASTTRIG continuously acquires and compares short segments of the incoming signals to the stored waveforms. When the incoming signal matches the stored waveforms, FASTTRIG.DSP makes a second acquisition. The data is frequency corrected if necessary, Fourier transformed, and control is passed back to the S1.EXE test or procedure which initiated it. The test or procedure may then make one or more measurements from the transform results.

Several choices of time delay may be inserted between the recognition and the start of the second acquisition to permit settling of companders in the audio system being tested. FASTTRIG may also be commanded to trigger by an externally-applied logic signal. Frequency correction may be required for cases where the signal has been recorded on analog tape and played back at a slightly different speed. It is also useful in cases of digital generation from another System One or playback of a digital recording from a DAT or CD player whose quartz clock frequency is slightly different from the clock in the System One acquiring the signal.

All .TST files for burst signal acquisition with FASTTRIG must have fixed input ranges selected. The time required for autoranging will prevent capture of the short signal bursts which FASTTRIG is designed to capture. The CHANNEL A RANGE and CHANNEL B RANGE fields both must be fixed at the range appropriate for the amplitude of signal to be captured. Thus, some fore-knowledge of signal amplitude must be available. If the selected fixed range is too low, higher amplitude signals will clip in the analyzer input stages and severe distortion will be produced. If the selected fixed range is too high, the full dynamic range of the A/D converters is not used. The internal noise and distortion of the converters and analyzer analog input stages may then become a limiting factor.

The furnished .WAV files have all been constructed for compatibility with the triggering and frequency error correction capabilities of FASTTRIG. If custom waveform files are to be constructed by the user, they must satisfy the criteria described below in the "Special FASTTRIG Signal Considerations" section.

In determining whether to trigger, FASTTRIG will determine amplitude in a band of frequencies around each reference fundamental frequency. This allows for devices or situations in which frequencies have been shifted. The percentage frequency range which will be examined around each fundamental frequency is controlled by the value entered into the FREQ RES field on the FASTTRIG.DSP panel. Even when zero is entered, a small range is searched consistent with the typical error between the crystal time bases of two System Ones, or between a System One and a digital reproducer such as an RDAT machine or CD player. To allow for larger values of frequency shift, enter a value up to 3.00% into the FREQ RES field.

The first field on the ACQUIRE line selects the triggering method and source and provides several choices of delay after signal recognition and before the final acquisition, transform, and measurement. The selections are TRIG+0, TRIG+100, TRIG+200, TRIG+400, TRIG+1000, TRIG+2000, DGEN, EXTERN, and OFF. The six TRIG+n selections all enable the recognition and triggering mode, in which (following an <F9> operation) the incoming signal is continuously compared to the stored waveforms. Only when a match occurs does a final acquisition occur and the DSP signals to S1.EXE software that data is ready to be transmitted to the computer. The TRIG+100, TRIG+200, TRIG+400, TRIG+1000, and TRIG+2000 selections each insert a time delay of the stated value (in milliseconds) following recognition before starting the acquisition. These choices permit allowance for compander settling before the final signal is acquired.

The DGEN selection is identical to FASTTEST as described earlier, triggering at the first passage through sample zero of the generator buffer after <F9> is pressed. The EXTERN selection gives control to pin 3 of the SERIAL connector (15 pin D-SUB) on the rear panel. This line is normally held high by an internal pull-up resistor. When it is pulled low by an external trigger signal (after <F9> has been pressed), an acquisition is immediately triggered. The OFF selection causes an immediate acquisition when <F9> is pressed. Thus, the DGEN, EXTERN, and OFF selections do not compare the acquired signal to waveforms stored in the generator buffers. However, FASTTRIG demands that waveforms be downloaded to the generator buffers regardless of the triggering selection.

The right-hand field on the ACQUIRE line controls the frequency error correction function. Correction will take place when ON is the selection. The amount of correction necessary is determined by comparison of the acquired signal to the stored waveform. Frequency error correction permits accurate measurements and low noise floors even when the acquired signal frequencies have been shifted as much as three percent. Shifts of more than three percent, poor signal-to-noise ratio, or additional distortion products added by the device under test may cause frequency error correction to fail. In this case, a warning message will occur. All warning messages relating to FASTTRIG generator waveforms and frequency error correction may be disabled by selecting "OFF" in the right-hand field on the MEASURE line. When automatic procedures are to run at unstaffed locations, warnings should be disabled since they would otherwise cause the procedure to pause indefinitely.

Frequency error correction is frequently necessary when testing analog tape recordings made on one machine and reproduced on another. It is also necessary when digitally-recorded signals are played back as analog signals from a machine whose sample rate is different from the receiving machine. It is similarly required with analog transmission between two System Ones whose sample rates are not identical. Frequency correction takes approximately two to three seconds with a full 16,384-sample acquisition. The time required for correction decreases rapidly with shorter acquisitions.

C.7.5. FASTTRIG Burst Length

The minimum length burst which FASTTRIG can be guaranteed to trigger on depends upon several things. The major factors are the generator buffer length of the signal and the value of any delay selected in the field after ACQUIRE on the DSP panel. Minor variations in required burst length are caused by whether or not a digital input or output is selected and by whether the Frequency Error Correction feature is turned on or off. The table below shows worst-case burst lengths for all generator record lengths and the digital input/output and Frequency Error correction conditions, assuming a zero time delay (ACQUIRE TRIG+0). If any time delay is used to allow for compander settling, the minimum required burst length increases by the delay used. For example, the 260 ms burst length required with a 2k record length, analog input-output, and Frequency Error Correction, would become 460 ms if ACQUIRE TRIG+200 is selected. All the values in the table below assume the 48 kHz rate. They would become proportionately longer at the 44.1 kHz or 32 kHz sample rates. In the typical case of acquisition of a distant signal, when the DSP generator not being used at all, the most rapid operation will result from selecting PARALLEL as the OUTPUT choice near the bottom of the DSP panel.

Generator Record Length of Signal	Analog input-output, Freq. Error Correction	Analog input-output, no Freq. Error Correction	AES/EBU input-output, Freq. Error Correction	AES/EBU input-output, no Freq. Error Correction
8k	1050 ms	1000 ms	1110 ms	1060 ms
4k	520 ms	500 ms	550 ms	530 ms
2k	260 ms	250 ms	280 ms	270 ms
1k	135 ms	125 ms	145 ms	140 ms
512	70 ms	65 ms	77 ms	75 ms
256	40 ms	36 ms	44 ms	41 ms

S1.EXE hardware and software does not include a precise method for generating short bursts. Several procedures are furnished with FASTTRIG (BURDIG-M,PRO, BURDIG-S.PRO, BURANA-M.PRO) whch generate approximately-timed bursts in both digital and analog domains, using the UTIL DELAY command. See the "Burst Generation Procedures" section near the end of this chapter for more details. Particularly when the burst signal is to be digitally recorded for convenient and repeatable access during broadcast programming, the burst may be best generated by use of a digital editor. Personal computer-based digital editors which will accomplish this task are available at modest prices.

C.7.6. Special FASTTRIG Signal Considerations

For a signal to be compatible with the "recognize-and-trigger" function (TRIG+n selections on the ACQUIRE line) and the frequency error correction function of FASTTRIG, certain additional criteria must be satisfied within a "searched" frequency range bounded by a low frequency value F_{low} and a high frequency F_{high}:

- A minimum of five sinewaves must exist within this range
- No sinewaves within this range should be spaced more closely than six times the generator frequency resolution value. This value is approximately 5.9 Hz for 8192-sample lengths, 11.8 Hz for 4096 samples, 23.75 Hz for 2048 samples, 46.9 Hz for 1024 samples, 93.75 Hz for 512 samples, and 187.5 Hz for 256 samples.

The definition of F_{low} is the higher of:

300 Hz plus one generator bin

or

5 times the generator frequency resolution value.

The definition of F_{high} is 0.208333 times the sampling rate. Thus, F_{high} is 10.0 kHz at the commonly-used 48 kHz sample rate, 9.187 kHz at the 44.1 kHz rate, and 6.667 kHz at the 32 kHz rate.

Only the range between F_{low} and F_{high} will be searched for compliance with the minimum spacing and minimum number of tones criteria. Outside this range, the signal may contain any number of tones at any spacing available with the generator sample length used (see /L option below). The device under test may attenuate or reject frequencies outside this range with no effect on FASTTRIG's ability to recognize the signal, trigger, and make accurate frequency error corrections.

Generated Waveform Length (/L)	F_{low}	Minimum Sinewave Spacing in the Range Between F_{low} and F_{high}
8k (8192)	306 Hz	35 Hz
4k (4096)	312 Hz	70 Hz
2k (2048)	323 Hz	141 Hz
1k (1024)	347 Hz	281 Hz
512	562Hz	562 Hz
256	1125 Hz	1125 Hz

C.7.7. Loading Waveforms and Running Tests, FASTTRIG

The desired generator waveform must be downloaded to the DSP module before testing can begin. The waveform will remain in the DSP module while different .TST files all using the same DSP program are consecutively loaded. ***Loading a test with a different DSP program attached, including going from FASTTEST to FASTTRIG or vice-versa, destroys any previously-loaded waveform in the DSP.***

To load the same waveform into both generator channel buffers, use the LOAD WAVEFORM command. Select the name of the .WAV file to be downloaded, and press <Enter>. Supply "1G" and <Enter> in response to the next prompt from S1.EXE software. This causes the binary data of the waveform file to be loaded into the channel one buffer only. Select LOAD WAVEFORM again, press <Enter> to select the same waveform file, and supply the "2G" argument to load the channel two buffer.

If different waveforms are needed in the two buffers, load the channel one waveform first using the "1G" argument, followed by the channel two waveform with the "2G" argument. An error message will be obtained if an attempt is made to use FASTTRIG.DSP with only one generator buffer loaded, or if waveforms of different lengths are loaded into the two buffers. If a stereo .WAV file has been created with different waveforms in the channel one and channel two segments of the file, it can be downloaded in one operation with the "1G,2G" argument after selecting the .WAV file name. Note that when different waveforms are loaded into the two generator buffers, they can only be sent out via the digital interface of a System One Dual Domain since only one D/A converter and only one analog generator output stage is present.

For many applications it is most convenient to use procedures when testing with the multitone technique. The procedure would first load the test intended for signal capture (with .DSP program attached) and then load the waveform(s). The output signal from the device under test would then be acquired and the Fast Fourier Transform performed by the <F9> keystroke. It is often desirable to measure several different characteristics (for example frequency response, total distortion, phase, and noise) and display each separately. To do this, the procedure would load a sequence of test files. Each would have the appropriate settings on the panels and the appropriate .SWP file attached. For each different measurement, the FFT data (still in DSP memory) would be re-processed and the results transmitted from DSP module to computer by execution of the <Alt><F6> keystroke.

C.8. Creating Custom Waveform and Sweep Files

A utility program named MAKEWAVE.EXE creates files to generate the multitone signals and .SWP tables used by FASTTEST and FASTTRIG. Even though many waveform files are furnished with these two programs, the user may desire to create his own. One reason may be to make a waveform whose spectral energy distribution matches that of some specific program material; see the Program Spectral Distribution section below for more information.

MAKEWAVE.EXE operates with a standard System One data file (.DAT) or the equivalent created with a word processor or text editor as its input. MAKEWAVE.EXE creates two output files; the downloadable waveform file (.WAV) and an ASCII file list of exact fundamental frequencies (.DAF file). See Figure C-20 for a simplified schematic diagram of the waveform creation, loading, and output process. A second furnished utility program, MAKEDIST.EXE, is used only when it is desired to measure harmonic and intermodulation distortion separately, or to make a single combined evaluation of response, harmonic, and intermodulation distortion.

C.8.1. Creating a .DAT File

The starting point to create a custom multitone waveform is entering the desired frequency, amplitude, and (optionally) phase information into a standard System One data (.DAT) file in the EDIT DATA mode of S1.EXE software. Alternately, any ASCII text editor could be used if the .DAT file format including the header line is duplicated.

To generate the waveform file specifications with S1.EXE in EDIT DATA mode, first use PANEL mode to set SOURCE-1 to a frequency parameter such as GEN FREQ. Set DATA-1 to ANLR and an amplitude parameter, such as DATA-1 ANLR LEVEL. Select any decibel unit (dBV, dBu, dBr, etc.) at DATA-1 if you prefer to specify amplitudes in dB. The specific dB unit selected is not important, since it is used only to specify the relative amplitude of the sinewaves to be generated. The absolute amplitude will be determined when MAKEWAVE.EXE is run. Select Volts if you wish to work in linear units. If you wish to also specify the starting phase of each sinewave to be generated, select DATA-2 ANLR PHASE with the degree unit desired. If you are willing to have all signal components start at zero degrees phase, you may leave DATA-2 NONE.

When you enter EDIT DATA mode, you should see column headers similar to those shown below, depending upon your choice of units:

Hz dBV deg (or OFF)

Move the cursor to the first row below the header. Type in one row for each frequency desired, with the amplitude and (optionally) phase values desired. Use a comma to separate the values on each line and the <Enter> key at the end of each line. Integer values must be used, with no imbedded commas within any number; EDIT DATA mode will not accept abbreviations such as k

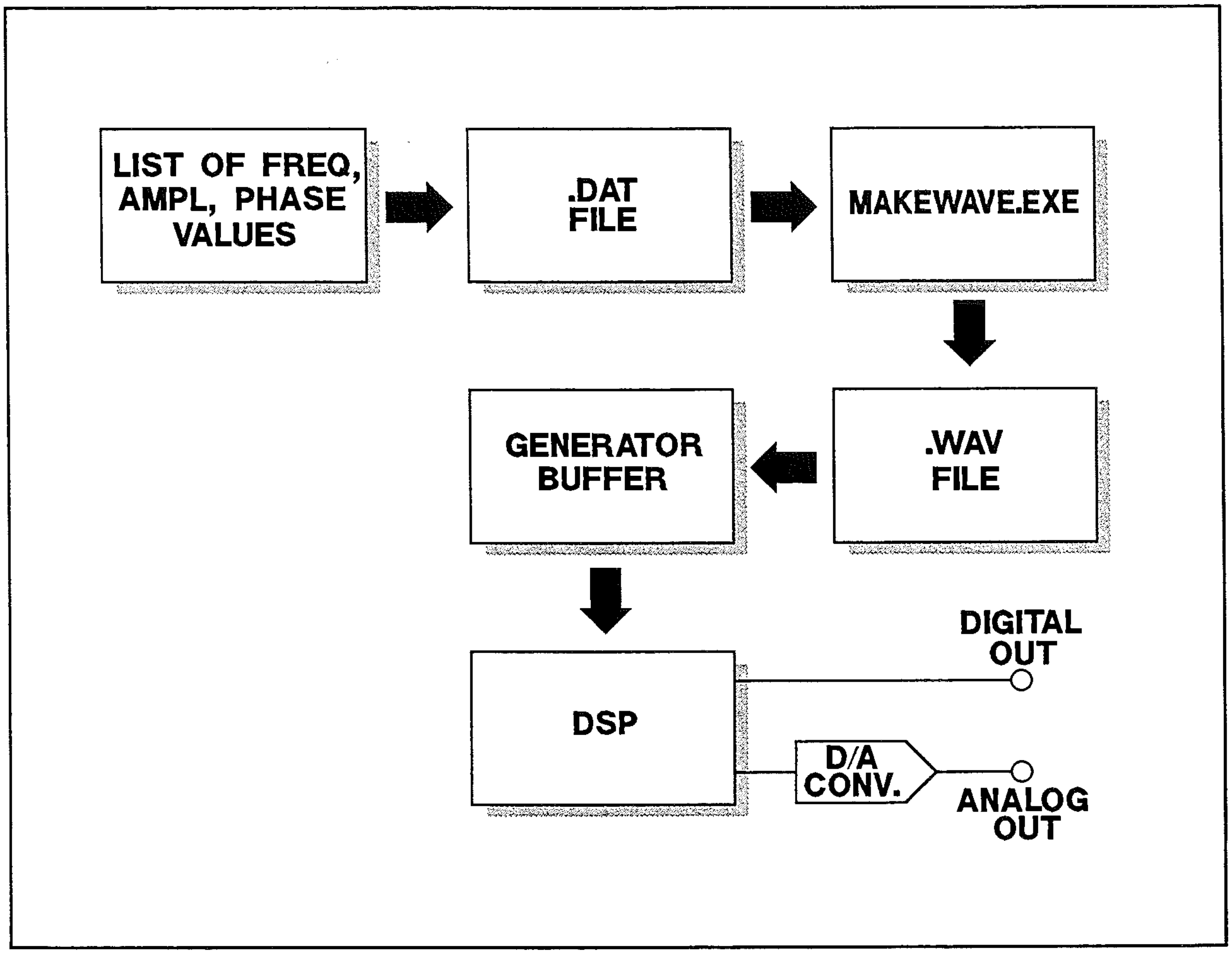


Figure C-20 Simplified Flow Diagram of Creation of Custom Waveform Files

for kilo. Use a minus sign for negative decibel values or phase values, but do not use a plus sign for positive values. When complete, save the data file to disk with a descriptive name. The five-frequency file illustrated below was saved as FIVE.DAT.

Hz	dBV	OFF
70,	0,	0
270,	0,	0
1020,	0,	0
3920,	0,	0
15000,	0,	0

At this point, QUIT from S1.EXE and use MAKEWAVE.EXE as described below, directly from the DOS prompt. The 64k of memory normally set aside by S1.EXE for programs to run under the XDOS or DOS commands is not sufficient for MAKEWAVE.EXE to run. The error message "Memory allocation failed; not enough memory available" will result. If desired, S1.EXE can be started with the /R or /B command line options to leave enough memory for MAKEWAVE to run. See the "Controlling Memory Usage" section of the "Creating Your Custom Software Start-Up Process" chapter in the main System One User's Manual for full details on these options.

C.8.2. Using MAKEWAVE.EXE

MAKEWAVE.EXE operates with a data (.DAT) file as input and creates two output files. One is a binary waveform file (.WAV) which will be downloaded to FASTTEST or FASTTRIG for actual generation of the multitone signal. The second is an ASCII .DAF file (meaning data fundamentals) containing a list of the exact fundamental frequencies which will be generated. This is an intermediate step toward production of the sweep table (.SWP) which will be used during data analysis. MAKEWAVE.EXE is run simply by typing at the DOS prompt its name, followed by the .DAT file name to be converted. For example:

MAKEWAVE FIVE <Enter>

for the case of FIVE.DAT created above. It is not necessary to supply the .DAT file extension for the input file nor file names for the output files, if you are willing for the output files to carry the same names as the input .DAT file except for the file extension. In the example above, MAKEWAVE will automatically name the two output files FIVE.WAV and FIVE.DAF. The .DAF file for the above example is reproduced below:

Hz	dBV	OFF
70.3125,	0,	0
269.53125,	0,	0
1019.53125,	0,	0
3919.92187,	0,	0
15000,	0,	0

Note that the requested frequencies have been rounded off by MAKEWAVE.EXE to the nearest "legal" frequency (integer multiple of 5.8593772 Hz for the default 48 kHz sample rate and 8,192 sample record). Every sinewave must go through an exact integral number of cycles in the generator buffer so that no discontinuity is created when the generator "splices" from the end to the beginning of the record; see Figure C-14. S1.EXE can then be restarted and FIVE.DAF loaded with the LOAD DATA command. Entering "*.DAF" after selecting LOAD DATA will display a list of all the .DAF files in the current directory and FIVE.DAF can be selected with the cursor. The same units used when creating the original .DAT file should be chosen at DATA-1, DATA-2, and SOURCE-1 to avoid an error message when the .DAF file is loaded. The SAVE SWEEP command can now be used and a file name furnished to make a sweep table. This table would be used with FASTTEST or FASTTRIG in NORMAL or RESPwW+F mode for frequency response measurements, in DISTORT mode for total distortion and noise measurements, and in NOISE mode for noise (empty bin) measurements.

If you wish the output files to have different names, you may supply those on the command line when running MAKEWAVE.EXE. However, file management is normally simplified by using the same file name for the original .DAT file, the waveform (.WAV), fundamental frequency list (.DAF), and the resulting sweep (.SWP) files containing fundamental frequencies.

MAKEWAVE.EXE makes a large number of complex computations in generating the .WAV file. The speed with which it runs depends strongly upon the type of computer used and presence or absence of a math co-processor in the computer. It also depends upon the number of signal frequencies specified, plus several of the options described below. Execution time of MAKEWAVE.EXE with a 386-based computer with co-processor can range from a few seconds for .DAT files containing only a few frequencies to a few minutes for complex files. With slower computers such as 8088-based or slow 80286-based machines and with no math co-processor, tens of minutes or longer will be required for complex signals. Operation speed can be improved by use of some of the command line options.

C.8.2.1. Program Spectral Distribution

To test audio devices and systems under conditions duplicating actual usage as nearly as possible, it may be desirable to create an amplitude-shaped multitone waveform whose spectral energy distribution is typical of program material. Here is how this can be accomplished.

Use FFTGEN with its transform averaging feature. Connect a source of typical program material to the inputs of System One. Select a spectral averaging value which will average across the desired length of time. With the MAXIMUM resolution selected, approximately one sample will be acquired, transformed, and added to the DSP register per average. Thus, 16x averaging will average across about 16 seconds of program material, 64x across 64 seconds, etc. Use dBV as the DATA-1 display units, with typically a 30-50 dB range between GRAPH TOP and BOTTOM. Set SOURCE-1 as DSP FREQ with a LOG display from 20 Hz to 20 kHz. Use a moderately-high # STEPS such as 250 to 500. After the final acquisition and average, the screen will display the spectral energy distribution of the program material. To reduce excessive detail, use COMPUTE SMOOTH with an argument of typically 50 to 200 passes of smoothing. Save the result as a .TST file with an appropriate name.

Then, use the COMPUTE NORMALIZE menu command with the arguments "1,1k,0". This will force the displayed amplitude at 1 kHz to exactly zero dBV (one Volt). Use the COMPUTE INVERT command to invert (reciprocate) this data. Re-save the .TST; use a new name if you wish to also retain the original measured data. This last (inverted) file will be used as the "delta" file with the COMPUTE DELTA command.

Load an equal-amplitude .DAT file which has the desired frequencies. Use NAMES DELTA and select the file name of the delta file created as described above. Use the COMPUTE DELTA command; press the <Enter> key again in response to the request for arguments (same as 1,1). The sinewave amplitudes will now be amplitude-shaped exactly like the program material averaged. The SAVE DATA command can be used and MAKEWAVE run as described above to create a waveform file. The same .SWP file can be used with this as with an equal-amplitude waveform using the same frequencies, since the sweep file does not deal with amplitude.

C.8.2.2. Command Line Options

There are a number of options for operation of MAKEWAVE.EXE, summarized below. These options may be viewed on screen at any time by typing MAKEWAVE /H (for help) or MAKEWAVE /?.

/L #	Waveform length in samples. (default and max = 8192).
/T	Truncate frequencies (default = round).
/R #	Sampling rate in Hz. (default = 48kHz).
/O #	Oversample ratio. (default = 8).
/F #	Number of coefficients (Taps) in filter. MUST BE ODD. (default = 401).
/M #	Margin (Headroom) to allow in dB (default = 1.0).
/W	Cancel waveform output (Do just .DAF file).
/C #	Conversion type. (default = 1 = .DAT - .WAV & .DAF, (2 = .DAT - .WAA & .DAF, 3 = .WAA - .WAV).
/S	Record length
/A	Absolute amplitude (dBFS) in output file of a unity (one Volt or zero dB) value in input .DAT file

To use one or more of these options, simply type them (separated by one space) on the command line, following the file name to be processed. For example, using FIVE.DAT as the input file and changing the waveform length and sample rate from the default values would be accomplished as follows:

MAKEWAVE FIVE /L2048 /R32000 <Enter>

C.8.2.3. Waveform Length (/L Option) and Record Size (/S Option)

To avoid using a window during FFT processing, the generated waveform length and the analysis record length must be synchronous. This is achieved when the generated waveform length is exactly equal to or an exact integer sub-multiple (1/2, 1/3, etc.) of the analysis record length.

FASTTEST and FASTTRIG have only a few specific analysis record length choices available. The required synchronization between generated and acquired record lengths must be established by controlling the generated record length appropriately for the intended acquisition record length. The /L option of MAKEWAVE determines the period length, in samples, of the lowest frequency sinewave which can be generated. All other frequencies which MAKEWAVE generates will be exact integer multiples of this basic frequency. The /L option thus determines the potential generator frequency resolution and the minimum analyzer acquisition buffer length.

The record size option (/S) determines the size in samples of the waveform file generated. If the waveform length and record size are equal, the lowest frequency possible would go through exactly one cycle in the waveform file. If the waveform length is an integral sub-multiple of the record size, the waveform file would contain two or more exact cycles of the lowest possible frequency. MAKEWAVE will not permit the waveform length to be greater than the record size. The /S option must be used to create the 2048-sample generator record length required to operate FASTTEST.DSP in an SYS-222 without MEM option. The /S option is also used to create shorter generator records for FASTTRIG.DSP in order to permit testing with shorter burst lengths of signal.

Creating Waveforms for FASTTEST.DSP

In full-memory units (SYS-222 with MEM or SYS-322), FASTTEST requires an 8192-sample generator record size. If the waveform length in this record is also 8192 samples, this is exactly 1/2 the 16384 sample analysis record length and must be used in conjunction with the MAXIMUM choice on the FASTTEST.DSP panel. Users can manually choose analysis record lengths of 16384 (selected on the FASTTEST.DSP panel as MAXIMUM when full memory is installed in the DSP) or the shorter analysis records of 4096, 2048, 1024, 512, or 256 samples. For NOISE mode of FASTTEST to work, the analysis record must be twice the waveform length. If NOISE mode is not used, the analysis record may be equal to the waveform length. Thus, for example, an analysis record length selection of 2048 requires a waveform length of 2048 if NOISE mode is not used and 1024 is NOISE is to be measured.

An SYS-222 without MEM option requires a 2048-sample generator record size. An SYS-222 without MEM option has a maximum analysis record length of 4096 samples, selected either by MAXIMUM or 4096. The 2048, 1024, 512, and 256 sample analysis record lengths can also be selected.

MAKEWAVE will by default generate an 8192 sample waveform length and record size when neither the /L nor /S option is used. All the example .WAV files furnished for FASTTEST.DSP were generated as full resolution 8192 sample records except for 15FAST.WAV. This file was generated with the /L2048 /S8192 option for reduced resolution (shorter fundamental period). It is 8192 samples long and repeats exactly four times in the 8192 sample buffer. 15FAST.WAV therefore can be used with the 2048-sample acquisition and transform buffer of FASTTEST.DSP to provide faster testing, although NOISE mode will not function when generator and acquisition buffers are the same length. The example procedure 16FAST.PRO uses this waveform and MON-FAST.TST to demonstrate this technique.

To generate waveforms for the small-memory SYS-222 units, MAKEWAVE must be operated with the /S2048 option to generate waveforms with a 2048-sample length. When only the /Snnnn or the /Lnnnn option is used, MAKEWAVE generate a waveform with both the length and record size set to the specified amount. For example, either MAKEWAVE /L1024 or MAKEWAVE /S1024 will have the same effect, producing a waveform length of 1024 samples in a record size of 1024 samples. If the waveform length and record size are to be different, both options must be specified with different numeric arguments for each, as the example for 15FAST.WAV in the preceding paragraph.

Creating Waveforms for FASTTRIG.DSP

FASTTRIG.DSP only operates with full memory DSP units (SYS-222 with MEM option or SYS-322). FASTTRIG automatically selects an FFT analysis record length twice the downloaded generator signal length. The analysis record length choices automatically selected by FASTTRIG are 16384, 8192, 4096, 2048, 1024, 512, or 256. Thus, the only allowable generator record sizes for FASTTRIG are 8192, 4096, 2048, 1024, 512, or 256 samples. Either the /S or the /L option with the desired record size will cause MAKEWAVE to generate the appropriate waveform.

Longer generator waveforms provide better generator frequency resolution. For example, with 8192 samples and the 48 kHz sample rate all signals must be multiples of the 5.859 Hz basic frequency. With 2048 samples and the 48 kHz sample rate, the basic frequency becomes 23.436 Hz. At 512 samples, the basic frequency is 93.75 Hz. Lower resolution means signals cannot be spaced as close together. This limits the ability to gain detailed information about low-frequency response of the system. Lower resolution becomes even more restrictive when additional criteria are established. For example, it may be desired that signals are not generated in the analyzer bins where AC mains hum signals fall (fundamental, second and third harmonics of AC mains frequency) so that FASTTEST-FASTTRIG measurements will be sensitive to hum. If harmonic distortion produced by low frequency sinewaves is to be measured, the bins at lower order harmonics of all generated sinewaves must be kept free from signals. The reduced resolution of shorter generator records is less likely to be a limitation at high frequencies where it represents a smaller and smaller percentage deviation from the desired frequency.

The advantage of shorter record lengths is the shorter required acquisition time and FFT computation time during measurements. The MAXIMUM analysis record length of FASTTEST or FASTTRIG takes 341.3 milliseconds to acquire at 48 kHz sample rate. A 2048 point analysis record length requires only 42.67 milliseconds to acquire. Computation time of the FFT after acquisition reduces even faster with shorter analysis record lengths. For applications such as a quick check, using only a few tones across the audio spectrum, a shorter sample length and consequently faster acquisition and FFT computation may be appropriate. Required signal burst length with FASTTRIG depends on the generator record length as shown in the table below.

Note that the NOISE mode of FASTTEST.DSP will only function properly when the MAXIMUM transform length is chosen on a full-memory DSP unit (16,384 samples). NOISE mode of FASTTRIG.DSP functions with any length waveform since the acquisition buffer is automatically selected to be twice the length of the generator buffer.

FASTTRIG Generator Record Length vs Attributes (48 kHz sample rate)		
Generator Record Length	Minimum Burst Length	Gen. Lowest Freq. & Minimum Spacing
8192	1.050 sec.	5.86 Hz
4096	520 ms	11.7 Hz
2048	260 ms	23.4 Hz
1024	135 ms	46.9 Hz
512	70 ms	93.7 Hz
256	40 ms	187.5 Hz

C.8.2.4. Truncate vs Round (/T Option)

With the default rounding operation, MAKEWAVE.EXE will round each user-requested frequency to the nearest "legal" frequency, with a consequent maximum discrepancy of one-half the basic frequency (±2.93 Hz with 8192 samples and 48 kHz). If truncation is selected by the /T option, MAKEWAVE.EXE will select the largest "legal" frequency equal to or smaller than the user-requested frequency. The resulting discrepancy can approach the basic frequency (5.859 Hz in this case).

C.8.2.5. Sampling Rate (/R# Option)

The sampling rate selected when generating files for the waveform and sweep table must be the exact sampling rate which will be used during acquisition and FFT analysis of the signal. Failure to use the same sampling rate during MAKEWAVE.EXE operation and actual waveform generation from FAST-TEST-FASTTRIG.DSP will scale all frequencies up or down by the ratio of the two sampling rates.

C.8.2.6. Oversample Ratio (/O# Option)

In producing the .WAV file, MAKEWAVE.EXE computes the peak amplitude of the complex signal and then scales the amplitudes of all components so that the peak approaches but stays below digital full scale. The oversample ratio used affects the accuracy with which MAKEWAVE.EXE predicts the amplitude of the complex waveform. This oversample ratio effectively determines the interpolation between waveform samples. A higher oversample ratio reduces the likelihood that peaks will fall between interpolated points, but increases the execution time of MAKEWAVE.EXE. If peaks are missed, the actual complex waveform will have peaks higher than predicted by MAKEWAVE.EXE and clipping could occur. Operation of MAKE-WAVE can be significantly speeded by using smaller oversample ratios, but with increased risk that actual complex signal peaks may exceed full scale. For applications where use of the full dynamic range is not important, the headroom option (see below) can be used in conjunction with a low oversample ratio to save time during waveform file generation. Oversample ratio must be an even number; two is the minimum acceptable.

C.8.2.7. Digital Filter (/F# Option)

This option determines the number of coefficients (taps) in the digital filter used in the computation of complex output waveform amplitude. A higher number of coefficients produces a more accurate estimate of the complex signal amplitude, but increases the execution time of MAKEWAVE.EXE. Operation of MAKEWAVE can be significantly speeded by using smaller numbers of taps, but with increased risk that actual complex signal peaks may exceed full scale. For applications where use of the full dynamic range is not important, the headroom option (see below) can be used in conjunction with a lower number of taps to save time during waveform file generation. The number of taps must be an odd number.

C.8.2.8. Headroom (/M# Option)

After computing the peak amplitude of the complex signal, MAKEWAVE.EXE sets that computed peak below digital full scale by an additional margin for headroom. The default value is one dB. If the oversampling and filter taps were sufficient to accurately estimate the peak amplitude, this default margin may be decreased in order to obtain a small improvement in signal-to-noise ratio in the signal. If lower values of oversampling and a smaller number of filter taps are used to speed execution of MAKEWAVE.EXE, a headroom value larger than one dB should be used to avoid the chance of clipping the generated signal.

C.8.2.9. No Waveform (/W Option)

If the waveform files (.WAV) are not desired, but merely the .DAF file which ultimately leads to .SWP tables, MAKEWAVE.EXE will accomplish this very rapidly (1-2 seconds) by using the /W option. This is useful to permit trial-and-error experimentation in finding the best frequency list without taking the time to generate the waveform file.

C.8.2.10. Absolute Amplitude (/A Option)

When the /A option is not used, MAKEWAVE automatically sets the time domain peak amplitude of the output .WAV file to -1.00 dBFS. The one dB headroom allowance is to cover possible small variations between the composite peak as computed by MAKEWAVE versus the actual amplitude at the output of the reconstruction filter following the D/A converter in the DSP module.

When creating waveforms for stereo separation tests, it is strongly desirable that the sinewaves in both left and right channel waveforms have the same amplitude. If they do not, a device will appear to have stereo imbalance. The /A option is used to force this equal amplitude condition. It is normally used by first running both the left and right channel .DAT files through MAKEWAVE without using the /A option, and noting the last message on the screen when MAKEWAVE runs. This message reads:

"Absolute output level = -nn.nn dBFS (corresponding to 1 Volt or 0 dB input file specification)"

For the typical case of an input .DAT file with all amplitudes specified at zero dBV (or zero in any other dB unit) and no positive dB values, this value shows the highest that a zero dB input can be placed without the composite signal exceeding -1.00 dBFS. Note the numbers from the left and right channel waveforms. Then, run both .DAT files again, using the /A option with the same numeric value for both channels. The numeric value must be chosen so that neither waveform will exceed full scale. For example, if the left waveform produces an absolute level of -20.43 dBFS and the right waveform produces an absolute level of -21.29 dBFS, run them both again with the /A-21.29 argument. The right channel will now still have about 1.0 dB of headroom; the left channel will have about 1.86 dB of headroom, and the sinewave amplitudes will be equal on both channels.

C.8.3. Making Sweep Files from .DAF or .DAD Files

To create a .SWP file from one of the .DAF or .DAD files, load a .TST file with appropriate units; typically, this will be the test file you plan to use with the .SWP file. Then, use the LOAD DATA command. Only the names of .DAT files will appear, but enter *.DAF (for fundamental files) or *.DAD (for distortion product files) and press <Enter>. All of the files in the current directory with .DAF extensions will then appear. Select the one you wish with the arrow keys and cursor and press <Enter>. Now, select SAVE SWEEP and supply the file name you wish to be used.

C.8.4. Waveform Files for Stereo Separation Testing

To make your own stereo waveform files which support stereo separation testing, each channel waveform must be made individually. Create two separate .DAT files, each listing the frequencies desired in the specific channel. Save them with names indicating which channel each is intended for; for example, ABCD-L.DAT and ABCD-R.DAT. Run MAKEWAVE.EXE individually for each .DAT file, using the /A capability to force the same sinewave amplitudes in these two files. Then, start S1.EXE software with FASTTRIG.DSP or FASTTEST.DSP loaded. Load the left channel waveform file using the "1G" argument. Load the right channel waveform using the "2G". Use the SAVE WAVEFORM command, supplying a final desired file name which should indicate that it is a stereo file. (The furnished stereo files for FASTTRIG all use "S" as the fifth character). Supply the "1G,2G" argument so that both channel buffers will be saved into this single disk file. This stereo waveform file can then be loaded into both buffers by a single LOAD WAVEFORM command and the "1G,2G" argument.

C.8.5. Using MAKEDIST.EXE

FASTTEST and FASTTRIG both measure the combination of harmonic and intermodulation distortion plus noise with the MEASURE DISTORT panel selection, as described earlier. ***This is the recommended method of distortion measurement for most applications.***

If separate harmonic or intermodulation distortion testing is desired, MAKEDIST.EXE can be used to process the .DAF file created by MAKEWAVE.EXE into .DAD files which contain the exact frequencies at which specified distortion products will fall. The input .DAF file lists the exact fundamental frequencies which will be generated by the DSP unit when the waveform file is downloaded. MAKEDIST.EXE is capable of handling .DAF files with fifty or fewer fundamental frequencies. The output .DAD file, depending upon user-specified command line options, may include harmonic product frequencies (odd only, or both odd and even), intermodulation product frequencies, fundamentals, or combinations as desired. The starting and ending distortion product order may be specified by the user. ***If only frequency response, total distortion and noise, phase, and noise testing is planned, MAKEDIST.EXE need not be run since the .DAF file produced by MAKEWAVE contains all the fundamental frequencies required for those modes.***

The .DAF from MAKEWAVE and the .DAD files from MAKEDIST are in S1.EXE data file format. They can be imported to S1.EXE software using the LOAD DATA command and then saved as a sweep table (.SWP file). That sweep table, when attached to a .TST file, causes S1.EXE to ask the DSP module only for data at the specific frequencies listed. A particular .SWP file can consist of:

1. the fundamental frequencies only, which thus asks for amplitudes to be evaluated as a frequency response test (MEASURE NORMAL and MEASURE RESPwW+F), total distortion plus noise test (MEASURE DISTORT), noise test (MEASURE NOISE), or phase values for a phase test

2. harmonic frequencies only, which asks for amplitudes to be evaluated as a harmonic distortion test (MEASURE NORMAL or MEASURE RESPwW+F)

3. intermodulation frequencies (of two-tone combinations only) which asks for amplitudes to be evaluated as an intermodulation distortion test (MEASURE NORMAL or MEASURE RESPwW+F)

4. any combination of the above, which permits one test to simultaneously determine frequency response and either or both types of distortion (MEASURE NORMAL or MEASURE RESPwW+F)

To run MAKEDIST.EXE, type the program name followed by the file name of the .DAF file to be used as input. To continue the previous example, typing:

MAKEDIST FIVE <Enter>

will take FIVE.DAF as input and produce FIVE.DAD as output, using the default values of the command line options described below. If you wish a different name for the .DAD output from the .DAF input, type the desired new file name on the command line in addition to the input file name as described above under MAKEWAVE.EXE. When generating several sweep tables which will all be used with one waveform file, different names are required in order to avoid over-writing previously-generated harmonic-only, intermod-only, or combined parameter files.

C.8.5.1. Command Line Options, MAKEDIST

There are several command line options for MAKEDIST.EXE, summarized below. These may be displayed on screen by typing MAKEDIST /H (for help) or MAKEDIST /?.

/R#	Sampling rate in Hz. (default 48000)
/V	Verbose—print info while calculating
/F	Do fundamentals (default = no)
/D#	Do harmonics to this order (0 = none) (default = 5)
/O#	Do Odd Harmonics to this order (0 = none) (default = 0)
/I#	Do IM products to this order (0 = none) (default = 3)
/S#	Set starting harmonic (default = 2, 3 for odds)
/H	Help. Print this message.

C.8.5.2. Sampling Rate (/R# Option)

It may sometimes be desirable to use a .WAV file generated at one sampling rate with FASTTEST-FASTTRIG.DSP set to a different sampling rate. To see where the frequencies will fall at the other sampling rate, MAKEDIST.EXE can be run with the /R value specifying the different rate. The /F /I0 /D0 options would normally be used in this case, producing fundamentals only in the .DAD output file.

C.8.5.3. Verbose Output (/V Option)

This option causes the program to print a complete "play-by-play" description of its current activity.

C.8.5.4. Fundamentals (/F Option)

MAKEDIST.EXE will include the sinewave fundamental frequencies of the signal in its output file only if the /F option is used. Fundamental frequencies are required in a .SWP table if a combined evaluation of frequency response is desired along with one or more distortion parameters. If only frequency response, total distortion and noise, phase, and noise are to be measured, it is not necessary to run MAKEDIST.EXE at all since the .DAF file already consists of the exact fundamental frequencies. In this case, the .DAF file can be loaded into S1.EXE with the LOAD DATA command and the resulting file saved as a .SWP file.

C.8.5.5. Harmonics (/D# Option)

If harmonic distortion is to be evaluated, the sweep table must cause S1.EXE to ask the DSP for amplitude measurements at the exact harmonics of some or all of the sinewave signals. The default value for harmonic order is five. Thus, if MAKEDIST.EXE is run without using the /D# option, it generates in the output file frequency values at twice, three times, four times, and five times each input fundamental frequency except as limited by half the sample rate (Nyquist frequency). If harmonic distortion evaluation is desired but to lower or higher order harmonics, the /D# option must be used with # being the highest order desired. If a list of harmonic frequencies starting higher than second harmonic is desired, the /S# option (see below) will define a higher starting value. If it is desired to measure harmonic distortion at some but not all of these frequencies, the undesired values can be edited out in the S1.EXE EDIT DATA mode after loading the .DAD file as data and before saving as a .SWP file. If harmonic distortion evaluation is not desired, the /D0 option should be used.

To continue our example, a list of second and third harmonic distortion products of FIVE.DAF can be made by typing

MAKEDIST FIVE /D3 /I0 FIVEHARM <Enter>

The /D3 option specifies a list of second and third harmonic frequencies to be generated. The /I0 option suppresses any generation of intermodulation products. Fundamentals will not be generated unless the /F option is invoked. FIVEHARM specifies the name of the .DAD file which will contain those frequencies, as limited by the Nyquist frequency. The resulting output file is illustrated below:

Hz	V	OFF
140.625,	1,	0
210.9375,	1,	0
539.0625,	1,	0
808.59375,	1,	0
2039.0625,	1,	0
3058.59375,	1,	0
7839.84375,	1,	0
11759.7656,	1,	0

FIVEHARM.DAD can then be loaded into S1.EXE and used to make a sweep table which will be used with FASTTEST-FASTTRIG in NORMAL mode to measure harmonic distortion.

C.8.5.6. Odd Harmonics (/O# Option)

When a list of only odd harmonics is desired, the /O# option may be used instead of the /D# option. The number value supplied defines the highest odd harmonic which will be calculated. If it is desired that the list not start with the third harmonic, the /S# option (see below) may also be used to define a higher harmonic order as the starting point.

As an example, assume the input .DAF file contains only the single frequency 47.875 Hz. Runing MAKEDIST with the /O7 option would produce an output .DAD file with the frequencies:

140.625 (3rd harmonic)
234.375 (5th harmonic)
328.125 (7th harmonic).

Using MAKEDIST with the /O7 /S5 options would produce a .DAD file with the frequencies:

234.375 (5th harmonic)
328.125 (7th harmonic).

C.8.5.7. Intermodulation (/I# Option)

If intermodulation distortion is to be evaluated, the sweep table must cause S1.EXE to ask the DSP for amplitude measurements at computed intermodulation product frequencies of some or all of modulation product frequencies of some or all of the sinewave signals. The default value for IMD product order is three. If MAKEDIST.EXE is run without using the /I# option, it generates in the output file the frequency values of all second order products and all third order products of input fundamental frequency tone pairs, except as limited by half the sample rate (Nyquist frequency). If any two fundamental frequencies are described as f_m and f_n, the resulting IMD product frequencies computed by MAKEDIST.EXE are as follows:

Second Order (/I2)
$f_m \pm f_n$

Third Order (/I3)
$f_m \pm 2f_n$
$2f_m \pm f_n$

Fourth Order (/I4)
$f_m \pm 3f_n$
$2f_m \pm 2f_n$
$3f_m \pm f_n$

Note that for third (and higher) order IMD evaluation, IMD products will also be generated in the device under test as the result of interaction of three (or more) fundamentals. MAKEDIST.EXE does not compute any of the frequencies resulting from these more complex interactions. It only computes products resulting from intermodulation of all possible sets of two fundamentals. To perform a three-tone IMD test, it will be necessary to separately compute the possible IM product frequencies and manually enter them into the .SWP file.

If intermodulation distortion evaluation is desired but to lower or higher order products than third order, the /I# option must be used with # being the highest order desired. If it is desired to measure intermodulation distortion at some but not all of these possible frequencies, the undesired values can be edited out in the S1.EXE EDIT DATA mode after loading the .DAD file as data and before saving as a .SWP file. If IMD is not to be evaluated, the /I0 option should be used.

Operating MAKEDIST with FIVE.DAF to create a list of second and third order intermodulation product frequencies is done as follows:

MAKEDIST FIVE /D0 FIVEIMD <Enter>

The /D0 option suppresses generation of harmonic distortion frequencies. Fundamentals are not generated unless the /F option is used. Second and third order is the default operation of MAKEDIST, so the /I option need not be used. FIVEIMD is supplied as the name of the output .DAD file to be created. The result is illustrated below:

Hz	V	OFF
128.90625,	0,	1
199.21875,	0,	1
339.84375,	0,	1
410.15625,	0,	1
468.75,	0,	1
480.46875,	0,	1
609.375,	0,	1
750,	0,	1
878.90625,	0,	1
949.21875,	0,	1
1089.84375,	0,	1
1160.15625,	0,	1
1289.0625,	0,	1
1558.59375,	0,	1
1769.53125,	0,	1
1880.85937,	0,	1
1968.75,	0,	1
2109.375,	0,	1
2308.59375,	0,	1
2900.39063,	0,	1
3380.85937,	0,	1
3650.39062,	0,	1
3779.29687,	0,	1
3849.60937,	0,	1
3990.23437,	0,	1
4060.54687,	0,	1
4189.45312,	0,	1
4458.98437,	0,	1
4939.45312,	0,	1
5958.98437,	0,	1
6820.3125,	0,	1
7160.15625,	0,	1
7570.3125,	0,	1
7769.53125,	0,	1
7910.15625,	0,	1
8109.375,	0,	1
8859.375,	0,	1
11080.0781,	0,	1
12960.9375,	0,	1

13980.4688,	0,	1
14460.9375,	0,	1
14730.4688,	0,	1
14859.375,	0,	1
14929.6875,	0,	1
15070.3125,	0,	1
15140.625,	0,	1
15269.5313,	0,	1
15539.0625,	0,	1
16019.5313,	0,	1
17039.0625,	0,	1
18919.9219,	0,	1
22839.8438,	0,	1

FIVEIMD.DAD can then be loaded into S1.EXE and used to make a sweep table which will be used with FASTTEST-FASTTRIG in NORMAL mode to measure intermodulation distortion.

The .DAD file may contain frequencies which you do not want to be in a particular .SWP file. For example, you may wish to measure harmonics up through the fifth harmonic of certain fundamental frequencies but only up through the third harmonic of others. MAKEDIST.EXE, if the /D5 option was used, will have generated frequencies of the second through fifth harmonic of all fundamental tones except as limited by the Nyquist frequency. Use the EDIT DATA command after loading the .DAD file. Use the <F6> key to delete any lines containing unwanted frequencies, then <Esc> to the menu and SAVE SWEEP.

C.8.5.8. Starting Harmonic (/S# Option)

The /S# option permits specifying the lowest value harmonics to be calculated. If /S is not used or used without a numeric argument, the list will start with the second harmonic except when the /O (odd harmonics) option is also used; in this case the default starting value is third harmonic. The command MAKEDIST with the options /D8 /S4 would produce a list of fourth through eighth harmonics of all frequencies in the input .DAF file (truncated at half the sampling rate). Running MAKEDIST with the options /O9 /S5 would produce a list of fifth, seventh, and ninth harmonics of the input frequencies.

C.9. Furnished Files

This section lists the files which are furnished along with the FASTTEST and FASTTRIG.DSP programs on the diskettes. The furnished files for FASTTEST.DSP and FASTTRIG.DSP are located on separate diskettes and are discussed in separate sections below. Waveform and sweep files distributed for FASTTRIG could be used with FASTTEST and vice-versa. Most procedures furnished for FASTTEST cannot be used with FASTTRIG without modification, due to the differences in the LOAD WAVEFORM arguments used when downloading generator waveforms.

C.9.1. Furnished Files, FASTTEST.DSP

Several procedures are included on this diskette. Each loads a test, loads the appropriate waveform, and then runs a series of tests. These tests measure several key audio characteristics of a device connected from the generator outputs to the analyzer inputs. The procedures include MONAURAL.PRO, STER15F.PRO, STEREO.PRO, STEREOLC.PRO, and XTLK.PRO. Each loads .TST or .OVL files to make the required tests. Each then saves the results under the file names RESPONSE.TST, DISTORT.TST, NOISE.TST, and PHASE.TST. Thus, these four file names will always contain the results of the most recently-run procedure. If it is desired to retain any or all of these test results, the tests must be copied to another directory, or re-named and saved, before running any of these procedures again. Otherwise, the test results will be over-written with the results of the next procedure.

MONAURAL.PRO is designed for single-channel (monaural) audio devices connected to the CHANNEL A OUTPUT and INPUT. It measures response, total distortion and noise, and empty-bin noise.

STEREO.PRO is designed for dual-channel (stereo) audio devices connected to both CHANNEL A and B. In addition to the functional measurements of MONAURAL.PRO, it measures interchannel phase.

STER15F.PRO is designed for still-faster evaluation of stereo audio devices. By using 15FAST.WAV, which was created with the 2048-sample record length option of MAKEWAVE.EXE, the test files can all use a 2048-sample acquisition and transform buffer. This reduces acquisition time from the normal 341 milliseconds to 42.7 milliseconds and also reduces the time required to compute the FFT. Empty-bin noise cannot be measured by this procedure. Since the generator and acquisition buffers are both the same length, there are no alternate empty bins. The total distortion and noise test (DIST-STF.TST) will show zero distortion and noise at frequencies below 100 Hz due to resolution limitations. With the 2048-sample transform length, analysis bins are about 23.4 Hz wide (48000/2048). DISTORT mode measures the amplitude of all bins between the bins containing fundamental tones. The four lowest sinewave fundamentals of 15FAST.WAV are at 23.44 Hz, 46.88 Hz, 70.31 Hz, and 117.2 Hz. Thus, the lowest-frequency bin not containing a fundamental is at 93.75 Hz and its amplitude will be plotted at 117 Hz in DISTORT mode.

STEREOLC.PRO uses the LOWCREST.WAV waveform which has the same 31 tones as ISO31.WAV plus an added 1125 Hz component 30 dB higher in amplitude. The resulting crest factor is nearer to that of a single sinewave. STEREOLC.PRO uses overlay files (.OVL) rather than test files (.TST) following the initial test. The procedure first loads the initial test and executes the <F4> keystroke to set the dBr REF value from the measured level. Since the RMS amplitude of LOWCREST.WAV is dominated by the 1125 Hz signal 30 dB higher than the other tones, this dBr reference value represents (within about 0.2 dB) the value of the 1125 Hz tone and is useful for all the remaining tests. Response is centered 30 dB below that value. The distortion and noise .OVL files have the dBr REF value "punched out", so the value remaining from the <F4> keystroke is retained and propagates down through the procedure. In addition to the response, noise, total distortion and noise, and phase tests of the other stereo procedures, STEREOLC.PRO includes DIST1KLC.OVL which is then saved as DIST1K.TST. This test measures the 2nd through 9th harmonic amplitudes of the 1125 Hz tone.

XTLK.PRO requires external stereo D/A converters to operate, since the DSP unit only has one D/A built in. An RDAT machine may be used in digital input mode, with its analog outputs feeding the stereo device under test. XTLK.PRO also uses .OVL files in a similar fashion to STEREOLC.PRO described above. XTLK.PRO loads XTLKLEFT.WAV into DSP generator buffer one and XTLKRIGH.WAV into DSP generator buffer 2. These waveforms both contain all the frequencies of ISO31.WAV. Each also contains four unique frequencies not duplicated in the other. XTLK.PRO measures response and phase with the signals which are common to both channels. It measures total distortion and noise with a .SWP file containing all the frequencies of both channels. It measures crosstalk from left-to-right and right-to-left with .SWP files containing only the unique frequencies being generated on the opposite channel. Empty-bin noise is measured with the 31-tone sweep.

A detailed list follows of these FASTTEST.DSP procedures with the .TST or .OVL files called and .WAV and .SWP files used.

15FAST.SWP fundamental frequency list of 15FAST.WAV

15FAST.WAV 15 log-spaced tones, 2048 record length

1KDISTLC.SWP 2nd thru 9th harmonics of the 1125 Hz high-amplitude tone of LOWCREST.WAV

31FUNDLC.SWP all 32 fundamentals of LOWCREST.WAV (ISO31 plus an 1125 Hz signal 30 dB higher)

DIST-MON.TST total distortion & noise, monaural, stored with ISO31.SWP

DIST-ST.TST same as DIST-MON.TST but stereo

DIST-STF.TST same as DIST-ST.TST but 2048-sample acquisition (fast)

DIST-STF.TST same as DIST-ST.TST but 2048-sample acquisition (fast)

DIST1KLC.OVL uses 1KDISTLC.SWP, measures 2nd thru 9th harmonic distort. of 1125 Hz high-amplitude tone of LOWCREST.WAV

DISTSTLC.OVL total distortion & noise, stereo, for LOWCREST.WAV

FASTTEST.DSP DSP program without capture feature

FIVE.WAV five-tone signal per example on how to generate waves

FIVEDIST.TST total distortion & noise, stereo, for FIVE.WAV

FIVEFUND.SWP fundamentals only of FIVE.WAV

FIVEHARM.SWP 2nd & 3rd harmonics of FIVE.WAV

FIVEHARM.TST test for harmonics of FIVE.WAV, uses FIVEHARM.SWP

FIVEIMD.SWP 2nd & 3rd order IMD products of FIVE.WAV

FIVEIMD.TST test for IMD of FIVE.WAV, uses FIVEIMD.SWP

FIVENOIS.TST empty-bin noise of FIVE.WAV, uses FIVEFUND.SWP

FIVEPHAS.TST inter-channel phase with FIVE.WAV, uses FIVEFUND.SWP

FIVERESP.TST stereo response with FIVE.WAV, uses FIVEFUND.SWP

ISO31.SWP fundamentals of ISO31.WAV

ISO31.WAV 31 tones

ISO60.SWP fundamentals of ISO60.WAV

ISO60.WAV 60 tones

LOWCREST.WAV ISO31.WAV plus additional 1125 Hz signal @ +30 dB

MONAURAL.PRO procedure for monaural devices, includes RESP-MON.TST, DIST-MON.TST, NOIS-MON.TST

NOIS-MON.TST empty-bin noise, 31-tone sweep

NOIS-ST.TST same as NOIS-MON.TST but stereo

NOISSTLC.OVL empty-bin noise for low-crest waveform

PHAS-ST.TST interchannel phase

PHAS-STF.TST interchannel phase with 15FAST.WAV

PHASELC.OVL interchannel phase with LOWCREST.WAV

RESP-MON.TST monaural frequency response

RESP-ST.TST stereo frequency response

RESP-STF.TST stereo frequency response with 15FAST.WAV

RESPSTLC.TST stereo frequency response with LOWCREST.WAV

RESPTAPE.TST stereo frequency response of tape machines with W&F

STER15F.PRO procedure for faster evaluation of stereo devices with 15FAST.WAV, uses RESP-STF.TST, PHAS-STF.TST, DIST-STF.TST

STEREO.PRO procedure for stereo devices, uses RESP-ST.TST, PHAS-ST.TST, DIST-ST.TST, NOIS-ST.TST

STEREOLC.PRO procedure for stereo devices using low crest-factor 32-tone signal with 1125 Hz sinewave 30 dB higher than others. Uses RESPSTLC.TST, PHASELC.OVL, DISTSTLC.OVL, DIST1K.OVL, NOISSOVL.TST

XDISTORT.OVL stereo total distortion & noise with crosstalk signals, uses XTLKDIST.SWP

XNOISE.OVL empty-bin noise with crosstalk signals, uses ISO31.SWP

XPHASE.OVL interchannel phase with crosstalk signals, uses ISO31.SWP

XRESP.TST stereo response with crosstalk signals, uses ISO31.SWP

XTLK.PRO procedure for stereo devices, requires external stereo D/A converters such as RDAT machine, uses XTLKLEFT.WAV, XTLKRIGH.WAV, tests are XRESP.TST, XPHAS.OVL, XDISTORT.OVL XNOISE.OVL, XTLK_L-R.OVL, XTLK_R-L.OVL

XTLKDIST.SWP list of all fundamentals in both channels of crosstalk signals

XTLKLEFT.WAV ISO31.WAV with four additional unique frequencies near 300 Hz, 1 kHz, 3 kHz, 10 kHz

XTLKRIGH.WAV similar to XTLKLEFT.WAV but different frequencies added

XTLK_L-R.OVL crosstalk from L to R at four L-channel unique frequencies

XTLK_L-R.SWP list of L-channel unique frequencies

XTLK_R-L.OVL crosstalk from R to L at four R-channel unique frequencies

XTLK_R-L.SWP list of R-channel unique frequencies

C.9.2. Furnished Files, FASTTRIG.DSP

C.9.2.1. Supplied Waveform Files and Sweep Tables

Thirty-two waveform files (.WAV) are furnished for FASTTRIG.DSP. Sweep tables (.SWP) are furnished for all the waveforms supplied for FASTTRIG.

The nomenclature system used for the FASTTRIG waveform files consists of seven characters. The first three characters describe the length of the waveform as follows:

8K_nnnn.WAV	8192 samples
4K_nnnn.WAV	4096 samples
2K_nnnn.WAV	2048 samples
1K_nnnn.WAV	1024 samples
512nnnn.WAV	512 samples
256nnnn.WAV	256 samples

The length of the waveform determines the worst-case burst length which FASTTRIG is guaranteed to capture. The 8k waveforms require approximately one second, 4k waveforms about one-half second, 2k waveforms one-quarter second, etc.

Generator frequency resolution reduces directly as the waveform length reduces. With the longer waveforms (8k, 4k, and 2k) the goal was to place tones near the ISO standard 1/3 octave frequencies. This goal was necessarily compromised more at low frequencies (especially in the shorter records) due to the limited resolution and the desire to leave analyzer bins open to measure hum. At the very short files (1k, 512, and 256 samples) the goal was 1/2 octave tone spacing. Again, the compromise is greatest at low frequencies.

The fourth character describes the audio spectrum width occupied by the tones. F stands for "full" audio bandwidth, and B for "broadcast" bandwidth:

nnnFnnn.WAV	20 Hz-20 kHz
nnnBnnn.WAV	50 Hz-15 kHz

8k Waveform Family. Burst Length to Acquire Approximately One Second					
Gen. Record Length 8192 bytes. Resolution 5.86 Hz. Analyzer Record Length 16384 bytes, Res. 2.93 Hz					
8K_FSU_.WAV		8K_BSU_.WAV		8K_FMU_.WAV	8K_BMU_.WAV
Sinewave Amplitudes -25.1 dBFS		Sinewave Amplitudes -24.6 dBFS		-23.8 dBFS	-23.5 dBFS
Frequencies Common to Both Channels (Hz)	Unique Left and Right Frequencies (Hz)	Frequencies Common to Both Channels (Hz)	Unique Left and Right Frequencies (Hz)	Frequencies (Hz)	Frequencies (Hz)
23				23	
29				29	
41		44.1k		41	
53		53 53.833		53	53
64		64 64.5996		64	64
76		76 75.3662		76	76
94		94 96.89914		94	94
117		117 ~~118.435~~	129.199	117	117
164		164 166.882		164	164
205		205 204.565		205	205
252		252 253.015		252	252
316		316 317.615		316	316
398		398 398.364		398	398
498	539L	~~498~~ ~~495.264~~	539L 489.8863	498	498
627	586R	627 629.846	586R 506.6302	627	627
803		803 802.112		803	803
1002	1084L	~~1002~~ ~~1001.29~~	~~1084L~~ 990.5272	1002	1002
1248	1166R	1248 1248.93	~~1166R~~ 1001.29	1248	1248
1600		1600 1598.84		1600	1600
1998	2162L	~~1998~~ ~~1997.20~~	2162L 1997.20	1998	1998
2501	2326R	2501 2503.23	2326R 2018.7375	2501	2501
3152		3152 3149.23		3152	3152
4002	4336L	~~4002~~ ~~3999.79~~	~~4336L~~ 3983.642	4002	4002
4998	4664R	4998 4995.70	4664R 4026.7084	4998	4998
6298		6298 6303.85		6298	6298
7998	8666L	~~7998~~ ~~8004.97~~	8666L 7983.4339	7998	7998
10002	9334R	10002 10002.20	9334R 8058.8004	10002	10002
12498	13670L	12498 12500.023	12489.256	12498	12498
16002	14836R	15000 14997.9	12516.148	16002	15000
19998		16004.6		19998	

Table C-1 *Frequencies Included in 8192-Sample Waveforms*

2299 = 12376.2096
2345 = 12623.8396

Detailed information on all 32 waveforms furnished for FASTTRIG may be found in Tables C-1 through C-9

12437.5, 2310 = 12435.423
12562.5, 2334 = 12564.622

C.9.2.2. Stereo Separation

The fifth character describes whether the waveform has different signals on the two stereo channels in order to measure stereo separation, or whether it is a single-channel waveform. S repre-

4k Waveform Family. Burst Length to Acquire Approximately 500 Milliseconds					
Gen. Record Length 4096 bytes. Resolution 11.7 Hz. Analyzer Record Length 8192 Bytes. Res. 5.86 Hz					
4K_FSU_.WAV		4K_BSU_.WAV		4K_FMU_.WAV	4K_BMU_.WAV
Sinewave Amplitudes -23.5 dBFS		Sinewave Amplitudes -24.1 dBFS		-22.8 dBFS	-23.6 dBFS
Frequencies Common to Both Channels (Hz)	Unique Left and Right Frequencies (Hz)	Frequencies Common to Both Channels (Hz)	Unique Left and Right Frequencies (Hz)	Frequencies (Hz)	Frequencies (Hz)
23				23	
35				35	
47				47	
70		70		70	70
82		82		82	82
105		105		105	105
129		129		129	129
176		176		176	176
199		199		199	199
234		234		234	234
305		305		305	305
410		410		410	410
504		504		504	504
633		633		633	633
797		797		797	797
996	1078L	996	1078L	996	996
1254	1172R	1254	1172R	1254	1254
1605		1605		1605	1605
2004	2168L	2004	2168L	2004	2004
2496	2332R	2496	2332R	2496	2496
3152		3152		3152	3152
3996	4324L	3996	4324L	3996	3996
5004	4676R	5004	4676R	5004	5004
6293		6293		6293	6293
8004	8660L	8004	8660L	8004	8004
9996	9340R	9996	9340R	9996	9996
125042		125042		125042	125042
15996		15000		15996	15000
20004				20004	

Table C-2 *Frequencies Included in 4096-Sample Waveforms*

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sents "stereo" and M represents "monaural". The "S" waveforms can be downloaded into both generator channel buffers in one operation by supplying the arguments "1G,2G" after the LOAD WAVEFORM command and selection of the .WAV filename. The "S" waveforms can only be transmitted from a digital output of System One Dual Domain, since System One has only one D/A converter and a single generator output amplifier and transformer with the signal then split to the two output connectors. For stereo separation testing of analog-domain devices, external dual D/A converters are required, such as those in a DAT machine. A DAT cassette can be recorded with the System One

2k Waveform Family, 50 Hz AC Mains. Burst Length to Acquire Approximately 250 Milliseconds					
Gen. Record Length 2048 bytes. Resolution 23.4 Hz. Analyzer Record Length 4096 Bytes. Res. 11.7 Hz					
2K_FS50.WAV		2K_BS50.WAV		2K_FM50.WAV	2K_BM50.WAV
Sinewave Amplitudes -22.6 dBFS		Sinewave Amplitudes -21.5 dBFS		-21.7 dBFS	-20.9 dBFS
Frequencies Common to Both Channels (Hz)	Unique Left and Right Frequencies (Hz)	Frequencies Common to Both Channels (Hz)	Unique Left and Right Frequencies (Hz)	Frequencies (Hz)	Frequencies (Hz)
23				23	
70		70		70	70
94		94		94	94
117		117		117	117
164		164		164	164
211		211		211	211
352		352		352	352
516		516		516	516
656		656		656	656
820		820		820	820
1008		1008		1008	1008
1242		1242		1242	1242
1594		1594		1594	1594
1992	2156L	1992	2156L	1992	1992
2508	2344R	2508	2344R	2508	2508
3141		3141		3141	3141
4008	4336L	4008	4336L	4008	4008
4992	4664R	4992	4664R	4992	4992
6305		6305		6305	6305
7992	8648L	7992	8648L	7992	7992
10008	9352R	10008	9352R	10008	10008
12492		12492		12492	12492
16008	17320L	15000		16008	15000
19992	18680R			19992	

Table C-3 *Frequencies Included in 2048-Sample Waveforms for Hum Measurement for 50 Hz Power*

Dual Domain digital output driving the DAT digital input. The analog outputs of the DAT are then used as the stereo test signal source while playing the cassette.

When "M" waveforms are downloaded to the DSP, the LOAD WAVEFORM command must be used twice. The same .WAV filename is supplied in both cases, with the "1G" argument furnished the first time and the "2G" argument the second time. ***Stereo device frequency response, distortion, noise, and interchannel phase may all be measured using an "M" waveform and the analog outputs of System One; only stereo separation testing requires the "S" waveform and external D/A converters.***

C.9.2.3. Hum Measurement

The sixth and seventh characters describe the ability of each waveform to be used when it is desired to measure power line-related hum:

2k Waveform Family, 60 Hz AC Mains. Burst Length to Acquire Approximately 250 Milliseconds					
Gen. Record Length 2048 bytes. Resolution 23.4 Hz. Analyzer Record Length 4096 Bytes. Res. 11.7 Hz					
2K_FS60.WAV		2K_BS60.WAV		2K_FM60.WAV	2K_BM60.WAV
Sinewave Amplitudes -23.0 dBFS		Sinewave Amplitudes -22.0 dBFS		-22.1 dBFS	-21.2 dBFS
Frequencies Common to Both Channels (Hz)	Unique Left and Right Frequencies (Hz)	Frequencies Common to Both Channels (Hz)	Unique Left and Right Frequencies (Hz)	Frequencies (Hz)	Frequencies (Hz)
23				23	
70		70		70	70
94		94		94	94
141		141		141	141
164		164		164	164
211		211		211	211
352		352		352	352
516		516		516	516
656		656		656	656
820		820		820	820
1008		1008		1008	1008
1242		1242		1242	1242
1594		1594		1594	1594
1992	2156L	1992	2156L	1992	1992
2508	2344R	2508	2344R	2508	2508
3141		3141		3141	3141
4008	4336L	4008	4336L	4008	4008
4992	4664R	4992	4664R	4992	4992
6305		6305		6305	6305
7992	8648L	7992	8648L	7992	7992
10008	9352R	10008	9352R	10008	10008
12492		12492		12492	12492
16008	17320L	15000		16008	15000
19992	18680R			19992	

Table C-4 *Frequencies Included in 2048-Sample Waveforms for Hum Measurement for 60 Hz Power*

U_ Universal, can be used for hum measurements at 50 Hz and/or 60 Hz
N_ Cannot be used for hum measurements
50 Can be used for hum from 50 Hz mains
60 Can be used for hum from 60 Hz mains

Waveforms with the "U_" nomenclature do not generate sinewave fundamentals which fall into the analyzer bins which will measure signal at 50, 60, 100, 120, 150, or 180 Hz (second and third harmonics of the power mains fundamental frequencies).

"N_" waveforms have signals in one or more of both the 50-Hz and 60-Hz-related bins. "50" waveforms leave 50, 100, and 150 Hz open but may have signals in 60-Hz-related bins. "60" waveforms leave 60, 120, and 180 open and may have signals in 50-Hz-related bins.

Generator and analyzer resolution increase directly with the length of the waveform. Thus, it was possible to construct signals at the 8k and 4k record length which allow universal hum measure-

1k Waveform Family, No Hum Measurement. Burst Length to Acquire Approximately 130 Milliseconds					
Gen. Record Length 1024 bytes. Resolution 46.9 Hz. Analyzer Record Length 2048 Bytes. Res. 23.4 Hz					
1K_FSN_.WAV		1K_BSN_.WAV		1K_FMN_.WAV	1K_BMN_.WAV
Sinewave Amplitudes -21.6 dBFS		Sinewave Amplitudes -20.7 dBFS		-19.9 dBFS	-19.1 dBFS
Frequencies Common to Both Channels (Hz)	Unique Left and Right Frequencies (Hz)	Frequencies Common to Both Channels (Hz)	Unique Left and Right Frequencies (Hz)	Frequencies (Hz)	Frequencies (Hz)
47		47		47	47
94		94		94	94
141		141		141	141
188		188		188	188
234		234		234	234
516		516		516	516
984		984		984	984
1406		1406		1406	1406
2016		2016		2016	2016
2766	3141L	2766	3141L	2766	2766
3984	3609R	3984	3609R	3984	3984
5625	6422L	5625	6422L	5625	5625
8016	7219R	8016	7219R	8016	8016
11203	12469L	11203	12469L	11203	11203
15984	13734R	15000	13734R	15984	15000
19969				19969	

Table C-5 *Frequencies Included in 1024-Sample Waveforms with No Hum Measurement Capability*

ments. At the 512 and 256 sample record lengths, it was not possible to make any useful hum measurements while also preserving useful low frequency response. If hum measurement needs are more important than low frequency response measurements with these very short records, new waveforms could be constructed which do not include any signals below 281 Hz for 512-sample waveforms or below 375 Hz for 256-sample waveforms.

C.9.2.4. Harmonic Distortion

Wherever permitted by the available frequency resolution for each record length, sinewave fundamental frequencies were located so that they would not prevent measurement of second or third harmonic distortion of lower-frequency sinewaves in the same record. The 8k waveforms are totally free of this problem. The 4K_F (full 20-20k bandwidth) waveforms have a 47 Hz signal which prevents measurement of the second harmonic of their 23 Hz signal, and a 70 Hz signal which prevents measurement of second harmonic of their 35 Hz signal or of the third harmonic of their 23 Hz signal. The 2K_F waveforms prevent measurement of third harmonic of 23 Hz, the 2K_x60 waveform interferes with measurement of the second harmonic of the 70 Hz signal, and all 2K_ waveforms prevent measurement of the third harmonic of the 70 Hz signal.

The limited resolution of the 1k, 512, and 256-sample waveforms prevents their being able to measure harmonic distortion of fundamental signals of 94 Hz and below (1k waveforms) or 188 Hz and below (512 and 256-sample waveforms).

1k Waveform Family, 50 Hz AC Mains. Burst Length to Acquire Approximately 130 Milliseconds					
Gen. Record Length 1024 bytes. Resolution 46.9 Hz. Analyzer Record Length 2048 Bytes. Res. 23.4 Hz					
1K_FS50.WAV		1K_BS50.WAV		1K_FM50.WAV	1K_BM50.WAV
Sinewave Amplitudes -19.8 dBFS		Sinewave Amplitudes -18.7 dBFS		-18.2 dBFS	-18.8 dBFS
Frequencies Common to Both Channels (Hz)	Unique Left and Right Frequencies (Hz)	Frequencies Common to Both Channels (Hz)	Unique Left and Right Frequencies (Hz)	Frequencies (Hz)	Frequencies (Hz)
188		188		188	188
234		234		234	234
516		516		516	516
984		984		984	984
1406		1406		1406	1406
2016		2016		2016	2016
2766	3141L	2766	3141L	2766	2766
3984	3609R	3984	3609R	3984	3984
5625	6422L	5625	6422L	5625	5625
8016	7219R	8016	7219R	8016	8016
11203	12469L	11203	12469L	11203	11203
15984	13734R	15000	13734R	15984	15000
19969				19969	

Table C-6 *Frequencies Included in 1024-Sample Waveforms for Hum Measurement for 50 Hz Power*

1k Waveform Family, 60 Hz AC Mains. Burst Length to Acquire Approximately 130 Milliseconds					
Gen. Record Length 1024 bytes. Resolution 46.9 Hz. Analyzer Record Length 2048 Bytes. Res. 23.4 Hz					
1K_FS60.WAV		1K_BS60.WAV		1K_FM60.WAV	1K_BM60.WAV
Sinewave Amplitudes -20.2 dBFS		Sinewave Amplitudes -20.3 dBFS		-19.4 dBFS	-18.5 dBFS
Frequencies Common to Both Channels (Hz)	Unique Left and Right Frequencies (Hz)	Frequencies Common to Both Channels (Hz)	Unique Left and Right Frequencies (Hz)	Frequencies (Hz)	Frequencies (Hz)
47		47		47	47
94		94		94	94
141		141		141	141
234		234		234	234
516		516		516	516
984		984		984	984
1406		1406		1406	1406
2016		2016		2016	2016
2766	3141L	2766	3141L	2766	2766
3984	3609R	3984	3609R	3984	3984
5625	6422L	5625	6422L	5625	5625
8016	7219R	8016	7219R	8016	8016
11203	12469L	11203	12469L	11203	11203
15984	13734R	15000	13734R	15984	15000
19969				19969	

Table C-7 *Frequencies Included in 1024-Sample Waveforms for Hum Measurement for 60 Hz Power*

512 Sample Waveform Family, No Hum or Stereo Separation Measurement. Burst Length to Acquire Approximately 70 Milliseconds	
Gen. Record Length 512 bytes. Resolution 93.8 Hz. Analyzer Record Length 1024 bytes. Resolution 46.9 Hz	
512FMN_.WAV	512BMN_.WAV
Sinewave Amplitudes -19.3 dBFS	Sinewave Amplitudes -18.9 dBFS
Frequencies (Hz)	Frequencies (Hz)
94	94
188	188
281	281
375	375
469	469
1031	1031
1594	1594
2812	2812
4031	4031
5531	5531
7969	7969
11250	11250
16031	15000
19969	

Table C-8 *Frequencies Included in 512-Sample Waveforms*

256 Sample Waveform Family, No Hum or Stereo Separation Measurement. Burst Length to Acquire Approximately 40 Milliseconds	
Gen. Record Length 256 bytes. Resolution 187.5 Hz. Analyzer Record Length 512 bytes. Resolution 93.8 Hz	
256FMN_.WAV	256BMN_.WAV
Sinewave Amplitudes -nn.n dBFS	Sinewave Amplitudes -nn.n dBFS
Frequencies (Hz)	Frequencies (Hz)
188	188
375	375
562	562
938	938
2062	2062
3188	3188
4312	4312
5625	5625
8062	8062
11062	11062
15938	15000
19875	

Table C-9 *Frequencies Included in 256-Sample Waveforms*

C.9.2.5. Introduction to Procedure Files, FASTTRIG

A large number of procedure files are furnished with FASTTRIG to make operation as easy as possible. These procedure files fall into several categories.

Setup procedure. TR-SETUP.PRO is an interactive procedure which prompts the user to enter values to customize the tests for the expected signal levels and to specify switcher channels if used. These values are saved and used as the starting point for all capture procedures.

Sweep-naming and waveform file-loading procedures (32 procedures, one for each .WAV file furnished). These procedures correspond exactly to the waveform file names; for example, 2K_FS50.PRO corresponds to 2K_FS50.WAV. Loading and running one of these procedures causes the proper sweep file names to be attached to all test and overlay files which may be later used by any of the signal capture procedures.

Capture procedures (five procedures). Each is designed for a specific class of equipment or system under test, to do a specific suite of tests. The five procedures are for a monaural device and for four variations of tests on stereo devices.

Burst generation procedures. Three procedures are furnished to generate timed bursts at the digital and analog outputs of System One.

C.9.2.6. Test and Overlay Files for Capture Procedures

The following tests and overlays are used by one or more of the five "capture" procedures to acquire and analyze data:

DIST-MON.OVL Total distortion and noise, monaural devices

DIST-STE.OVL Total distortion and noise, stereo devices

HUM-MON.OVL Hum, monaural

HUM-STE.OVL Hum, stereo

NOIS-MON.OVL "Empty bin" noise, monaural

NOIS-STE.OVL "Empty bin" noise, stereo

PHASE.OVL Interchannel phase

PHASGAIN.OVL Interchannel phase and differential gain

SEP-L-R.OVL Stereo separation left-to-right

SEP-R-L.OVL Stereo separation right-to-left

TRG-MON.TST Signal acquisition and frequency response analysis, monaural

TRG-RESP.TST Signal acquisition and frequency response analysis, stereo

```
PROCEDUREv2.10
LOAD TEST trg-resp/R
UTIL PROMPT/r/r/r/r
        On the following screen, enter the value corresponding to/R
        the maximum expected input signal amplitude into the analyzer/R
        CHANNEL A and CHANNEL B INPUT RANGE fields and the dBr REF field./R
        Press <Enter> to make the entry into each field. Note that/R
        the input range fields will display the range maximum value/R
        rather than the exact value which you enter./E
PANEL la/F10/R
/F10/R
lr/F10/R
/E
UTIL PROMPT/r/r/r/r
        On following screen, enter the input switcher (SWR-122F) channel/R
        numbers where signal is to be monitored into switcher CHANNEL A/R
        INPUT and CHANNEL B INPUT. If you are not using switchers,/R
        simply press <Enter> on each field./E
PANEL si/F10/R
/F10/R
/E
SAVE TEST /R
yPANEL s2/R
/E
SAVE TEST trg-mon/R
UTIL END
```

Figure C-21 Listing, TR-SETUP.PRO to Set Input Ranges, dBr Reference, Switcher Channels

Each of the five procedures starts with either TRG-MON.TST or TRG-RESP.TST and uses the <F9> keystroke to acquire a signal when the incoming signal matches the stored waveform and provide a frequency response measurement.

All of the remaining processing is done by loading an overlay (.OVL) file which has the appropriate FASTTRIG function selected (NORMAL, NOISE, DISTORT, DIFF, etc.) and appropriate settings at DATA-1 and DATA-2. The CHANNEL A INPUT range field, CHANNEL B INPUT range field, analyzer dBr REF field, and the switcher CHANNEL A and CHANNEL B INPUT fields are all "punched out" of these overlays. Thus, the settings of those fields in TRG-MON.TST or TRG-RESP.TST "propogate" on through all overlays in the procedure.

C.9.2.7. Setup Procedure

Figure C-21 is a listing of TR-SETUP.PRO. The procedure loads TRG-RESP.TST (stereo signal acquisition test). This test is set up with the TRIG+0 acquisition mode and NORMAL processing following the FFT. The procedure prompts the user to enter the expected signal amplitude into both channel input range fields and the dBr REF field. Fixed input ranges are required to capture the short bursts used with FASTTRIG. Distortion and noise are measured in dBr units, normally with reference to the expected signal amplitude. Signal amplitude should be stated in terms of the rms value of a sine wave whose peak-to-peak value equals the multitione signal peak-to-peak value. In a broadcast system, for example, the value to be entered would normally be the level of a sine wave at 100% modulation.

The procedure then prompts the user to enter the switcher channels to be used. If no switcher is used, these entries are irrelevant and the user may simply press <Enter> on both fields.

At the end of this "customizing", the procedure saves the result under the original file name of TRG-RESP.TST. The next two lines of the procedure turn off the DATA-2 selection and save the otherwise-identical result as the monaural acquisition and response test TRG-MON.TST.

```
PROCEDUREv2.10        ;4K_BSU_.PRO
LOAD OVERLAY phase/R
NAMES SWEEP 4k_bmu_/R
SAVE OVERLAY/R
LOAD OVERLAY phasgain/R
NAMES SWEEP 4k_bmu_/R
SAVE OVERLAY/R
LOAD OVERLAY dist-ste/R
NAMES SWEEP 4k_bsu_/R
SAVE OVERLAY/R
LOAD OVERLAY dist-mon/R
NAMES SWEEP 4k_bsu_/R
SAVE OVERLAY/R
LOAD OVERLAY nois-ste/R
NAMES SWEEP 4k_bmu_/R
SAVE OVERLAY/R
LOAD OVERLAY nois-mon/R
NAMES SWEEP 4k_bmu_/R
SAVE OVERLAY/R
LOAD OVERLAY hum-ste/R
NAMES SWEEP u-hum/R
SAVE OVERLAY/R
LOAD OVERLAY hum-mon/R
NAMES SWEEP u-hum/R
SAVE OVERLAY/R
LOAD OVERLAY sep-l-r/R
NAMES SWEEP 4k_blu_/R
SAVE OVERLAY/R
LOAD OVERLAY sep-r-l/R
NAMES SWEEP 4k_bru_/R
SAVE OVERLAY/R
LOAD TEST trg-resp/R
NAMES SWEEP 4k_bmu_/R
SAVE TEST/R
LOAD TEST trg-mon/R
NAMES SWEEP 4k_bmu_/R
SAVE TEST/R
LOAD WAVEFORM 4k_bsu_/R
1g,2g/R
UTIL END
```

Figure C-22 Listing, Typical Sweep Naming Procedure

TR-SETUP.PRO should be run whenever signal is to be picked up at a different point or whenever the expected signal level is different from the previous value. It does not need to be run for changes in waveform or changes between monaural and stereo tests.

C.9.2.8. Sweep Naming Procedures

Figure C-22 is a listing of 4K_BSU_.PRO, typical of all 32 of the sweep naming procedures. This procedure, in sequence, loads each of the overlay and test files which are used by any of the capture procedures. It uses the Names Sweep command to "attach" the proper .SWP file to each, and re-saves it to disk. Each of these procedures thus automatically manages the relationship between .WAV and .SWP files as described in one row of Table C-10. Finally, each of these procedures downloads the .WAV file into the DSP unit as required for FASTTRIG to function.

Quitting from S1.EXE software, turning off the power to System One, or loading a test file with another DSP program attached or no DSP program attached will cause loss of the waveforms in the DSP unit. In such a case, one of these 32 sweep-naming procedures must be run before a capture procedure can be successfully run.

C.9.2.9. Capture Procedures

The five capture procedures furnished are as follows:

TR-MONO.PRO Measures frequency response, noise, hum, and distortion on monaural devices and systems. See Figure C-23 for a listing of this procedure.

TR-STER1.PRO Measures frequency response, interchannel phase, noise, hum, and distortion on stereo devices and systems. Does not measure stereo separation and does not plot differential gain between stereo channels. Similar to TR-MONO.PRO with the addition of an inter-channel phase test.

TR-STER2.PRO Measures frequency response, interchannel phase, noise, hum, distortion, and stereo separation on stereo devices and systems. Stereo separation is plotted as two separate graphs, one showing the amplitudes of both driven and non-driven channel at the unique Channel A frequencies

```
PROCEDUREv2.10              ;MONAURAL CAPTURE AND ANALYSIS PROCEDURE
LOAD TEST trg-mon/R
/F9/A6/E
SAVE TEST response/Ry
yLOAD OVERLAY dist-mon/R
/A6/E
PANEL ç/E                   ;restores punched-out fields so .OVL can save as .TST
SAVE TEST distort/R
yLOAD OVERLAY nois-mon/R
/A6/E
PANEL ç/E
SAVE TEST noise/R
LOAD OVERLAY hum-mon/R      ; REMOVE THESE FOUR LINES WHEN HUM
/A6/E                       ; MEASUREMENTS ARE NOT DESIRED, OR
PANEL ç/E                   ; NOT POSSIBLE DUE TO USE OF WAVEFORM
SAVE TEST hum/R             ; WITH "N_" AS 6TH & 7TH CHARACTERS.
UTIL END
```

Figure C-23 Listing, Capture Procedure TR-MONO.PRO

```
PROCEDUREv2.10              ; STEREO CAPTURE AND ANALYSIS PROCEDURE WITH
LOAD TEST trg-resp/R        ; STEREO SEPARATION L-R & R-L PLOTTED ON SEPARATE
/F9/A6/E                    ; GRAPHS, NO GRAPH OF DIFFERENTIAL GAIN
SAVE TEST response/Ry
LOAD OVERLAY phase/R
/A6/E
PANEL ç/E                   ;restores punched-out fields so .OVL can save as .TST
SAVE TEST phase/R
yLOAD OVERLAY dist-ste/R
/A6/E
PANEL ç/E
SAVE TEST distort/R
yLOAD OVERLAY nois-ste/R
/A6/E
PANEL ç/E
SAVE TEST noise/R
LOAD OVERLAY hum-ste/R      ;  REMOVE THESE FOUR LINES WHEN HUM
/A6/E                       ;  MEASUREMENTS ARE NOT DESIRED, OR
PANEL ç/E                   ;  NOT POSSIBLE DUE TO USE OF WAVEFORM
SAVE TEST hum/R             ;  WITH "N_" AS 6TH & 7TH CHARACTERS.
LOAD OVERLAY sep-l-r/R
/A6/E
PANEL ç/E
SAVE TEST sep-l-r/R
LOAD OVERLAY sep-r-l/R
/A6/E
PANEL ç/E
SAVE TEST sep-r-l/R
UTIL END
```

Figure C-24 Listing, Capture Procedure TR-STER2.PRO

and the other at the unique Channel B frequencies. A System One Dual Domain with external D/A converters or a digital tape or disc previously recorded from a System One Dual Domain is the required signal source for this procedure. See Figure C-24.

TR-STER3.PRO Measures frequency response, interchannel phase, noise, hum, and distortion on stereo devices and systems. Does not measure stereo separation. The differential gain between stereo channels is automatically derived and plotted on the same graph with interchannel phase (see Figure C-26). This processing involves the utility program COMBINE.EXE which operates during a DOS operation (shell). This requires more than the usual S1.EXE setaside of 64k of memory, so the /B and or /R command line options must be used when S1.EXE is loaded to provide the additional memory. The command

S1 /R100

provides 100k of memory setaside, which is sufficient for COMBINE to operate. If the procedure halts with the message that OUTFILE.DAT cannot be found, this is usually an indication that insufficient memory was set aside for COMBINE.EXE. This same processing for differential gain and phase is also performed in procedure TR-STER4.PRO, described and listed below.

```
PROCEDUREv2.10              ; STEREO CAPTURE AND ANALYSIS PROCEDURE WITH
DOS ERASE OUTFILE.DAT/R     ; STEREO SEPARATION L-R & R-L PLOTTED ON SAME
LOAD TEST trg-resp/R        ; GRAPH, DIFFERENTIAL GAIN PLOTTED WITH PHASE
/F9/A6/E
SAVE TEST response/Ry       ;saves absolute response of both stereo channels
NAMES DELTA response/R      ;establishes same file on disk as "delta" reference
COMPUTE DELTA 1,2/R         ;subtracts right channel on disk from left in memory
SAVE DATA INFILE2/Ry        ;resulting channel imbalance saved as ASCII data file
LOAD OVERLAY phasgain/R     ;interchannel phase as DATA-1, amplitude units in DATA-2
/A6/E                       ;obtains interchannel phase from acquired waveform
SAVE DATA INFILE1/RY        ;saves phase as ASCII data file
DOS COMBINE -3/R            ;combines phase into DATA-1 and chan. balance into DATA-2
LOAD DATA OUTFILE/R         ;loads combined data
PANEL ç/E                   ;restores punched-out fields so .OVL can save as .TST
SAVE TEST phasgain/R
yLOAD OVERLAY dist-ste/R
/A6/E
PANEL ç/E
SAVE TEST distort/R
yLOAD OVERLAY nois-ste/R
/A6/E
PANEL ç/E
SAVE TEST noise/R
LOAD OVERLAY hum-ste/R      ;  REMOVE THESE FOUR LINES WHEN HUM
/A6/E                       ;  MEASUREMENTS ARE NOT DESIRED, OR
PANEL ç/E                   ;  NOT POSSIBLE DUE TO USE OF WAVEFORM
SAVE TEST hum/R             ;  WITH "N_" AS 6TH & 7TH CHARACTERS.
LOAD OVERLAY sep-l-r/R      ; attached .SWP file has left channel unique freq's
/A6/E
PANEL ç/E
SAVE TEST sep-l-r/R         ; DATA-1 driven channel (left), DATA-2 right
NAMES DELTA sep-l-r/R       ; establishes same file on disk as reference
COMPUTE DELTA 2,1/R         ; subtracts left from right, results into DATA-2
COMPUTE EXCHANGE/E          ; exchanges DATA-2 into DATA-1
SAVE DATA INFILE1/Ry        ; saves result as ASCII data file
LOAD OVERLAY sep-r-l/R      ; attached .SWP file has right channel unique freq's
/A6/E
PANEL ç/E
SAVE TEST sep-r-l/R         ; DATA-2 driven (right), DATA-1 left
NAMES DELTA sep-r-l/R       ; establishes same file on disk as reference
COMPUTE DELTA 1,2/R         ; subtracts right from left, results into DATA-1
SAVE DATA INFILE2/Ry        ; saves result as ASCII data file
DOS COMBINE -3/R            ; combines results of separation L-R and R-L
LOAD DATA OUTFILE/R         ; loads ASCII data
SAVE TEST separati/R        ; saves result
UTIL END
```

Figure C-25 Listing, Capture Procedure TR-STER4.PRO

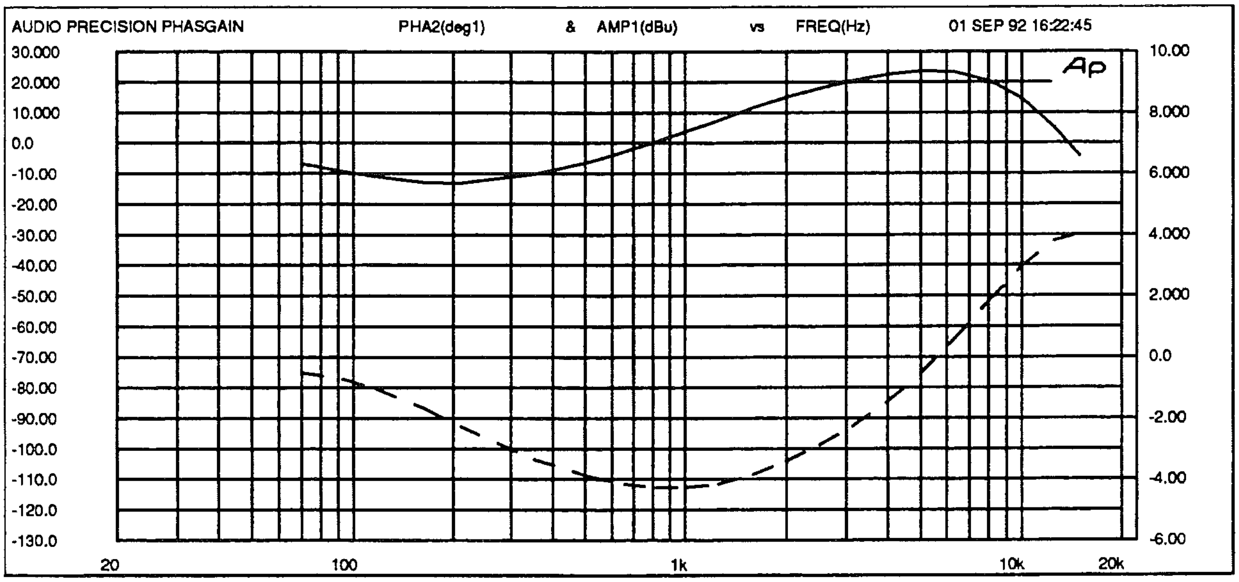

Figure C-26 Example of Differential Phase and Gain Plot from TR-STER3.PRO or TR-STER4.PRO

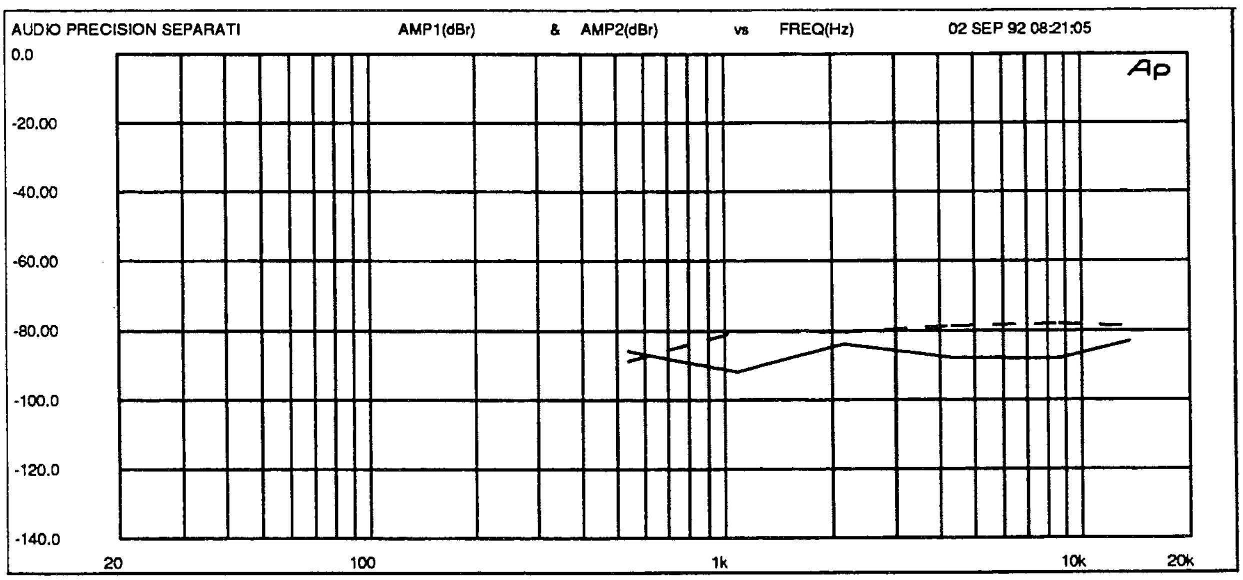

Figure C-27 Example of Stereo Separation L-R and R-L Combined in One Graph, from TR-STER4.PRO

TR-STER4.PRO Measures frequency response, interchannel phase, noise, hum, distortion, and stereo separation on stereo devices and systems. The differential gain between stereo channels is automatically derived and plotted on the same graph with interchannel phase. For convenience, stereo separation both left-to-right and right-to-left are plotted on the same graph, with the levels on the driven channels not shown (see Figure C-27). Note that combining both L-R and R-L stereo separation on one graph results in both being plotted at the frequencies of the unique left channel tones. The R-L separation measurements are actually made at unique right channel frequencies, which are typically about eight percent higher than the left channel tones. Both the differential gain processing and combined stereo separation graph involve the utility program COMBINE.EXE as described above, so sufficient memory must be set aside when S1.EXE is started. See the procedure listing in Figure C-25.

All five of these procedures save results into a common set of test names. The frequency response result is RESPONSE.TST. Distortion is stored into DISTORT.TST. Noise is saved into NOISE.TST. Hum measurements are stored to HUM.TST. Simple interchannel phase is saved into PHASE.TST; the combined differential phase and gain results from TR-STER3.PRO and TR-STER4.PRO are saved into PHASGAIN.TST. The combined stereo separation results of TR-STER4.PRO are saved into SEPARATI.TST. The two separate stereo separation measurements of TR-STER2.PRO are stored into SEP-L-R.TST and SEP-R-L.TST. Since the stored results names are common, previously-saved results files will be over-written when any of the procedures runs. Thus, if it is desired to make a permanent record of test results, the tests should be copied to other file names or moved to another directory before any of the procedures is run again.

All five capture procedures as furnished include hum measurement processing. Hum measurements are not effective with any waveform with "N" as the sixth character. "N" waveforms have a sinewave fundamental in the same FFT bin as one or more of the hum frequencies, which prevents measurement of any hum signals. If an "N" waveform is used, the capture procedure should be edited by deleting the four lines dealing with hum measurement, or placing a semi-colon (;) at the beginning of each line which causes them to be ignored during execution.

C.9.2.10. Burst Generation Procedures

Three procedures are furnished to generate a timed burst of multitone signal. These procedures can be used to record a burst onto digital tape for later use as the signal source for FASTTRIG acquisitions. The two procedures with "DIG" in their names, BURDIG-M.PRO and BURDIG-S.PRO, can only be used with the digital outputs of a System One Dual Domain. BURANA-M.PRO can be used with the analog generator outputs of any System One DSP model. The "M" or "S" as the last character of these burst generation procedures indicates that they were designed for waveforms with "M" or "S" as the fifth character of the file name. See Figure C-28 for a listing of one of these procedures.

The third line (LOAD WAVEFORM . . .) of any of these procedures must be edited to replace the "8K_BSU_.WAV" file as furnished with the file name of the desired waveform. The waveforms with "S" as their fifth character are two-channel waveforms and can only be transmitted in the digital domain from a System One Dual Domain. The waveforms with "M" as their fifth character are sin-

```
PROCEDUREv2.10              ;BURDIG-S.PRO, for stereo separation waveforms
LOAD TEST DIG-OUT/R         ;adjust DGEN AMPL field of DIG-OUT.TST for desired amplitude
LOAD WAVEFORM 512bmn_/R     ;replace with desired waveform
1G,2G/R                     ;loads dual waveform file into both generator buffers
UTIL DELAY 5/R              ;stabilization time for RDAT
PANEL DO/R                  ;turns DGEN digital outputs on
/E
UTIL DELAY .07/R            ;length of burst, in seconds
/E
PANEL /R                    ;turns DGEN digital outputs off
/E
UTIL END
```

Figure C-28 Listing, Burst Generation Procedure BURDIG-S.PRO

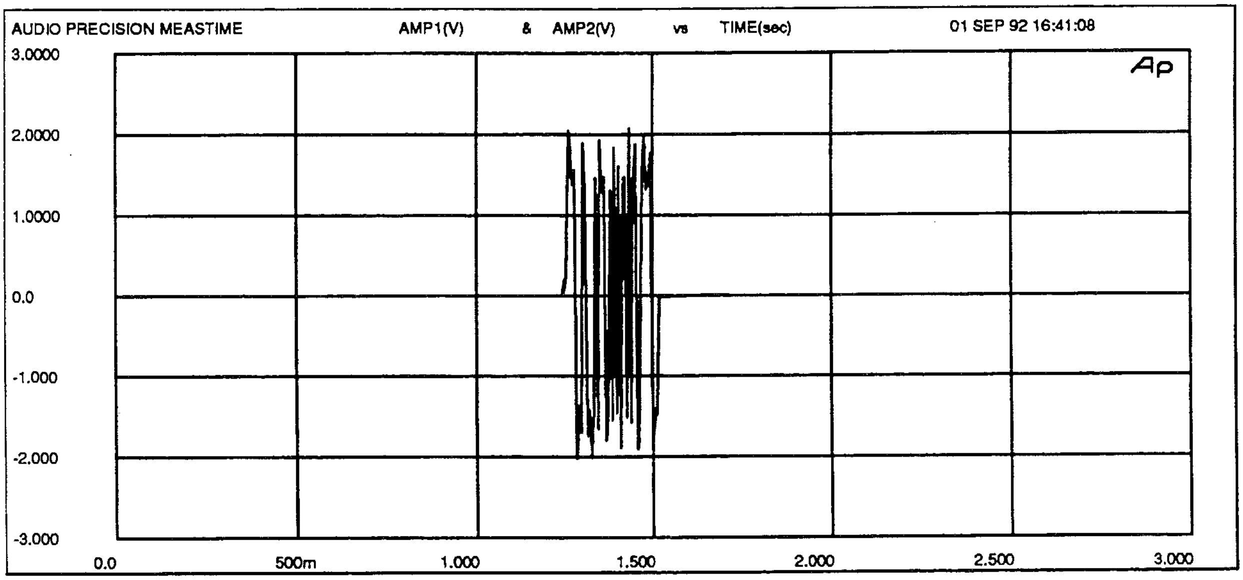

Figure C-29 Example of Burst Acquired and Displayed with MEASTIME.TST

gle channel waveforms which are loaded into both generator buffers and and can be used for monaural tests or for all stereo tests except stereo separation. The analog burst procedure BURANA-M.PRO can only be used with a single channel (M) waveform, since the analog multitone output capability of System One comprises only a single D/A converter and a single output amplifier and transformer, resistively split into two channels.

The number following "UTIL DELAY" determines the length of the burst, in seconds. The value "1" stored will produce a nominal one second burst; for other values, it must be edited to the desired value. With fast computers such as 486-based, the actual burst produced is typically 10-20 milliseconds longer than the value in the UTIL DELAY command. With slower computers, it may be necessary to experiment with the UTIL DELAY value until the desired burst length is produced.

The test file MEASTIME.TST is furnished to make measurements of burst length. This test produces a display of amplitude versus time (digital storage oscilloscope mode). MEASTIME.TST uses FFTSLIDE.DSP with the 8 kHz sample rate to produce a record length of 3.84 seconds (assuming full DSP memory), although as stored it displays only the first three seconds. It uses the Peak function to produce a representation of the burst envelope. After using one of the burst procedures to generate a burst and record it onto digital tape, MEASTIME.TST can be loaded and the <F9> key pressed one or two seconds before the burst is expected during reproduction of the tape. The resulting display (3.84 seconds later) should look similar to Figure C-29. The graphic cursors can be used to determine the start and stop time of the burst. Adjustments in the UTIL DELAY value of the burst procedure can then be made to compensate for delays in procedure execution of slower computers.

C.9.2.11. Sweep Files, FASTTRIG

Sweep files (.SWP) are furnished for all FASTTRIG waveforms. The sweep files all have seven character names except for the three sweep files used for hum measurements. The nomenclature system of the sweep files (other than the hum files) is as follows. The first three characters describe the length of the waveform they are used with, using the same system described above under waveforms.

The fourth character designates bandwidth, using F (full) or B (broadcast) as with waveforms.

Sweep Table Usage vs .WAV File and Measurement Made							
	.SWP Files to Be Used						
.WAV File	Response, Phase, and Noise Measurements	Total Distortion Measurements	Stereo Separation Left-Right	Stereo Separation Right-Left	Hum at 50 and 60 Hz Mains	Hum at 50 Hz Mains	Hum at 60 Hz Mains
8K_BMU_	8K_BMU_	8K_BMU_	N/A	N/A	U-HUM	U-HUM	U-HUM
8K_BSU_	8K_BMU_	8K_BSU_	8K_BLU_	8K_BRU_	U-HUM	U-HUM	U-HUM
8K_FMU_	8K_FMU_	8K_FMU_	N/A	N/A	U-HUM	U-HUM	U-HUM
8K_FSU_	8K_FMU_	8K_FSU_	8K_FLU_	8K_FRU_	U-HUM	U-HUM	U-HUM
4K_BMU_	4K_BMU_	4K_BMU_	N/A	N/A	U-HUM	U-HUM	U-HUM
4K_BSU_	4K_BMU_	4K_BSU_	4K_BLU_	4K_BRU_	U-HUM	U-HUM	U-HUM
4K_FMU_	4K_FMU_	4K_FMU_	N/A	N/A	U-HUM	U-HUM	U-HUM
4K_FSU_	4K_FMU_	4K_FMU_	4K_FLU_	4K_FRU_	U-HUM	U-HUM	U-HUM
2K_BM50	2K_BM50	2K_BM50	N/A	N/A	N/A	50-HUM	N/A
2K_BM60	2K_BM60	2K_BM60	N/A	N/A	N/A	N/A	60-HUM
2K_BS50	2K_BM50	2K_BS50	2K_BL50	2K_BR50	N/A	50-HUM	N/A
2K_BS60	2K_BM60	2K_BS60	2K_BL60	2K_BR60	N/A	N/A	60-HUM
2K_FM50	2K_FM50	2K_FM50	N/A	N/A	N/A	50-HUM	N/A
2K_FM60	2K_FM60	2K_FM60	N/A	N/A	N/A	N/A	60-HUM
2K_FS50	2K_FM50	2K_FS50	2K_FL50	2K_FR50	N/A	50-HUM	N/A
2K_FS60	2K_FM60	2K_FS60	2K_FL60	2K_FR60	N/A	N/A	60-HUM
1K_BM50	1K_BM50	1K_BM50	N/A	N/A	N/A	50-HUM	N/A
1K_BM60	1K_BM60	1K_BM60	N/A	N/A	N/A	N/A	60-HUM
1K_BMN_	1K_BMN_	1K_BMN_	N/A	N/A	N/A	N/A	N/A
1K_BS50	1K_BM50	1K_BS50	1K_BL50	1K_BR50	N/A	50-HUM	N/A
1K_BS60	1K_BM60	1K_BS60	1K_BL60	1K_BR60	N/A	N/A	60-HUM
1K_BSN_	1K_BMN_	1K_BSN_	1K_BLN_	1K_BRN_	N/A	N/A	N/A
1K_FM50	1K_FM50	1K_FM50	N/A	N/A	N/A	50-HUM	N/A
1K_FM60	1K_FM60	1K_FM60	N/A	N/A	N/A	N/A	60-HUM
1K_FMN_	1K_FMN_	1K_FMN_	N/A	N/A	N/A	N/A	N/A
1K_FS50	1K_FM50	1K_FS50	1K_FL50	1K_FR50	N/A	50-HUM	N/A
1K_FS60	1K_FM60	1K_FS60	1K_FL60	1K_FR60	N/A	N/A	60-HUM
1K_FSN_	1K_FMN_	1K_FSN_	1K_FLN_	1K_FRN_	N/A	N/A	N/A
512BMN_	512BMN_	512BMN_	N/A	N/A	N/A	N/A	N/A
512FMN_	512FMN_	512FMN_	N/A	N/A	N/A	N/A	N/A
256BMN_	256BMN_	256BMN_	N/A	N/A	N/A	N/A	N/A
256FMN_	256FMN_	256FMN_	N/A	N/A	N/A	N/A	N/A

Table C-10 *FASTTRIG .SWP Files Cross-Reference to .WAV File and Measurement Type*

The fifth character may be M (monaural), S (stereo), L (left), or R (right).

The "M" .SWP files contain the list of frequencies in all monaural waveforms and common to both channels of all "S" waveforms. Thus, "M" .SWP files are to be used with all frequency response tests (monaural or stereo), all stereo interchannel phase tests, all noise tests (monaural or stereo), and with distortion tests with "M" waveforms.

The "S" .SWP files contain all the frequencies in both left and right channels of "S" waveform files. They are intended for use only with distortion measurements with "S" waveforms.

The "L" .SWP files contain only the frequencies unique to the left channel of "S" waveforms. They are to be used with tests measuring left-to-right stereo separation (which require "S" waveforms). The "R" .SWP files contain only right-channel frequencies of "S" waveforms and are to be used with right-to-left stereo separation tests.

The hum files are simply named 50-HUM.SWP for measuring equipment powered by 50 Hz AC mains, 60-HUM.SWP for 60 Hz power lines, and U-HUM.SWP for use with either 50 or 60 Hz mains.

Table C-10 shows which sweep files are to be used with which waveform files for the various types of measurements which FASTTRIG is capable of. The 32 "sweep naming" procedures furnished automate the assignment of the correct sweep files to all relevant tests and overlays for each furnished waveform.

D. Maximum Length Sequence Analysis Program MLS.DSP

D.1. Introduction

MLS.DSP is a program for the Audio Precision[tm] System One + DSP[tm] or the System One Dual Domain[tm]. MLS.DSP uses Maximum Length Sequence (MLS) testing to characterize the linear response of acoustical and electronic devices. It permits time-selective measurements in which one signal, such as the direct sound from a loudspeaker, may be separated from another similar signal, such as a room reflection. The time window may be adjusted to allow measurement of any arrival in a complex reverberation pattern. These signals may be examined in the time domain (showing "energy" as a function of time) or in the frequency domain (amplitude and phase vs frequency). Impulse responses may be saved to disk for later down-load to the DSP and further analysis.

This program generates a special digital noise signal called a Maximum Length Sequence (MLS) in the DSP module. For typical loudspeaker measurement applications this digital signal is converted to the analog domain in the D/A converter and fed through the analog generator output stage. The generator typically connects to a power amplifier which drives the loudspeaker under test. A measurement microphone is used to pick up the acoustical signal and return it to the System One analog input. System One performs balanced-to-unbalanced conversion, automatic gain ranging, and drives the DSP's A/D converters. The DSP module and MLS software then perform a cross-correlation between the received and transmitted signals to obtain the impulse response which is stored into DSP memory. The impulse response may be displayed on the computer, permitting the user to select the portion of impulse response of interest. This portion may be transformed into the frequency domain to study both magnitude and phase response versus frequency. As with any other System One test data, limits may be applied for go/no-go testing and tests may be incorporated into procedures to totally automate loudspeaker testing.

> The expression "loading" a DSP program is used frequently throughout this manual. In fact, NAMES PROGRAM is the specific command required to download a .DSP program from computer disk to DSP unit. When a .TST file is saved to disk after using the NAMES PROGRAM command, the DSP program will also be automatically downloaded each time the .TST file is loaded thereafter (unless the DSP program is already in place from the previous test).

These properties are of obvious use when measuring loudspeakers or other electroacoustic devices. The time-selective capability permits separating the device-under-test response from that of the room in which the measurements are made. Alternately the room itself may be measured, studying the reflection characteristics of each surface in the room or of the room taken as a whole. Electronic effects devices such as digital delays and reverberators may be measured for both frequency and time domain characteristics. The noise rejection properties of MLS analysis may be valuable when dealing with noisy electronic devices such as communications lines or two-way radio equipment. The analysis technique depends on repeatable time domain behavior in the device under test and is therefore not suited to measuring tape recorders, pitch shifters, time compressors or other devices with wow and flutter or speed error.

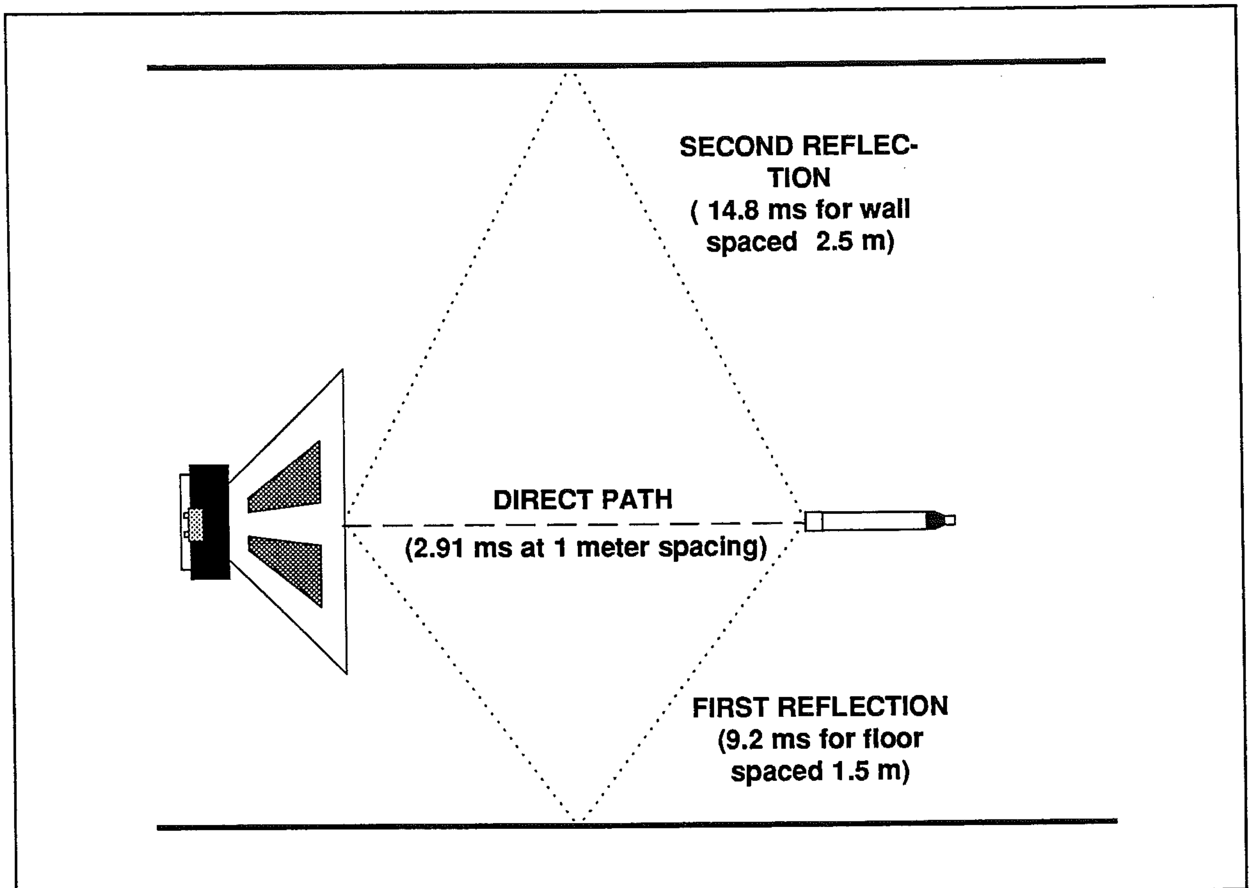

Figure D-1 Reflection Examples in a Typical Room

D.1.1. Quasi-Anechoic Measurements and Low Frequency Limitations

Other techniques exist which provide measurement capability similar to maximum length sequence analysis. These include Time Delay Spectrometry (TDS) and impulse testing. They all provide quasi-anechoic frequency response measurements of the loudspeaker alone, unaffected by room reflections. All share the limitation that this anechoic response is useful only above a critical frequency determined by the physical dimensions of the test environment. In the illustration of Figure D-1, for example, the direct signal arrives 2.91 milliseconds after it leaves the loudspeaker and the first reflection arrives 9.2 milliseconds after leaving the speaker. An anechoic measurement, by any of these techniques, must ignore reflections and therefore can only look at about 6.3 milliseconds of pure first-arrival signal before the first reflection. Meaningful amplitude measurements cannot be made without acquiring at least one full cycle of signal, and accurate measurements require several cycles. For the 6.3 millisecond example, the frequency corresponding to this time span is about 160 Hz. If useful frequency response is desired down to 35 Hz, for example, then the portion of signal to be converted must be a minimum of 1/35 second (28 milliseconds) and ideally closer to 1/10 second (100 milliseconds). In the example of Figure D-1 and most realistic actual rooms, this longer time necessarily includes several reflections and the response is no longer anechoic. True anechoic response with even moderate accuracy down to 35 Hz requires the first reflection path to be at least 28 milliseconds longer than the direct signal path. This requires a room with the nearest reflecting surface at least 5 meters away (about 16 feet). This calls for a room with a 10-meter (32 foot) high ceiling with the speaker and microphone on 5-meter (16 foot) tall stands!

D.1.2. MLS Advantages

The maximum-length-sequence (MLS) test signal is a psuedo-random noise with special properties which allow analysis of response without the variability commonly associated with noise-based measurements. The analysis technique provides a high degree of immunity from interfering noise, allowing accurate measurements when the interference actually exceeds the test signal level. The MLS signal more closely resembles program material than does a sinewave and will therefore measure under conditions approximating normal use.

The MLS technique has several significant advantages over either basic impulse testing or the TDS technique. The principal advantage of MLS over impulse testing is in signal-to-noise ratio improvement. The 32k point sequence used in MLS.DSP is equivalent to averaging 32,768 individual impulses of the same amplitude, and thus produces a signal-to-noise advantage of about 45 dB.

Compared to the TDS technique, MLS.DSP has a number of advantages. To obtain low-frequency response data, TDS must sweep very slowly. MLS has the same testing time whether evaluating only the anechoic portion (first arrival) or a longer portion of the signal for accurate bass response. With MLS, one acquisition and correlation produces an impulse response which may then be evaluated over and over to look at anechoic response, response of any selected reflection, integrated room response, etc. The TDS technique requires that the generator-to-bandpass filter frequency offset (delay time) be reset to equal the acoustical propagation delay for each acoustical path to be measured. Sweep speed is critical for the TDS technique, and an operator may easily select a speed which produces erroneous data without knowing he is making an error.

Ambient acoustic noise tends to be greatest at low frequencies due to heating and air conditioning systems, traffic noise, etc. To improve signal-to-noise ratios under these typical conditions, MLS.DSP is normally used with pink spectral shaping (high-frequency attenuation) of the generated pseudo-random noise. A complementary filter in the analysis process produces overall flat response. To accomplish the equivalent function in the TDS technique would require logarithmic sweeps and a variable-bandwidth filter, which is more complex and not done in available TDS instruments. Additionally, pink noise is similar in spectral distribution to voice and music, so the heating effect on individual loudspeakers of a multi-way loudspeaker system with pink MLS noise is similar to that which occurs during normal operation. The TDS technique (or any swept-sine method) concentrates all the input energy into a single loudspeaker driver at any single moment during the sweep, which can change driver characteristics during the test.

D.1.3. MLS Overview

MLS.DSP performs typical loudspeaker production tests very rapidly. Stimulus generation, acquisition, cross-correlation, and fast Fourier transform operations are all performed within the DSP module of System One + DSP or System One Dual Domain and are thus independent of computer speed. On the order of two seconds total is typically required for these functions. Transmission time of the data to the computer for graphing and/or limits comparison is computer-speed-dependent, and typically takes one to two seconds with a 386-based computer. Unlike the TDS technique, where the sweep speed must be reduced to obtain accurate low-frequency response data, the operating speed of MLS.DSP is identical whether evaluating a short section of the record for anechoic response or longer portions for integrated room response. The signal need not be re-acquired or re-correlated in order to evaluate response under both anechoic and wide-bandwidth conditions. The impulse response remains stored in DSP memory until a new acquisition is made. Thus, FFT transforms may be rapidly, sequentially made of both the anechoic portion and a longer portion for full-range response in about four to five seconds total time.

```
DSP HELP SCREEN
Psuedo random noise is generated and used to measure the impulse response
in either the time or frequency domains.  DSP panel and bargraph readings
display the real-time peak input for setting levels.  To see the impulse
response in the time domain select DSP TIME for SOURCE 1 on the sweep panel.
The impulse response may be transformed with an FFT to see the frequency
response by graphing DSP AMPL vs DSP FREQ.

The time record to be transformed is the time segment displayed on the last
DSP TIME sweep.  THIS MEANS YOU MUST PLOT THE IMPULSE RESPONSE OR THE
ENERGY TIME CURVE BEFORE YOU CAN FFT IT.  The panel REF TIME value adjusts
start of the transformed record to compensate for the time delay between
source and measurement device.

The F9 function key causes a new data acquisition and transform.  The F6 key
causes previously acquired data to be transformed.  The maximum acquired record
is 8191 points with standard memory, 32767 points with full memory.

CAUTION: DSP Panel readings must use %FS or dBFS.  F7 does not allow units to
be changed, use F6.  DATA1 and DATA2 must not be the same (AMPL-1 or AMPL-2).
```

Figure D-2 DSP HELP Panel with MLS.DSP Loaded

A common manufacturing defect in multi-way loudspeaker systems is for one of the drivers to be connected with the opposite polarity to that which the designer intended, creating response dips due to cancellation near the crossover frequencies. It is also common for the external system terminals to have their polarity reversed, producing severe stereo imaging problems. These defects may be detected by applying upper and lower limits to the phase-vs-frequency data.

MLS.DSP offers a selection of four different Maximum Length Sequences, each 32k long. This feature permits up to four different System One-equipped test stations to be located near one another on the production test floor with no interference effects between them.

See Figure D-2 for an illustration of the DSP HELP screen when MLS.DSP is loaded.

D.2. Panel Field Description

Figure D-3 shows the DSP panel with the MLS.DSP program loaded. Each of the fields has been highlighted to show the information and choices which appear at the bottom of the screen when the cursor lands on that field.

D.2.1. Display Fields

In panel mode the first two fields, labeled AMPL-1 and AMPL-2, are real-time displays of channel 1 and 2 input amplitudes respectively. These displays indicate one-half the peak-to-peak value, making them insensitive to dc voltages and improving readings of low level signals. Their principal purpose is to help set input levels and verify the presence of signal before making measurements. When signal is acquired via the analog analyzer and autoranging is used in the analog analyzer, overload will not normally be a problem. When displaying these signals the data will be scaled according to the ranges in use at the time of the last <F9> operation if analog display units are used. If the signal amplitude or

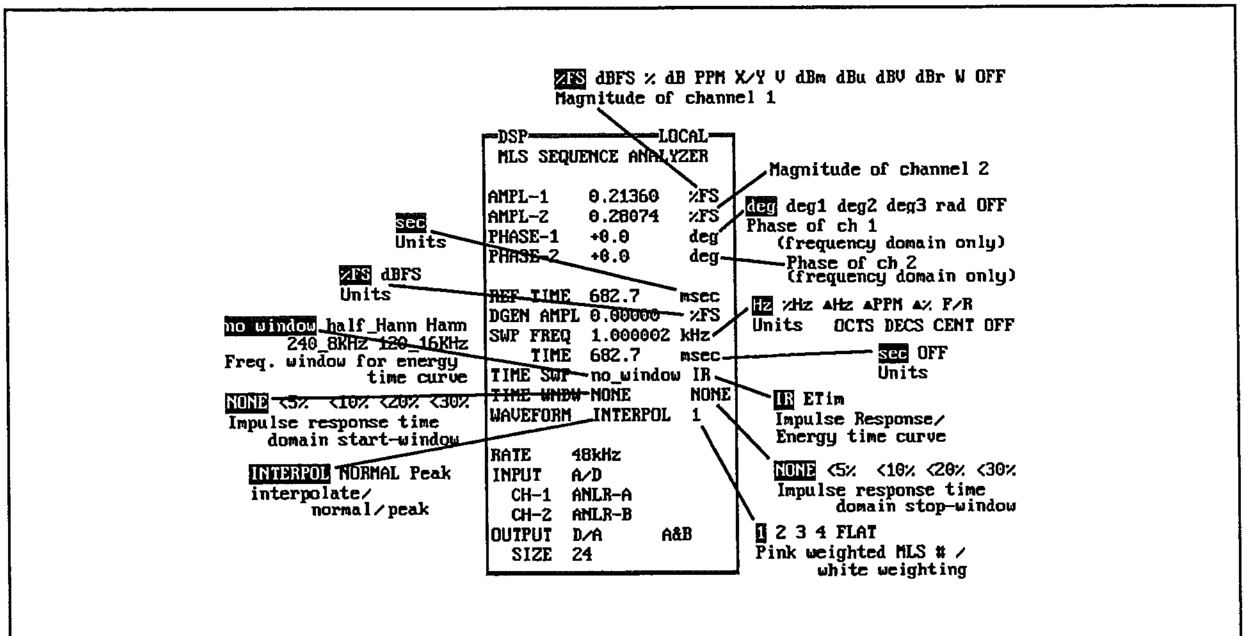

Figure D-3 DSP Panel Fields with MLS.DSP Loaded

the signal itself changes, these real-time displays will be in error. Therefore, only dBFS or %FS units should be used on the FFT real-time displays to avoid erroneous readings and possible overload. When analog signals are being acquired via the DSP BNC connectors, analog units may be used on the FFT panel since no ranging is involved.

When selecting data for display in a sweep, the AMPL-1 and AMPL-2 values are plotted as the time domain or frequency domain signal amplitude, as appropriate. The units used for a sweep may be selected independently of those used on the panel. Since the input ranging circuits are re-locked at the start of each acquisition, there is no problem with using any of the available units on a graph.

The second pair of fields, labeled PHASE-1 and PHASE-2, are not functional real-time readings on the panel. The phase parameter may be graphed during sweeps versus frequency. These fields are present on the panel only so that the SWEEP (F9) DEFINITIONS panel DATA-1 and DATA-2 fields have readings to access. The graph units may be any one of four System One degree units, (deg, deg1, deg2, deg3). The standard degree unit "deg" produces unwrapped phase plots which can accumulate many thousands of degrees of phase shift. The other units restrict the graphed values to a range of +/-180, -90/+270 and 0/360 degrees respectively. These units allow the phase data to fit comfortably on screen even before the initial delay time between speaker and microphone is not known. They are also useful when the expected range of data is not known and the graph limits would be inconvenient to set.

D.2.2. Reference Time for Phase Displays

The REF TIME field is used to tell the DSP the distance from the speaker under test to the measurement microphone as a reference for the phase measurements. This information allows the DSP to subtract out the transit time delay from the phase readings. As the REF TIME is adjusted the phase response will slope up or down reflecting the constant time delay component of the data.

D.2.3. Generator Amplitude

DGEN AMPL is used to set the MLS generator output amplitude when it is not routed through the analog generator outputs. This includes the digital outputs and the BNC connector D/A output. When the signal is routed through the analog generator this field is over-ridden. All level adjustment is then performed with the analog generator AMPLITUDE setting, while channel selection is performed with the analog generator OUTPUT settings.

The generator operates continuously and normally during panel mode and signal acquisition. It is, however, shut down during correlation computation, FFT computation, SAVE WAVEFORM and LOAD WAVEFORM operations. The dither feature described in the first chapter is operational with MLS but since MLS does not measure distortion it will have no detectable effect.

The SWP FREQ and SWP TIME fields are not used on the panel but are provided for the SWEEP (F9) DEFINITIONS panel to refer to during sweeps.

D.2.4. Time Domain Display Fields

The TIME SWP fields control the operation of time domain displays. The field on the right selects between impulse response (IR) displays and energy-time curve (ETim) displays. The IR selection will show the results of the MLS correlation which is the actual impulse response of the device under test. The ETim selection causes the DSP to transform the impulse response into the frequency domain, apply a frequency-domain window and Hilbert Transform to the complex frequency response, and then inverse transform the data back to the time domain to produce a plot of the estimated arrival of energy versus time. This "energy-time curve" is an approximation to the actual energy arriving at the microphone, since this energy can never be known without the simultaneous measurement of the velocity and pressure of the sound. Since the Energy-Time curve is computed from the pressure response alone it can never be complete.

D.2.5. Frequency Windows for Energy-Time Transforms

The energy-time (ETim) curves are obtained by an automatic sequence of operations which includes transforming the impulse response to the frequency domain, then reverse-transforming the frequency data back to the time domain. The left TIME SWP field selects the window function applied to the frequency-domain data when computing energy-time curves. The available window choices are no_window, Half_Hann, Hann, 240_8kHz, and 120_16kHz. The Hann window, although used on other measurement equipment, yields inaccurate results because it excludes behavior of the majority of most device's frequency range. The Half_Hann, 240_8kHz, and 120_16kHz are preferable for most applications. See the Frequency Windows section under Advanced Theory near the end of this chapter for more information on the effects of these selections and for illustrations of the window shapes.

D.2.6. Time Windows for Time-to-Frequency Transforms

When a section of the impulse response (direct arrival signal before reflections, for example) is isolated and transformed into the frequency domain, the impulse amplitude at the beginning and ending of that section will generally not be exactly the same and thus will not "splice" smoothly. The sharp edges introduced into the impulse response by splicing unequal amplitudes will produce ripples in the resulting frequency response plot. Windowing the time domain data by attenuating the amplitude at the beginning and end of the section to be transformed will reduce this rippling, but also reduces the steepness of transitions in the frequency response plots. The two TIME WNDW fields select the window applied to the impulse response (time domain) when transforming it to the frequency domain.

The available time windows are made up by separately selecting a "half-window" beginning at the selected START time (left-most field) and a second "half-window" ending at the selected STOP time (right-most field). Separate selection of the START "half-window" and STOP "half-window" permits creation of asymmetrical windows, which provide the optimum match to the asymmetrical shape of the typical impulse response. The available selections at both START and STOP are a family of half-cycle raised cosine functions labeled NONE, <5%, <10%, <20% and <30%. The numeric value refers to the amount of the data record (time span multiplied by sample period) taken up by the window's transition from zero to full amplitude. The START "half-window" starts with an amplitude of zero at the specified START time and climbs to an amplitude of 1.00 (no attenuation) at or before the selected percentage of the record. The STOP "half-window" starts with an amplitude of 1.00 at or following a point during the record which is within the selected percentage of the record end, and falls to zero at the specified STOP time. The windows with a steeper transition will alter the data less but will also have less impact on the frequency response ripples. The more gradual transitions have greater ripple reduction but alter the data more. See the Time Windows section under Advanced Theory near the end of this chapter for more details and illustrations of the available choices.

D.2.7. Display Mode Selection Field

The left WAVEFORM field selects the display mode for all graphing of results in time or frequency domain. ***This contrasts with FFTGEN.DSP and FFTSLIDE.DSP, where the similar field affects only time-domain displays.*** If INTERPOL is selected the DSP will compute the data value corresponding to the exact time or frequency value specified by the S1.EXE software during a sweep. If the specified time or frequency value exists the DSP will return the corresponding measurement value from memory. If the value does not exist it will be interpolated from the nearby measured values. If NORMAL is selected the DSP will return the closest actual measured value and will never alter the data. The Peak mode will return the largest value between the last requested sweep point and the current one. If the vertical axis of the graph is set to LIN on the SWEEP (F9) DEFINITIONS panel the largest data point will be returned without regard to its sign. If LOG (or LIN with a dB unit) is selected the absolute value of the data will be returned, allowing proper log computations in the computer.

Peak is recommended for time domain MLS displays (IR and ETim). NORMAL is recommended for most frequency domain displays. When LOG frequency domain displays are made down to frequencies of 100 Hz or below, INTERPOL will produce a more attractive display at low frequencies since it smooths through the discrete bin amplitudes.

D.2.8. Sequence Selection Field

The right WAVEFORM field selects the MLS sequence to be used for the signal generation and measurement. There are four different 32k point maximum length sequences provided, numbered 1 through 4. Each will cross-correlate to approximately -45 dB against any of the other three. This feature permits up to four different System One-equipped test stations to be located near one another on the production test floor with no significant interference effects between them. These four sequences have been weighted with a pink noise filter to increase their low frequency energy and provide a constant power per octave across the audio band. This greatly improves the signal-to-noise ratio at low frequencies, increasing measurement accuracy in typical room ambient noise conditions. A single white noise sequence is provided, labeled FLAT, for unusual applications where the large high frequency energy level may be desired and signal-to-noise ratio is not a concern. If the sequence is recorded on RDAT or other digital tape for later measurement, it is important that the same sequence number be selected on playback. Otherwise no impulse response will be obtained.

D.3. Making Measurements

D.3.1. Basic Panel Setup

A number of .TST files are furnished with appropriate settings for linear or log display of impulses, energy-time curves, and frequency response with and without phase response. These are described in the "Furnished Files" section at the end of this chapter. If you wish to set up test files from the basic panels, follow these main steps:

1. Select the appropriate signal source (DC-coupled BNC connectors, balanced and autoranged analyzer inputs, analyzer READING path, etc.). Most commonly the measurement microphone or preamplifier output will be connected to System One's CHANNEL A INPUT, AMPLITUDE function selected on the analog ANALYZER panel, and A/D RDNG selected as INPUT on the MLS.DSP panel. With the analyzer in AMPLITUDE mode the READING selection provides enough gain for most microphones without additional preamplifers.

2. Select the appropriate output path for the MLS signal (DC-coupled BNC connector, analog generator outputs via DGEN selection on generator panel, digital outputs, etc.). Most commonly the DGEN route will be chosen, with the MLS noise signal thus available at the analog generator outputs with selectable source impedance, balanced/unbalanced selections, and wide-range controllable amplitude.

3. Display the time domain response over the period of interest. For impulse response display, select DSP TIME as SOURCE-1 on the SWEEP (F9) DEFINITIONS panel. Typically, in an investigative mode, start and stop time values such as 0 and 30 milliseconds will display most of the area of interest. It is necessary to display time domain (IR or ETim) response at least once after starting S1.EXE software and using MLS.DSP, even if only the frequency domain behavior is desired. This step tells the DSP which portion (time span) of the impulse response is desired for subsequent frequency domain analysis. Select DSP AMPL-1 or AMPL-2 as DATA-1 if only one channel is to be displayed. If two channels are desired, select one DSP channel for DATA-1 (usually AMPL-1) and the other for DATA-2. Other measurement modules such as the DCX or the analog ANALYZER may not be used for DATA-1 or DATA-2 along with an MLS measurement. For a linear bipolar vertical display, select WAVEFORM NORMAL, Volts as the DATA units, and appropriate graph top and bottom values (equal but opposite in sign) for the expected peak-to-peak signal amplitude. For much better display of the later, lower-amplitude portions of the impulse response, use WAVEFORM Peak and a LOG display or LIN with dB units.

4. Set an appropriate # STEPS value for display of the impulse. This parameter determines how many points on the impulse response will be sent by the DSP module to the computer for display. The typical S1.EXE default value of 30 steps is inadequate for most applications, unless the impulse response itself is not of interest. Values of 200 to 500 points are the typical range for display clarity, with graphing speed favoring the smaller numbers. See below for more information on the display modes.

5. Press <F9> to acquire, correlate, and display the impulse response. The messages "Triggered and Acquiring" and "Transforming" should be seen at the top of the graph until the data is transmitted to the computer and displayed. If a frequency response display of only a selected portion of the impulse response is ultimately desired (such as the direct sound or a specific reflection), that desired time span should be the only portion of the impulse response visible on the screen. If this is not so, return to the SWEEP (F9) DEFINITIONS panel and enter new START and STOP values appropriately. Then press <F6> to re-display the data with the new time selection or press <F9> to acquire new data with the new time selection. The section of the impulse response which is to be transformed into the frequency domain is selected by the last set of START and STOP time values used in a time domain display sweep. Before selecting a frequency domain display it is essential that a time domain display must first have been selected and plotted. The specific portion of the record bounded by those last START and STOP time values will then be subse-

quently used for all future frequency domain displays unless a different .DSP program is used or operation of S1.EXE is terminated.

6. For frequency response display, select DSP FREQ as SOURCE-1. Select SOURCE-1 START and STOP frequencies for the range to be displayed, not exceeding the Nyquist frequency (one-half the sampling rate) for the upper frequency limit. Select a combination of DSP AMPL-1, DSP AMPL-2, DSP PHASE-1, DSP PHASE-2 and NONE as DATA-1 and DATA-2. Select DATA units such as Volts with a LOG display or dBV. Other measurement modules such as the DCX or the analog ANALYZER may not be used for DATA-1 or DATA-2 along with an MLS measurement. Set an appropriate # STEPS value for display of the response. This parameter determines how many points of the response will be sent by the DSP module to the computer for display. The typical S1.EXE default value of 30 steps is rarely adequate for loudspeaker frequency response graphs. Values of 100 to 300 points are the typical range for display clarity, with graphing speed favoring the smaller numbers. See below for more information on the display modes.

7. In general, you need not acquire again (<F9>) in order to change from time domain to frequency domain display or make other changes in the display of information. The acquired data remains in DSP memory and will be re-transformed according to the most recent DSP and SWEEP (F9) DEFINITIONS panel settings each time the <F6> key is pressed. Several .TST files can be set up with the various desired time and frequency display parameters, all with the MLS program named. The user may then select among these several tests and press <F6> to cause a new display of the data still in the DSP module memory according to the current test panel settings. Each time a frequency display is selected, the response will correspond to the section of impulse response bounded by the START and STOP times used the last time that a time domain display was loaded and <F6> or <F9> pressed.

The following sections provide more detail on operating the MLS program.

D.3.2. Display Modes

The WAVEFORM field (INTERPOL, NORMAL, Peak) affects both waveform display and spectral displays. This contrasts with FFTSLIDE.DSP and FFTGEN.DSP where the similar field only affects waveforms (time domain).

D.3.2.1. Waveform Display

When waveform interpolation is in the NORMAL condition, S1.EXE software follows its normal practice of drawing straight-line vectors between adjacent data points. This may produce adequately faithful reproduction of impulse responses when the signal frequency being displayed is low relative to the sample rate. Each signal peak is then represented by many samples and the display will be relatively smooth. When the impulse variation is fast relative to the sample rate, however, each cycle is represented by only a small number of points. When S1.EXE software "connects the dots", the result will be ragged and not representative of the true response.

When INTERPOL is selected instead of NORMAL, an additional software routine in the DSP unit computes what the signal waveform must have looked like to produce those samples. It assumes that the signal had been band-limited by an anti-alias (low-pass) filter before sampling. Given a sufficiently large # STEPS, the waveform will be faithfully reproduced.

The Peak choice on the WAVEFORM line selects a DSP routine which sends the largest positive or negative value since the previously-plotted point to the computer, preserving the sign. The purpose of Peak display mode is to avoid missing signal peaks due to an unfortunate combination of signal characteristics, sample rate, and the displayed # STEPS and SOURCE-1 START and STOP times (graphic aliasing). When used with the LOG selection or a decibel unit for DATA-1 or DATA-2, the Peak selection on the WAVEFORM line returns the absolute value of the largest peak value since the previous plotted point. This permits dB units to be used for the vertical display which is helpful in viewing low level reflections in impulse responses.

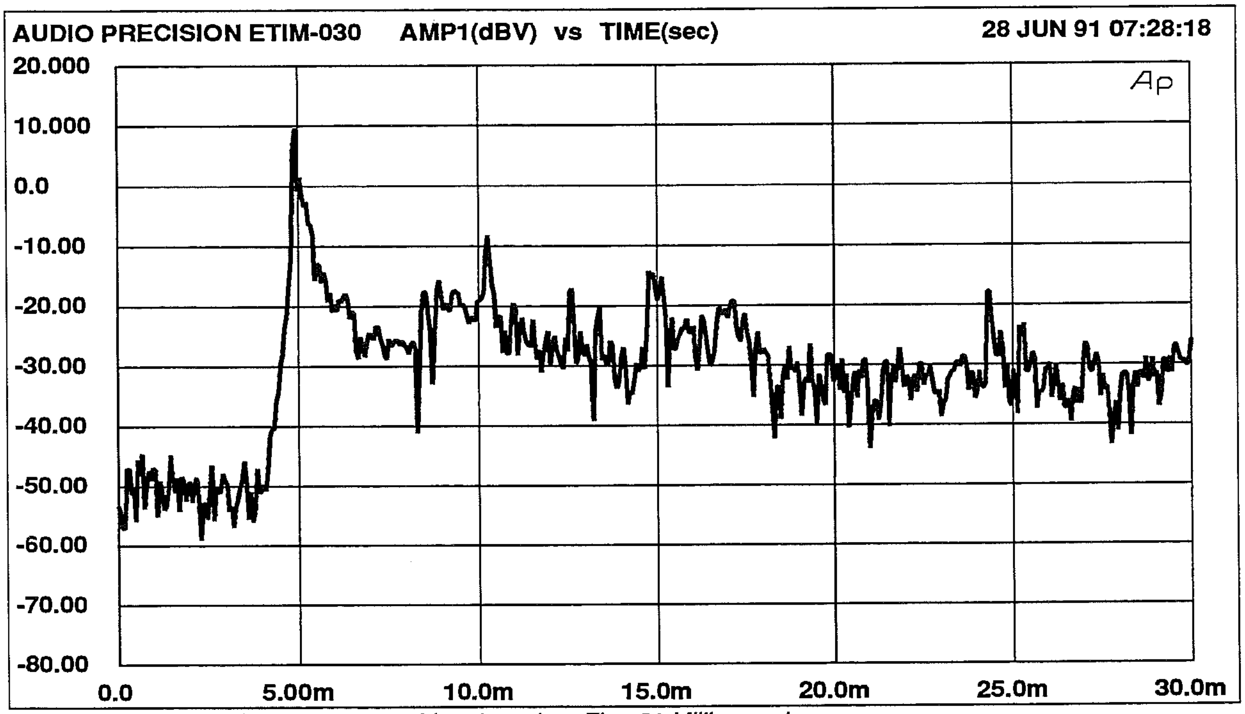

Figure D-4 Energy-Time Response of Loudspeaker, First 30 Milliseconds

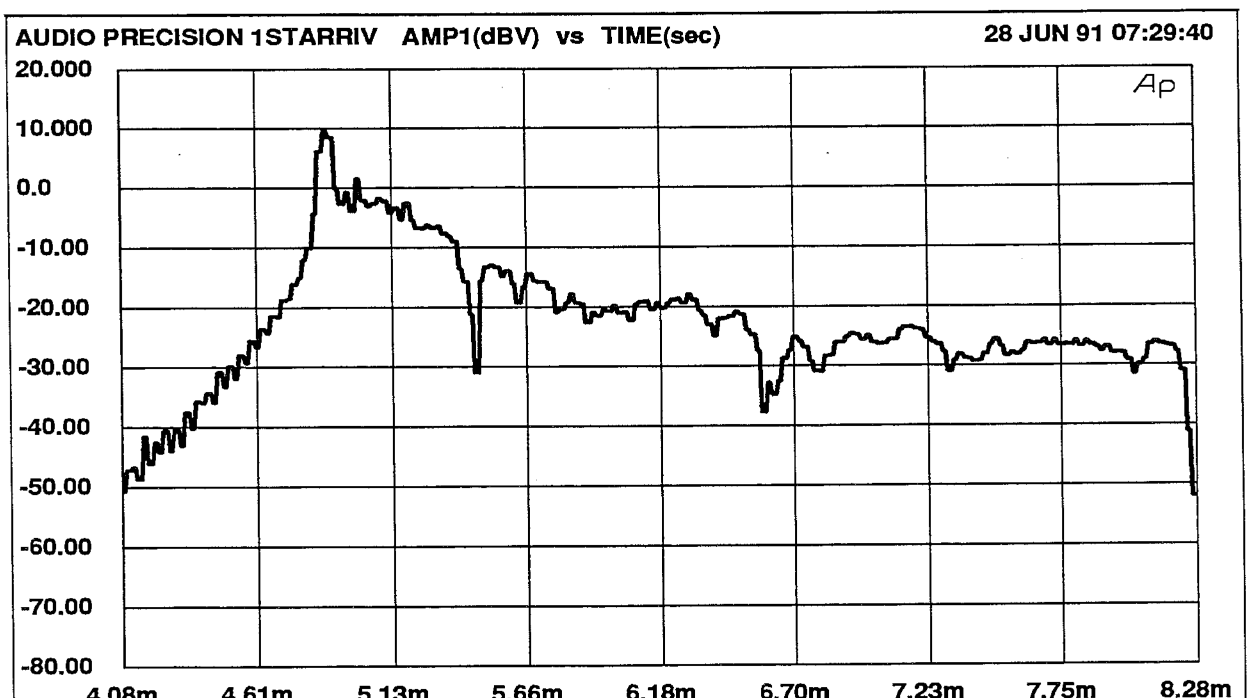

Figure D-5 Energy-Time Response of Loudspeaker, First Arrival Signal Only, Before Reflections

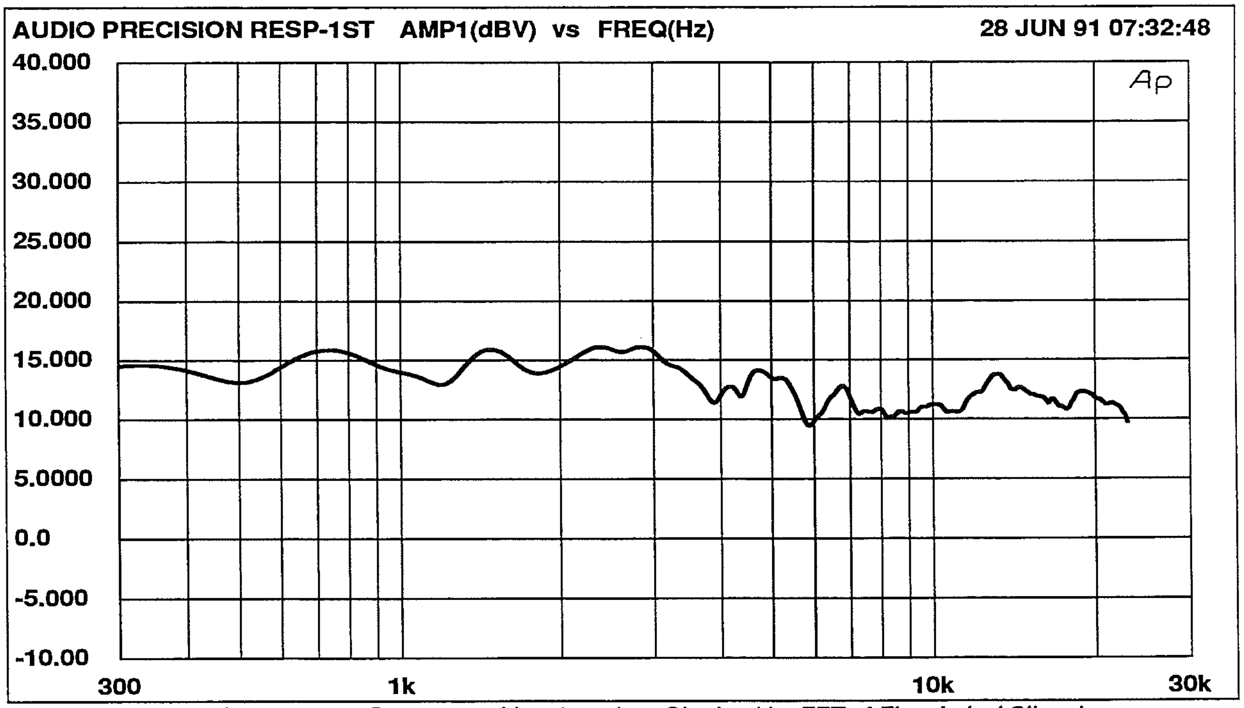

Figure D-6 Anechoic Frequency Response of Loudspeaker, Obtained by FFT of First-Arrival Silgnal

D.3.2.2. Frequency Domain Display

With DSP FREQ selected at SOURCE-1 as the horizontal axis, NORMAL display mode results in the amplitude value of each requested bin frequency being sent from DSP to computer with no further processing. NORMAL is the recommended display mode for frequency response data with a LIN horizontal axis or with a LOG axis above 100 to 300 Hz. In these cases, the jagged lines caused by the FFT bin width are not usually noticeable.

The INTERPOL display mode with frequency domain displays interpolates a value from the 15 FFT bins centered at the requested frequency and sends that computed value to the computer for plotting. INTERPOL thus smooths out the "stair-step" appearance of frequency response curves at low frequencies with a LOG horizontal axis, where the bin width (usually 2.93 Hz at the 48 kHz sample rate) occupy a significant portion of the screen.

Peak display mode provides the same "peak-picking" operation between computer-requested frequencies which is the standard frequency-domain display mode of FFTSLIDE.DSP and FFTGEN.DSP. If the bin width, LOG vs LIN selection at SOURCE-1, and # STEPS parameter at SOURCE-1 result in any FFT bins not being directly interrogated by the computer, the DSP will scan across all bins between the last-requested and currently-requested value and send to the computer the amplitude of the bin with the strongest signal. Peak mode would not normally be used for frequency response displays with MLS.DSP, since high values are of no more interest than low values when plotting frequency response.

D.3.3. Setting Time Spans for Frequency Domain Analysis

The section of the impulse response which is to be transformed into the frequency domain is selected by the most recent set of START and STOP time values used in a time domain display sweep. Before selecting a frequency domain display it is es-

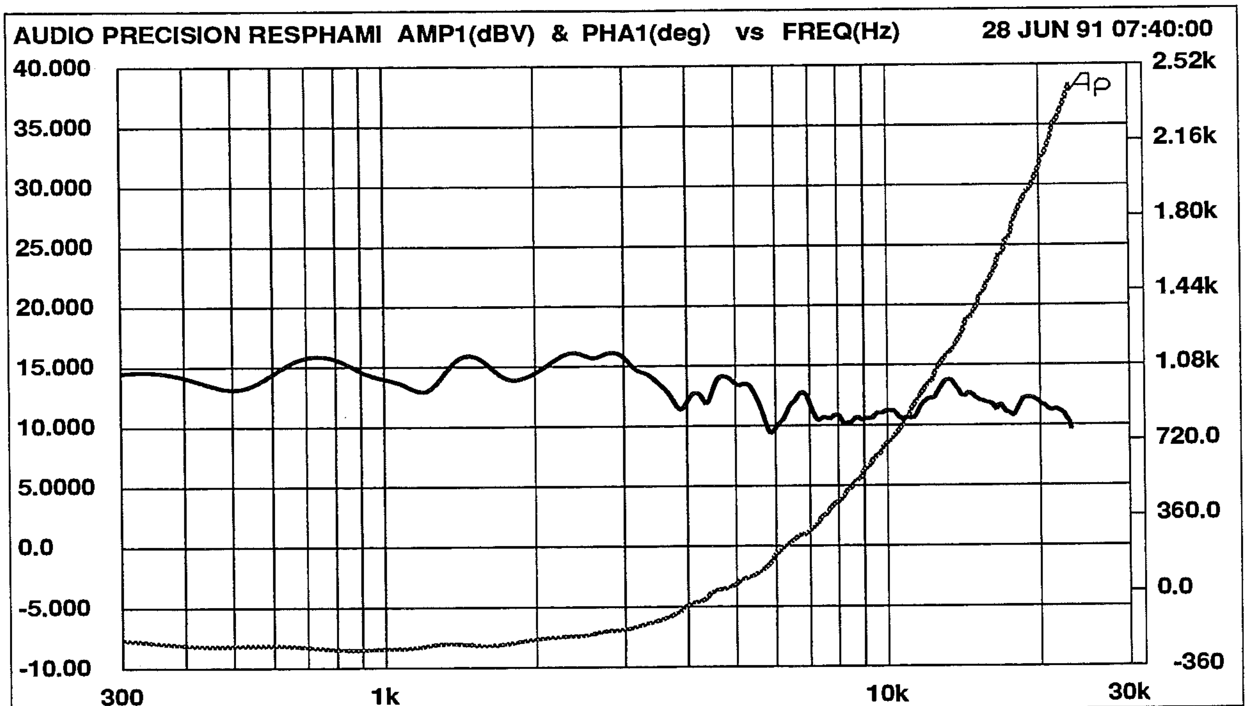

Figure D-7 Magnitude (Darker Trace) and Phase Response (Lighter Trace) of Three-Way Loudspeaker, REF TIME Adjusted for Flattest Phase Response Through Mid-Range Frequencies

sential that a time domain display be selected and plotted. See Figure D-4 for an illustration of the first 30 milliseconds of an impulse response of a loudspeaker. If it is desired to limit the analysis to only the direct sound, for example, the SOURCE-1 START and STOP values must set to the beginning and end of the direct sound arrival. The <F6> key must then be pressed to send this time span information to the DSP before frequency-domain analysis of an already-acquired signal, or the <F9> key pressed to acquire a new signal. See Figure D-5 which is the same impulse response, but with START and STOP times set to span only the first-arrival signal before reflections.

If the characteristics of the impulse response are not known in advance, the START and STOP values can be set to a wide span which includes the whole impulse response. For typical loudspeaker testing applications this might be 0 and 50 ms respectively. The direct sound and the first reflections can be identified and the graph "zoomed in" by reselection of START and STOP times to display only the portion of interest. This is typically the direct sound only, and the start time is set to the beginning of the build-up of the first peak in the response. By studying the decay of this response, the onset of the first reflection can be determined. The STOP value is then set to the beginning of the reflected signal. Figure D-6 shows the frequency response of the first-arrival impulse from Figure D-5, thus excluding the effects of all reflections.

D.3.3.1. Relationship Between Time Span and Lowest Usable Frequency

The setting of START and STOP values limits all subsequent frequency domain analysis to the portion of the time record bounded by these values. An impulse response will oscillate or be active for a length of time roughly proportional to the period of its lowest frequency component. The time record must be long enough to include this oscillation to obtain meaningful information about the lowest frequency component. For accurate measurements this oscillation must be allowed to die down for several cycles, requiring a time record approximately three times the period of the lowest frequency of interest.

For example, the period of a 200 Hz signal is 5 ms, implying a 15 ms required time span. For loudspeaker measurements this also implies an approximate 5 meter (16 foot) distance between arrival paths of the direct sound and the interfering reflections, which requires an unusually large space for testing. Note that the time axis may be displayed with feet or meter units in addition to seconds.

D.3.3.2. Setting Phase Offset Value

The REF TIME field is used to compensate for the effect of time delay on the measurements. This is of concern only when making phase measurements since a time delay becomes an increasing phase shift with frequency. This field is normally set to a value slightly greater than the START TIME value entered on the SWEEP DEFINITIONS panel to select the time response. These values are usually not identical since the impulse being measured is spread out in time by the device under test and the START TIME must be set to include the build-up of the impulse, while the REF TIME is set to the exact peak of the impulse. In practice, the REF TIME is "fine tuned" by examining the unwrapped phase response and adjusting the value for the flattest curve. For the example in loudspeaker impulse response in Figure D-5, an appropriate START TIME would be 4.08 ms to include the build-up of the impulse, while the optimum REF TIME is found to be 5.25 ms for the flattest phase response through mid-range. The corresponding phase response plot is shown in Figure D-7, resulting in an accurate representation of the loudspeaker phase response. Note that the phase response curve has a large inflection to a different slope at the midrange-tweeter crossover frequency, indicating that the tweeter and midrange drivers are not precisely time-aligned (linear phase). By setting the REF TIME to a value smaller than the correct value, the phase response exhibits a downward slope with increasing frequency. The additional time delay added to the measurement causes an increasing number of degrees of phase shift in direct proportion to the frequency. If the REF TIME is set too large the phase slope would be reversed.

D.3.4. Displaying Energy-Time Response

The time domain behavior may be examined using the impulse response (IR) or the energy-time response (ETim). The term energy-time response is a misnomer since a true computation of energy requires knowledge of both kinetic and potential energy and a microphone signal can only supply one of these. A more accurate term sometimes used in technical papers is the analytic signal magnitude. However, the term energy-time has become common usage and, to avoid confusion, will be used here. ETim curves may also be considered as similar to the envelope of the impulse response.

The software computes an estimate of the energy by applying a Hilbert transform to the frequency domain data before transforming back to the time domain. The resulting trace will not show the negative excursions of the impulse response. This display is useful for determining arrival times and relative energy distribution in time.

The energy-time mode is selected with the right-most TIME SWP field. The choices are IR for impulse response and ETim for energy-time response. When the IR mode is selected, all time plots will display the impulse response. If ETim is selected, the software will estimate the energy-time response of the device under test and display it when a time graph is requested.

Since the energy-time graph is computed with transforms, a window must be applied to the data to prevent alias behavior. The window to be used is selected with the left-most TIME SWP field. The choices are no_window, half_Hann, Hann, 240_8kHz, and 120_16kHz. These windows operate in the frequency domain and are plotted in Figure D-14 and Figure D-15 near the end of this chapter. The no_window selection will perform the required transformations with all frequency components of the signal included in the computations. The deviations from a flat frequency response create ripples in the time domain energy response. The Hann selection is the one window found on software from other manufacturers. This reduces both high and low frequency energy, concentrating on ar-

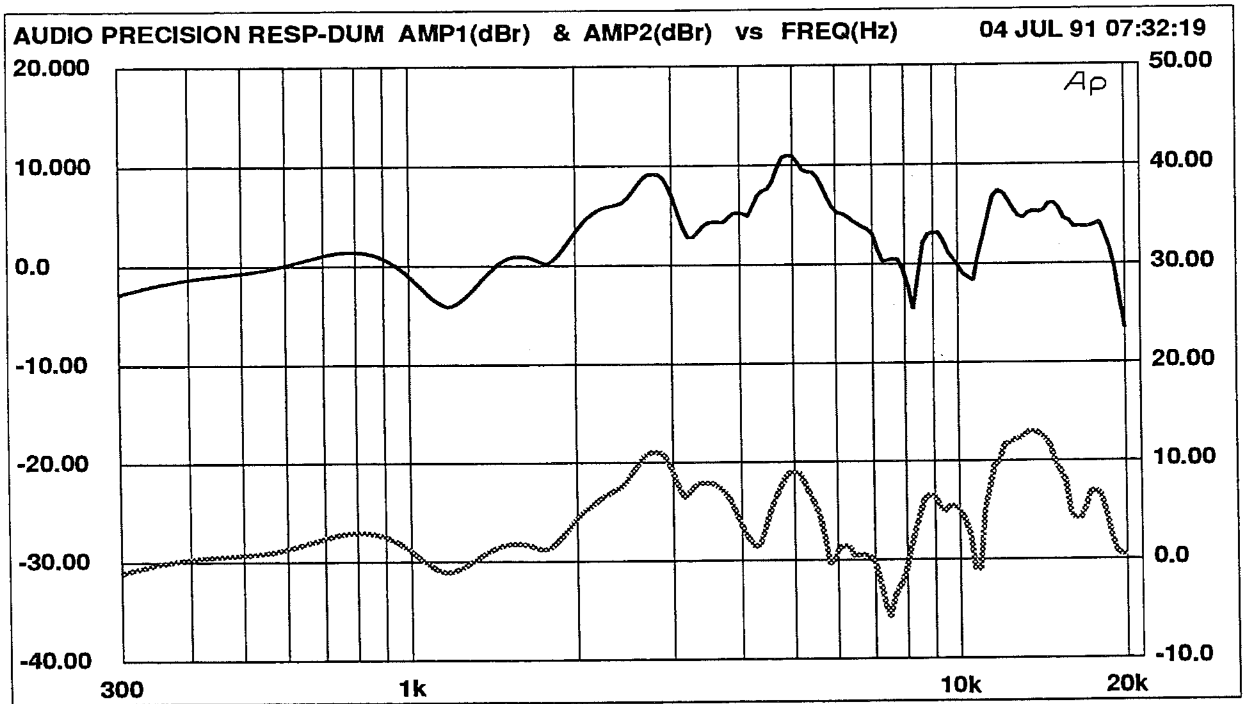

Figure D-8 Frequency Response at Left Ear (Darker Trace, Calibrated at Left) and Right Ear (Lighter Trace, Calibrated at Right) of Dummy Head

rivals at the center of the frequency range. Since the processing occurs on a linear frequency scale, this will focus analysis on signals around one quarter of the sample rate. At 48 kHz this will result in the 12 kHz energy dominating the energy-time display. This produces very attractive displays which are very wrong.

The half_Hann selection is a window suggested by Lipshitz and Vanderkooy which only reduces the contribution of high frequencies. The low frequency information remains unchanged. When operating at the 48 kHz sample rate this window filters out energy above 12 kHz. Audio Precision developed the remaining two windows for even more accurate measurements of typical audio signals. The 240_8kHz window filters energy below 240 Hz and above 8 kHz, producing equal sensitivity to signals over a 5 octave range. The 120_16kHz window spreads the analysis over a 7 octave range. Both windows produce much more accurate results than the Hann window with only minor increases in alias behavior.

Software from other manufacturers which performs energy-time computations does not allow selection of these windows and usually defaults to a Hann window. The errors introduced can be considerable, so care should be taken when comparing data from alternative measurement equipment.

For an excellent technical discussion of the effect of windows on energy-time-curve computation see Lipshitz and Vanderkooy, "Uses and Abuses of the Energy-Time Curve", Journal of the AES, Vol. 38, No. 11, November 1990, pp. 819-836.

D.3.5. Sweep (F9) Panel Fields

All four DSP readings are available as measurement choices at both the DATA-1 and DATA-2 fields for plotting onto graphs. Similarly, all four DSP numerical settings are available as horizontal axis choices at both SOURCE-1 and SOURCE-2. However, two of these choices are only useful as SOURCE-2 sweeps. Other measurement modules such as the DCX or the analog ANALYZER may not be used for DATA-1 or DATA-2 along with an MLS measurement.

D.3.5.1. Data-1 and Data-2 Choices

With MLS.DSP loaded the DATA-1 and DATA-2 choices are AMPL-1, AMPL-2, PHASE-1 and PHASE-2. These are the results for the two DSP channels (CH-1 and CH-2). Most amplitude and phase units used elsewhere in S1.EXE software are available. Several of the amplitude units are absolute: %FS, dBFS, V, dBV, dBu, dBm, and W. The two FS units are meaningful only when a digital input signal has been acquired (Dual Domain units only), where the FS refers to digital full scale. dBFS will normally be used for frequency response measurements of digital equipment, and %FS (linear) for impulse response displays of digital equipment.

The dBr unit is relative to the analog ANALYZER dBr REF value, which may be entered from the keyboard on the ANALYZER panel or may be automatically set to a value being presently measured by the analog analyzer by pressing the <F4> key. The dBm and W (Watts) units use the analog analyzer REF dBm/W field impedance value to perform their power computations with the measured voltage.

For frequency response displays, the usual DATA unit selection will be either V (Volts) RMS with a LOG vertical display, or a dB unit (dBV, dBu, dBm, etc.) with a LIN vertical display. For impulse response displays, DATA may either be displayed in V units with a LIN vertical display (typically with a positive value for graph top and the symmetrical negative value for graph bottom) or in LOG V or dB units with the Peak selection on the DSP panel.

D.3.5.2. Split Screen Effects

While S1.EXE software does not have a true split-screen mode, it is possible to display two spectra or two waveforms simultaneously on the upper and lower portions of the screen by proper selection of DATA-1 and DATA-2 GRAPH TOP and BOTTOM values. See Figure D-8 for an example of such a display. The upper half of the screen shows the response of one ear on a dummy head, the lower half shows the other ear.

All the information in each test lies between the amplitudes of +20 dBr and -10 dBr. The span between these values is 30 dB. The DATA-1 TOP and BOTTOM values were set to twice that span, or 60 dB, with the TOP value set to the upper end of the range (+20 dBr). The DATA-2 TOP and BOTTOM values were also set for a 60 dB span, but with the BOTTOM value set to the lower end of the range (-10 dBr). Since in this case the two sets of data have identical numbers of samples, the frequency values (or time values for waveform display) will exactly correlate. The graphic cursor can thus be moved along the horizontal axis with both signal amplitudes being numerically displayed in the windows at the screen top.

D.3.5.3. Source-1 and Source-2 Choices

MLS offers REF TIME, GAMP, TIME and FREQ as SOURCE-1 and SOURCE-2 choices. TIME is selected for waveform display, with the START and STOP values chosen to display the desired section of the waveform. This is also used to select the segment of the impulse response to be transformed into the frequency domain. TIME is normally displayed on a LIN horizontal axis, but LOG displays are possible. FREQ is selected to display the frequency response of the particular impulse arrival. The frequency START and STOP values may be chosen freely based on the frequency range of interest. However, it is essential that the appropriate time span be set using a TIME sweep before performing a FREQ sweep. When used as a SOURCE-2 selection, GAMP allows measurements to be repeated at several different stimulus amplitudes. This can be useful when studying the device response as a function of signal power. With loudspeakers the response may vary with level due to voice coil heating. Electronic noise reduction and dynamic range control devices may purposely introduce response variations with level which can be quickly characterised with MLS.DSP. The GAMP selection may not be used as a SOURCE-1, improper operation will result. The REF TIME is not useful for sweeping.

D.3.5.4. Resolution and Display Steps

The frequency resolution depends upon the record length (number of samples) upon which the FFT was performed; there will be half as many FFT bins as there are amplitude samples in the record transformed. Shorter spacing between START time and STOP time will result in a smaller record being transformed and a correspondingly poorer frequency resolution. See the "Actual vs Apparent Resolution" section below for more information on this aspect.

The graphically displayed resolution also depends upon the # STEPS parameter in the SOURCE-1 area of the SWEEP (F9) DEFINITIONS panel and upon the graphics display system in use. With transform lengths longer than about 1000 points (impulse time spans greater than approximately 21 milliseconds with a 48 kHz sample rate), frequency resolution better than 500 points will result. With # STEPS much larger than 500, the horizontal resolution of the computer display system will also act as a limiting item. VGA, EGA, and CGA monochrome all have 640 pixel resolution across the entire screen. About 81% to 88% of this value is available for graphic data after space is taken up for borders and calibration at the left or both sides. Thus, on the order of 520 to 540 pixels are available horizontally. Numbers of sample points within the selected START-STOP time span greater than about 1040 to 1080 and higher # STEPS than 520 to 540 will result in more than one data point being plotted in the some pixel columns. This may still be useful if the ultimate graph is to be reproduced on a high resolution laser printer or plotter via the PLOT or POST utilities, since this technique is not limited by display system resolution.

If the # STEPS parameter is set to a smaller number than the available data points, and Peak mode is selected, DSP program software will select the highest amplitude in each group of points as it sorts them into the smaller number of points to be displayed. Thus, peaks in the frequency response or impulse response will not be missed even when the displayed resolution is less than the MLS resolution.

It is possible to use a LOG frequency axis for frequency response display, though the data itself is inherently linear on the frequency axis. For a LOG display, the START frequency must be a positive non-zero value.

If response data is displayed with a linear frequency axis, it is often convenient to select the # STEPS parameter and STOP frequency to produce round numbers for convenient use of the graphic cursor feature. For example, with a 0 Hz START frequency, 20 kHz STOP, LIN display, and # STEPS 500, the displayed data points will spaced 40 Hz. The graphics cursor will then move in individual 40 Hz steps when the horizontal arrow keys are used alone, 200 Hz steps with <Ctrl> arrow (5 times magnification), and exact one kHz steps when the <Shift> key is held down while a horizontal arrow key is pressed (25 times magnification). If a mouse is used, rolling the mouse horizontally would move the cursor through every point. Holding down the left mouse button while rolling would move through every fifth point (200 Hz steps), and holding down the right button while moving would move the cursor through every 25th point (one kHz steps).

D.3.5.5. Graphic Aliasing

Depending upon the exact relationship between # STEPS, the signal, and the START and STOP values, some features of the impulse response may be missed. To help insure freedom from display aliasing effects it is best to use WAVEFORM Peak mode for IR and ETim displays, since Peak will return the highest waveform sample value in the span between the current display STEP and the previous STEP.

D.3.5.6. Spectral Displays

FREQ is selected at SOURCE-1 for spectral displays. An FFT is inherently a frequency-linear process starting at zero Hz, so the LIN horizontal axis is often selected with a START frequency of 0. LOG may be selected for the display if desired.

Note that to obtain a LOG display, the START frequency must be a positive non-zero value. If the WAVEFORM INTERPOLATE mode is selected the visual effect will be to make the frequency response smooth at low frequencies. If the WAVEFORM NORMAL mode is selected the visual effect will be to make the frequency response look coarse at low frequencies since the actual FFT bins are actually linearly spaced. If the WAVEFORM INTERPOLATE mode is selected the frequency response will be smoothed because the DSP will interpolate values between the actual transformed points (bins).

The ultimate actual resolution of the data will always be limited by the length of impulse response selected for the transformation. For a minimum time between START and STOP values in the time domain display of 5 milliseconds, for example, the frequency response resolution is limited to 200 Hz. Interpolation will make it appear that the resolution is better, but the fundamental resolution cannot be greater than the frequency whose period equals the selected time span.

D.3.6. <F9>, <F7>, <F6> Keys

The <F9> key initiates a new cycle of signal acquisition, processing, transmission to computer, and display. If it is necessary to go to the panel or menu, the display may then be re-graphed with the <F7> key. Changes in GRAPH TOP and GRAPH BOTTOM values may be made and re-graphed via the <F7> key since all the displayed information is in computer memory.

If the display is to be changed from impulse response to frequency response (or vice versa), it is not necessary to re-acquire data. The <F6> key will cause a new cycle of processing (FFT transform if frequency domain display is selected at SOURCE-1), transmission to computer, and display to take place from the acquired data which is still in DSP memory.

Changes between interpolated versus non-interpolated waveform or different windows in the Energy-Time mode may be effected with the <F6> keystroke. Changes in the # STEPS value and SOURCE-1 START and STOP values on the Sweep (F9) Panel may also be effected with the <F6> key in order to re-process the DSP-memory-stored data according to the new parameters.

D.3.7. Changing Between Waveform Display and Frequency Response

Waveform display and frequency response tests may be loaded without destroying the waveform data in DSP memory if both tests have the same DSP file attached. You may acquire a signal with any one of these tests and then easily switch analysis domains by simply loading another of these tests and pressing <F6>. This is generally significantly easier than making all the necessary DATA-1, DATA-2, and SOURCE-1 changes to go between time and frequency domain displays.

D.4. Memory Size Effects

The impulse response measurement length is always the maximum which can be acquired into the available memory. This is done to reduce the risk of errors which occur when the impulse response being measured is longer than the measurement length. When this happens the portion of the impulse response which exceeds the measurement buffer will fold over to the beginning of the buffer.

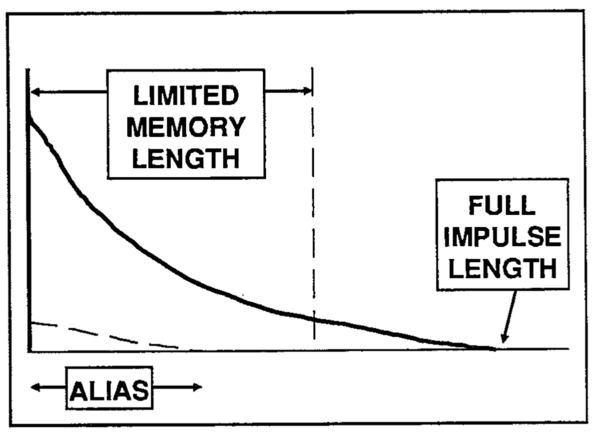

Figure D-9 Time Domain Alias When Impulse Length Exceeds Memory Length

This is a form of time domain aliasing as illustrated in Figure D-9. The measurement accuracy will be limited by the amplitude of the portion which folds relative to the desired signal. If the largest amplitude in the aliased portion of the impulse response is 1% (-40 dB) of the non-aliased portion, frequency response errors will be approximately 1% or 0.1 dB. With a standard memory System One + DSP the maximum impulse length before aliasing is 8191 samples. This is 170 ms at a 48 kHz sample rate or 256 ms at a 32 kHz rate. A System One Dual Domain or a System One + DSP with the MEM option will provide 32787 samples of impulse response before aliasing. This is 683 ms at a 48 kHz sample rate or 1.024 seconds at a 32 kHz rate. The amount of memory available will be displayed as part of the HELP panel information.

Loudspeaker impulse responses are seldom long enough to create a problem on any memory size System One. However, when a loudspeaker is measured in a non-anechoic room, the room impulse response will dominate and can create errors on a System One + DSP with standard memory. The room reverberation time is a rough indicator of when trouble might occur. The reverberation time is the time required for the room energy to decay by 60 dB. If measurements to 1% accuracy are desired the energy must decay by 40 dB within the time represented by the measurement buffer. Assuming the reverberation is a single exponential decay, the allowable reverberation time (at a 48 kHz sample rate) would be 1.024 seconds for full memory but only 256 ms with standard memory.

D.4.1. Actual vs Apparent Resolution

The equivalent frequency resolution of an FFT in the general case is obtained by dividing the sample rate by the number of waveform samples. With maximum memory (Dual Domain unit or a unit with MEM option), the MLS.DSP transform length is 16,384 samples. At a 48 kHz sample rate, the resulting FFT will consist of 8192 spectral lines (bins) evenly spaced from zero Hz to the Nyquist frequency (1/2 sampling rate). The resulting bin width and frequency resolution is therefore approximately 2.93 Hz. An SYS-200 series unit without the MEM option has a memory length of 4096 samples and thus provides one fourth this resolution, or 11.7 Hz.

With MLS.DSP, actual resolution is further limited by the fact that only a specific time span of the acquired signal is usually selected for analysis by the SOURCE-1 START and STOP times. The remainder of the transform buffer is padded out with zeros. Actual resolution is determined by the number of non-zero samples. That actual resolution is then effectively interpolated, with the interpolation resolution (apparent resolution) being the 2.93 Hz or 11.7 Hz value described above. Typical selected time spans are on the order of several milliseconds for anechoic measurements (with limited low-frequency response) up to tens of milliseconds for full frequency-range measurements. At the 48 kHz sample rates, these time spans may thus include on the order of 150 to 1500 data samples and produce actual resolutions on the order of 320 Hz (48000/150) to 32 Hz (48000/1500).

D.5. Dual Channel Operation

The DSP hardware and MLS software are both capable of two-channel operation. Two independent A/D converters permit simultaneous acquisition of two analog signals. The digital interfaces are also of two-channel architecture. The input signal selection capability of the analog interface permits assigning a signal to one DSP channel and another related or completely independent signal to the other channel. The digital interface permits selecting the "A" digital signal channel as one signal and the "B" signal as the other. Both channels will be acquired, processed, and displayed when the <F9> key is pressed. Both will be re-processed and displayed when the <F6> key is pressed. See the Split Screen Effects section on page B-10 for suggestions on how to display dual channel data.

It is also possible to acquire a signal into only one channel and then later acquire another signal into the other channel. If one of the two channel input selections near the bottom of the DSP panel is set to NONE, no signal will be acquired into that channel when <F9> is pressed. Conditions or devices may be changed to another desired test condition. Then, the originally-used input channel can be set to NONE and the desired signal source selected at the channel where no signal has yet been acquired. A second operation of the <F9> key will acquire data into that channel but not over-write the data in the channel with NONE selected for input. The data in both channels will then be processed, transmitted to the computer, and displayed.

Some conditions and precautions must be observed when acquiring signal into the two channels at different times:

- Any acquired signals will be lost when AC mains power to System One is turned off or when another DSP program is named (changing from MLS.DSP to FFTGEN.DSP, for example). Thus, the two signals must normally either be acquired during the same testing session, or saved to disk using the SAVE WAVEFORM capability and later combined via LOAD WAVEFORM.
- Signal in both channels will be processed every time <F6> or <F9> is pressed, ***according to the present settings on the DSP panel.*** It is thus not possible to use different windows or different display options for the two channels. ***More important, both sets of data must be acquired at the same sample rate.*** Processing will be done according to the rate selected at the time of pressing (<F6> or <F9>). If the data in one channel had been acquired at a different rate, it will be erroneously transformed or displayed with a resulting frequency error directly proportional to the difference in the two rates. When loading a waveform the sample rate will be automatically set to the value in effect when that waveform was saved. If two waveforms are loaded in sequence, the rate will be left at the rate used by the last waveform loaded.

D.5.1. Program Operating Speeds

The time required to acquire the signal may be calculated by dividing the memory length being filled by the sample rate. Thus, when a full memory unit which acquires 32373 samples is operating at a 48 kHz sampling rate, the acquisition will take about 680 milliseconds after the <F9> is pressed. Note that MLS.DSP always fills the entire memory, so acquisition times are independent of the time segment being displayed. The time required for the DSP chip to actually perform the correlation and FFT is typically small compared to the time to transmit the results to the computer. The transmission time from DSP module to computer is proportional to the number of data points (# STEPS + 1) and depends strongly on the speed of the personal computer. The table below shows the approximate times for acquisition (assuming a 48 kHz sample rate) and computation (windowing, correlation, transforming, magnitude and phase). The additional time for transmission to the computer plus units conversion and graphing is approximately 2.2 seconds for 512 points plotted on a 20 Mhz 80386-based computer with 80387 math co-processor and color VGA display system. All other speeds shown are independent of the computer type being used.

# of channels	Acquire Time	Computation Time
1	680 ms	600 ms
2	680 ms	800 ms

D.6. Saving Waveforms to Disk

The SAVE WAVEFORM and LOAD WAVEFORM commands permit saving acquired waveforms to disk and later loading those waveforms back into the DSP module for further analysis. The .WAV file extension is automatically supplied by S1.EXE software during SAVE operations, and is expected by the LOAD command in order to display a menu of available waveforms in the current directory. Optional implementations of these commands permit saving waveforms from either or both channels of the DSP. The first 128 characters of the Edit Comments buffer (or text up to the first carriage return, which ever comes first) will be saved into the .WAV file.

D.6.1. Save Waveform Command

The SAVE WAVEFORM command may be used whenever acquired data is in the DSP module, resulting from either an <F9> key operation or a previous LOAD WAVEFORM command. SAVE WAVEFORM causes the impulse response values (not the FFT or the interpolated-for-display values) to be transferred from data memory within the DSP modules to the computer and stored in a disk file. SAVE WAVEFORM must be invoked with one or two numerical arguments. The general form of the command is:

SAVE WAVEFORM filename <Enter>

followed by

[,#] <Enter>

where # may be 1 or 2 and refers to CHAN-1 and CHAN-2 waveform data in the DSP modules.

Thus, selecting SAVE WAVEFORM results in the user being asked to supply a legal DOS file name into which the data will be saved. The .WAV file extension is automatically furnished by the system. Upon furnishing the file name and pressing <Enter>, the system will say:

Enter buffers to be used "#[T or G] [,#[T or G]]":

The # represents the digit 1 or 2 to control which data will be saved into which portions of the disk file. The first numeric code entered defines whether CHAN-1 or CHAN-2 data from DSP memory is to be stored in the first of two buffers within the disk file. The second (optional) numeric code permits specification of the alternate DSP channel data to be stored into a second buffer within the disk file. The optional alphabetical codes (T and G) are not relevant to the MLS program. For example:

SAVE WAVEFORM filename <Enter> 1,2 <Enter>

will cause data from CHAN-1 of the DSP to be saved into the first portion of the disk file and data from CHAN-2 to save into the second portion of the file. This is also the default operation if no codes are entered.

The command:

SAVE WAVEFORM filename <Enter> 2 <Enter>

will cause only the CHAN-2 DSP data to save into the disk file.

SAVE WAVEFORM filename <Enter> 2,1 <Enter>

causes CHAN-2 data to save into the first portion of the disk file and CHAN-1 DSP data to save into the second part of the file.

It is not currently possible to save less than the full DSP memory contents by using any of the SAVE WAVEFORM options, thus the T argument is not useful.

In all cases, the necessary range information and sample rate information is transferred with the data and saved into the file.

D.6.1.1. Waveform File Size

The disk files created by SAVE WAVEFORM are very large compared to normal System One .TST files. With a maximum-memory unit, saving one channel results in about a 96 kbyte file and both channels will result in a file of approximately 190 kbytes. The exact file size will consist of 3 bytes per sample per channel (since the DSP acquires 24-bit data words and computer disk files bytes are 8 bits wide) plus 256 bytes of header per channel. Thus, a full 32,767 sample acquisition will occupy 98,301 bytes per channel, plus 256 header bytes for a total 98,557 bytes per channel. A dual-channel file is thus 197,114 bytes.

D.6.2. Load Waveform Command

The LOAD WAVEFORM command may be used to down-load the stored information from disk files back into the DSP modules for further analysis and display. Waveforms saved from MLS may not be loaded into another program. LOAD WAVE-FORM is also invoked with one or two numeric arguments, which can be 0, 1, or 2. The position of these numbers in the command defines whether they refer to the first or second portion of data in the disk file. The value of the argument determines whether that data should simply be discarded (0), loaded into DSP CHAN-1 (1), or loaded into DSP CHAN-2 (2). Thus, the command:

LOAD WAVEFORM filename <Enter> 1,2 <Enter>

causes the first portion of the disk file to load into CHAN-1 and the second portion to load into CHAN-2. This is the default operation if no numbers are entered. The command:

LOAD WAVEFORM filename <Enter> 0,1 <Enter>

causes the first portion of a two-record disk file to not load into the DSP, while the second portion loads into CHAN-1 of DSP memory. The optional G argument is used with certain other programs (FASTEST.DSP, FFTGEN.DSP) to download waveforms for arbitrary waveform generation. MLS waveforms are the only generation capability of MLS.DSP, thus the download capability does not exist and the G option is not used.

D.6.3. Using Down-Loaded Waveforms

When a stored waveform has been downloaded into the DSP, the <F6> function key must be used to transform the data and send it to the computer for graphing. As with a freshly-acquired waveform, the SWEEP PANEL parameters may be changed for time domain or frequency domain display, individual sections of the time or frequency axis may be expanded, interpolation turned on and off, and a re-display made each time with the <F6> key.

D.6.4. Combining Waveforms Acquired at Different Times

The LOAD and SAVE capabilities may be used to combine into the dual channels of the DSP waveforms acquired at different times or places. Proper use of the numeric arguments during the LOAD operation allows any saved waveform to be placed into either the CH-1 or CH-2 data memory of the DSP. However, it is critical that both these waveforms must have been acquired at the same sample rate. If waveform data files acquired at two different sample rates are loaded into DSP memory and the <F6> key pressed, both channels will be transformed and displayed according to the sample rate on the DSP panel (that of the last waveform loaded) when <F6> is pressed. The result will be directly proportional frequency errors in the record acquired at a different rate.

D.7. Testing Applications Examples

The MLS program diskette contains several typical tests already set up for use with the MLS program. This section describes the use of these tests.

D.7.1. Basic Impulse Response Measurement

IMPLSLOG.TST on the DSP diskette is set up for signal acquisition through the analog analyzer balanced inputs. It will perform impulse response display with a 22 kHz bandwidth (48 kHz sample rate). The DATA-1 graph top and bottom values are selected for optimum log display of an impulse of approximately one Volt, which may be appropriate when a measurement amplifier follows the microphone. For an unamplified microphone output, levels on the order of a few millivolts are more typical.

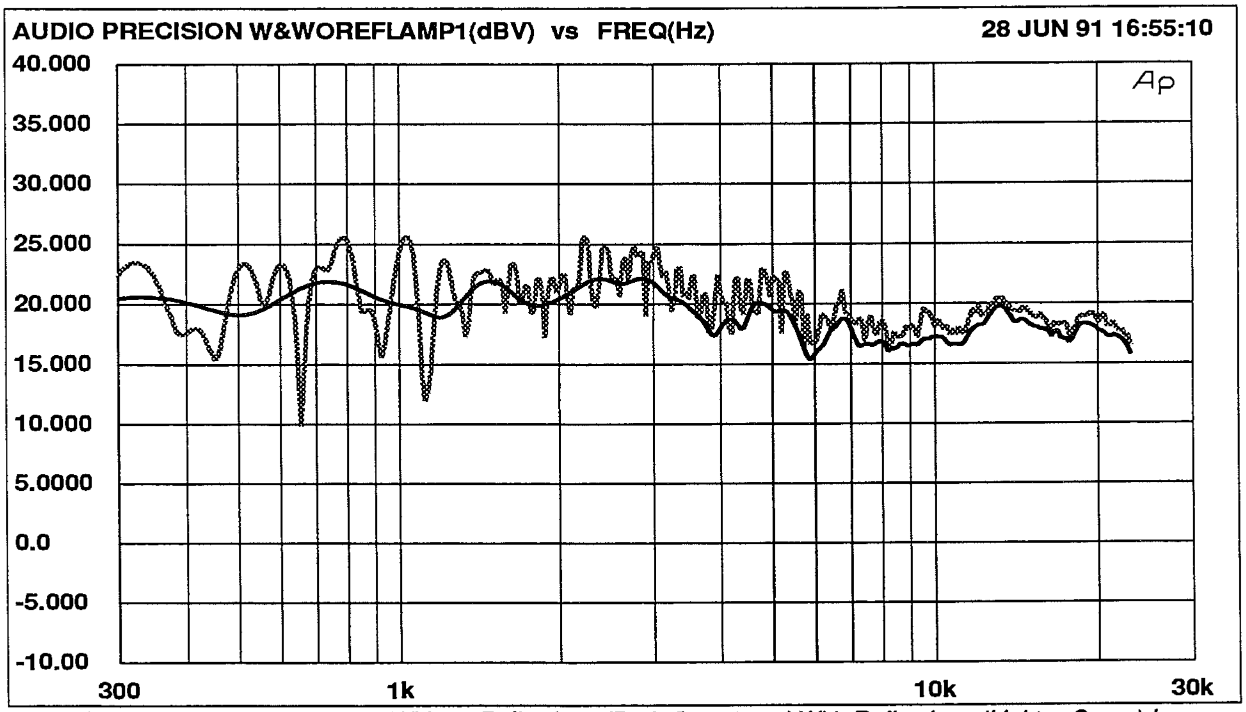

Figure D-10 Frequency Response Without Reflections (Dark Curve) and With Reflections (Lighter Curve) Included in the Measurement

Note that the horizontal axis units may be seconds, feet, or meters as desired.

D.7.2. Speaker Response Measurement

When measuring a speaker in a room, the microphone will receive three types of signals; sound travelling directly from the speaker to the microphone, sound from the speaker which has reflected off surfaces in the room, and interfering noise. The DSP algorithms in MLS.DSP drastically reduce the effect of interfering noise but cannot reduce the sound from reflected surfaces. These reflections may be excluded using MLS.DSP by proper selection of START and STOP times, but the action of excluding them limits low frequency measurement capability.

The direct sound from the loudspeaker reaches the microphone after a time delay proportional to the distance between them. At the 21 degree Celsius sound propogation velocity of 1129 feet/second (344 meters/second), this delay is approximately one millisecond per 1.13 feet (13.6 inches or 34.4 cm). When the sound from the speaker reaches the wall some will be reflected back toward the microphone, resulting in a reflection or echo in the impulse response. Additional reflections will occur from the other walls, the floor, the ceiling and other surfaces such as desks or cabinets. Each will arrive with a delay proportional to the length of its path from the speaker to the microphone. The delayed reflections change the speaker frequency response as illustrated in Figure D-10.

In a typical room the floor-to-ceiling height will be the limiting dimension when optimizing speaker and microphone location for maximum spacing to the first reflection. This optimum will be achieved with both speaker and microphone midway between floor and ceiling. When setting up the equipment, do not ignore the presence of reflecting objects in the room such as desks, bookcases, lamps (including ceiling-mounted hanging lamps) and even the stands used to hold the speaker and microphone during the measurements. These can all create reflections which can be bothersome, especially at high frequencies where the size of the object may be large compared to the wavelength of sound. The clamp and stand holding the measurement microphone are often between 1/2 and one inch in diameter. The presence of this reflective surface can have several dB effect on response between 10 kHz and 20 kHz. Wrapping the stand and clamp with sound absorbing material will usually eliminate this error source. The stand used to hold the loudspeaker is similarly important to the response. Stacking the speaker on top of shipping boxes, usually having several feet of reflective surface, will result in a quite different response from mounting the speaker on a heavy duty tripod which is only an inch wide.

The distance between the speaker and microphone should be minimized to increase the ratio of reflected sound path length to direct sound path length. However, care should be taken not to make the speaker-to-microphone spacing so small that the near field response dominates if a far field measurement is desired. Near field measurements will emphasize individual driver response at the expense of system response. In a multiway system, one driver may be emphasized to the detriment of another. Edge diffraction effects on cabinet-mounted drivers may not be correctly measured in the near field. A good rule of thumb is not to place the speaker and microphone closer than one wavelength of the lowest frequency for which far field characteristics are desired. Similar problems can occur with ported boxes, but the frequencies at which the port is active are typically so low that accurate far field measurements may only be made outdoors. The directivity of the drivers will also be important when the microphone is very close to a multiway speaker. It will be impossible to be on-axis of every driver in the system simultaneously. A compromise must be made and this is usually based on the expected listening position of a typical user.

D.7.3. Phase Response and Driver Polarity

Since MLS.DSP measures phase as well as amplitude, it is possible to test driver polarity and crossover wiring. The REF TIME value may be adjusted to compensate for the transit delay from speaker to microphone. When it is correctly adjusted the phase response will be relatively flat. It is also possible to use the COMPUTE LINEARITY function of System One software to determine the phase deviation from a linear phase (constant time delay) measurement. This reduces the need to adjust the REF TIME for the exact value of transit delay. The discontinuities resulting from a 360 degree phase wrap will preclude proper operation of the COMPUTE LINEARITY function so it is necessary to use the "deg" unit rather than one of the "deg1, "deg2" or "deg3" units.

The amplitude and phase response of a 3-way loudspeaker with the drivers correctly phased is shown in Figure D-11. The REF TIME value in MLS.DSP has been correctly adjusted to compensate for the transit delay from speaker to microphone as evidenced by the relatively flat phase response. The response of the same 3-way loudspeaker is shown in Figure D-12 with the midrange driver phase inverted. Note the sharp dips in the amplitude response at the woofer-to-midrange and midrange-to tweeter crossover frequencies, and the abrupt 180 degree steps in phase response at the same frequencies.

D.7.4. Subtracting Out Calibration Response with Compute Delta

The COMPUTE DELTA menu command will compute the difference between the data in memory and the data in another file. The other file is specified with the NAMES DELTA menu command. This may be used to calculate the difference between two measurement sets when wanting to study the effect of a design change. It may be helpful when comparing two different types of speaker or to match individual speakers for a stereo pair. When measuring with a microphone whose response is known to be non-flat this data may be entered into a test file for use as a COMPUTE DELTA reference.

COMPUTE DELTA subtracts the reference data file from the data currently in memory. It operates on the data in the units they are currently set to display in. Therefore, for proper operation with fre-

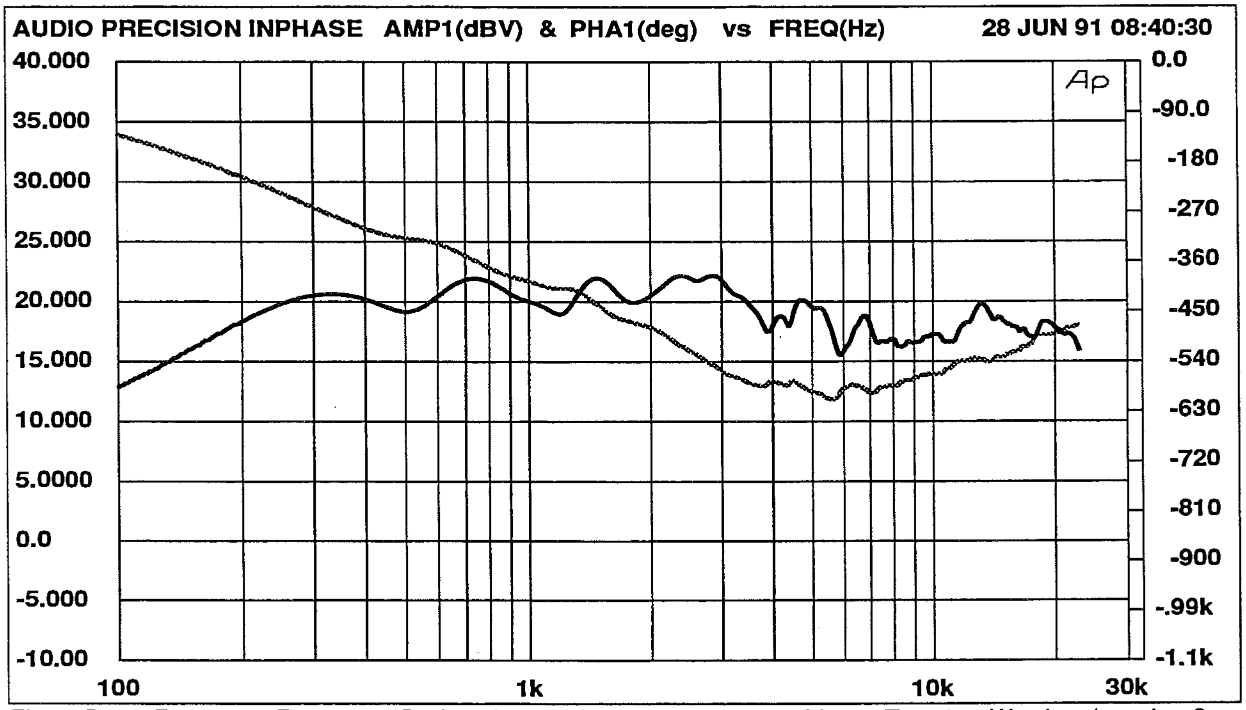

Figure D-11 Frequency Response (Darker Trace) and Phase Response (Lighter Trace), 3-Way Loudspeaker System With All Drivers Correctly Phased

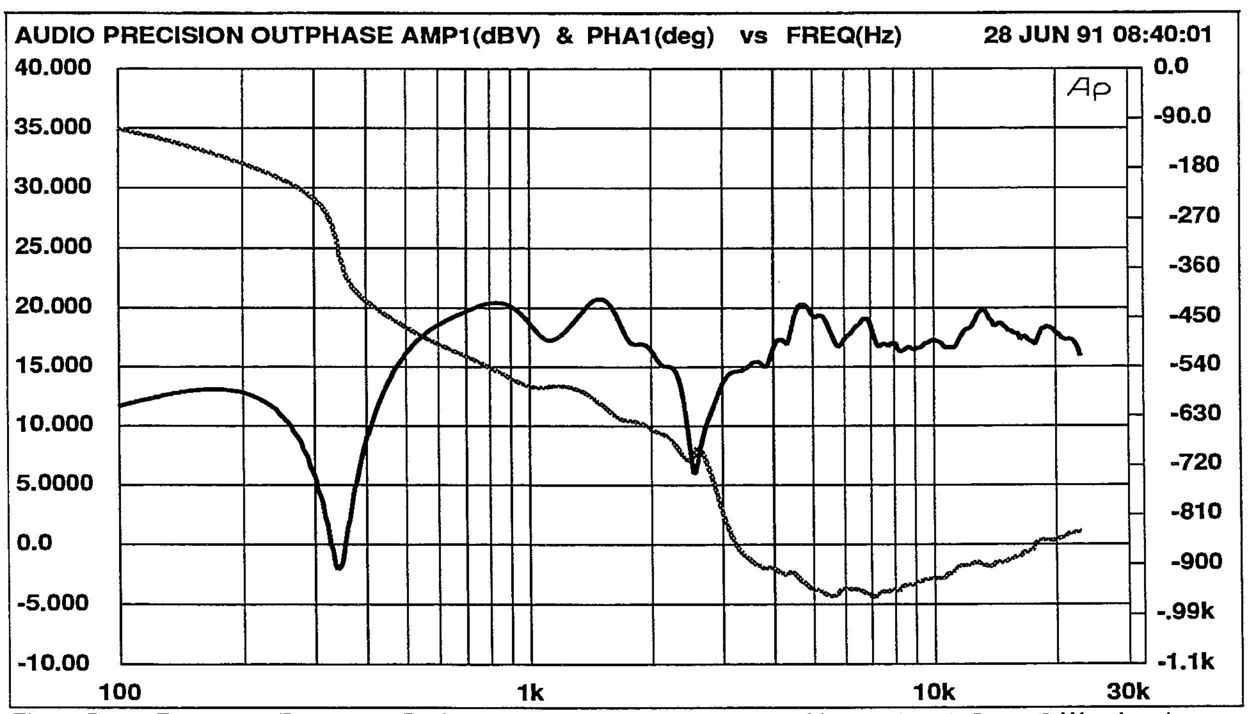

Figure D-12 Frequency Response (Darker Trace) and Phase Response (Lighter Trace), Same 3-Way Loudspeaker System as Above With Mid-Range Driver Polarity Reversed quency response data the amplitude units should be a form of dB (dBV, dBm, dBFS, etc.) or one of the degree selections. The form of the command is:

COMPUTE DELTA *data,data*

The first DATA specifies whether the DATA-1 or DATA-2 values presently in memory are to be processed. The second DATA specifies whether the DATA-1 or DATA-2 values in the file are to be subtracted from the values in memory. If the second DATA is not specified it is assumed to be the same selection as the first.

To compare two speaker measurements, load the set of data into memory by typing LOAD TEST *filename*. If it is in memory but has not yet been saved it is wise to do so (using SAVE TEST *filename*) since the data will be lost. Specify the file containing the set of data to be used as the reference by typing NAMES DELTA *filename* on the command line. Assuming the data to be compared is in DATA-1, select COMPUTE DELTA 1. Press <F7> to see the resulting difference data. If the two data sets were approximately equal the resulting data will be near 0. Therefore, for the data to be visible, the DATA-1 graph top and bottom values must include zero dB somewhere between them.

D.7.5. Microphone Measurement Using Dual Channel Operation

By using the two input channels of the System One DSP module, the response of two microphones may be compared. By using COMPUTE DELTA these two responses may be subtracted to remove the loudspeaker response from the measurement and leave the difference between the microphone responses. If the reference microphone is a known flat device, the response will be that of the microphone under test. Sensitivity and response shifts of loudspeakers with temperature and time are typically two orders of magnitude greater than those of air-condenser microphones. The simultaneous measurement technique effectively removes drifts in the loudspeaker excitation source. Gain variations in the drive amplifier will similarly not affect the results.

D.7.6. Production Test Example

Most production speaker testing is concerned with on-axis amplitude response and driver polarity. These may be very efficiently measured using MLS.DSP. Note that it is not necessary to display impulse response for each new loudspeaker tested. With proper test fixtures and operator training to assure a constant speaker-to-microphone distance as each new loudspeaker comes down the production line, the proper values of START and STOP time to select the anechoic response will be fixed. These values can be determined once and sent once from computer to DSP unit by loading a time-domain test with those values set and pressing <F6>. This must be done each time S1.EXE software is loaded or each time another DSP program is loaded and MLS.DSP is then re-loaded. Once the correct START and STOP time values have been sent to the DSP, regular production frequency response testing can then consist of using the same frequency-domain test and pressing <F9> for each new loudspeaker.

MLS.DSP offers a selection of four different 32k long Maximum Length Sequences, selected by the rightmost WAVEFORM field. Each sequence will reject the other three by 45 dB. This mutual rejection, combined with the attenuation typical between neighboring test stations, permits up to four different System One-equipped test stations to be located near one another on a production test floor with no significant interference effects between them. These four sequences have been weighted with a pink noise filter to increase their low frequency energy and provide a constant power per octave across the audio band. This greatly improves the signal-to-noise ratio at low frequencies, increasing measurement accuracy in typical room ambient noise conditions.

D.8. Advanced Theory

D.8.1. Impulse Response of Linear Systems

Any linear device or system may be completely characterized by knowing its impulse response. The impulse response is the output which results when the device under test is stimulated with an infinitely narrow pulse of infinite amplitude. In practice, if the stimulus pulse width is short compared to the length of the impulse response the measurements will still be accurate. This duration is typically 10 microseconds for audio frequency work. However, in spite of the short impulse duration, data must be acquired for the full length of time occupied by the impulse response of the device under test. This time is dictated by the lowest frequency desired in the measurement. An additional limitation which is usually more stringent is that the reverberation time of the space in which the measurements are performed must allow the signal to adequately die out before another impulse may be generated.

The amplitude of the pulse will directly affect the amplitude of the output response. Since the allowable impulse amplitude is limited by the peak handling capacity of the device under test, the signal-to-noise ratio of impulse measurements is quite poor. For measurements on electronic systems this is typically not a problem. In loudspeaker or acoustic measurements the poor signal-to-noise of impulse measurements can be a severe limitation.

The impulse amplitude is limited by the peak signal-handling capability, but the energy in the test is represented by the rms signal value. The energy directly sets the signal-to-noise ratio for any given background noise level. A single impulse has infintesimal energy content, and so has very poor signal-to-noise ratio. Because of this, most engineers working with impulse test signals average the results of many impulses to improve signal-to-noise ratio. However, the impulse repetition rate is limited by the required acquisition time and reverberation time. If the impulses are coherently averaged and the noise is not synchronous with the impulse repetition rate, the impulses will re-inforce each other and the interfering noise will tend to zero.

D.8.2. Impulse Response from Psuedo-Random Noise

Psuedo-random noise can be viewed as a random sequence of impulses, some positive and some negative. This sequence repeats at a specific rate called the repetition rate of the noise. If the device under test is linear, the response to the psuedo-random noise will be the sum of the responses to the individual impulses. When these impulses arrive to be measured, the DSP effectively shifts each one in time to align them at the same point and averages them together. As long as the interfering noise is asynchronous to the signal, the noise will average out toward zero. The averaging operation creates a single impulse response which has lower noise than any of the individual impulse responses. The signal-to-noise ratio improvement is proportional to the square root of the number of impulse responses averaged. For the 32767-point psuedo-random sequence used in the DSP, the noise improvement relative to a single impulse is 181 times or 45 dB.

In the frequency domain, psuedo-random noise has a flat spectrum, with components spaced at the repetition rate of the noise. For a 32,767 point sequence operating at a 48 kHz sample rate the psuedo-random sequence will repeat every 0.68 seconds, or a rate of 1.46 Hz. This is called white noise and has equal energy per unit of bandwidth when analyzed on a linear frequency scale. The ear hears on a logarithmic scale (in fractions of an octave) and the spectrum of most interfering noise is also flat on a logarithmic frequency scale. Consequently, white noise produces more energy than is necessary at high frequencies and, conversely, less energy at low frequencies than is desirable. For example, in the octave band from 40 Hz to 80 Hz there will be 27 frequency components of the noise. In the octave band from 10 kHz to 20 kHz there will be 6849 components, 254 times as many or a power level about 24 dB higher. To compensate for this effect, the DSP software filters the test signal with a pinking filter which attenuates the higher frequency components in direct proportion to their number. The result is an equal power in each octave band, providing a more constant signal-to-noise ratio across the measurement frequency range.

D.8.3. Hadamard Transforms

The cross-correlation operation required to shift the individual impulses in time and average them together is accomplished with a Fast Hadamard Transform.

For a description of the Fast Hadamard Transform and its application to MLS testing see Borish and Angell, "An Efficient Algorithm for Measuring the Impulse Response Using Psuedo-Random Noise", Journal of the AES, Vol. 31, No. 7, July/August 1983, pp. 478-488.

For a description of a technique to simplify computation of the Hadamard Transform see Borish, "Self-Contained Crosscorrelation Program for Maximum Length Sequences", Journal of the AES, Vol. 33, No. 11, November 1985, pp. 888-891.

D.8.4. Frequency Response From Impulse Response

The impulse response is a time domain expression of the device behavior but most people find a frequency domain expression more useful. To convert the time domain display to a frequency domain display it is merely necessary to transform the impulse response into the frequency domain using a Fourier transform. The result of a Fourier Transform on a perfect impulse is a flat spectrum in the frequency domain. If the impulse is modified by the device under test its Fourier transform will be a display of the frequency response.

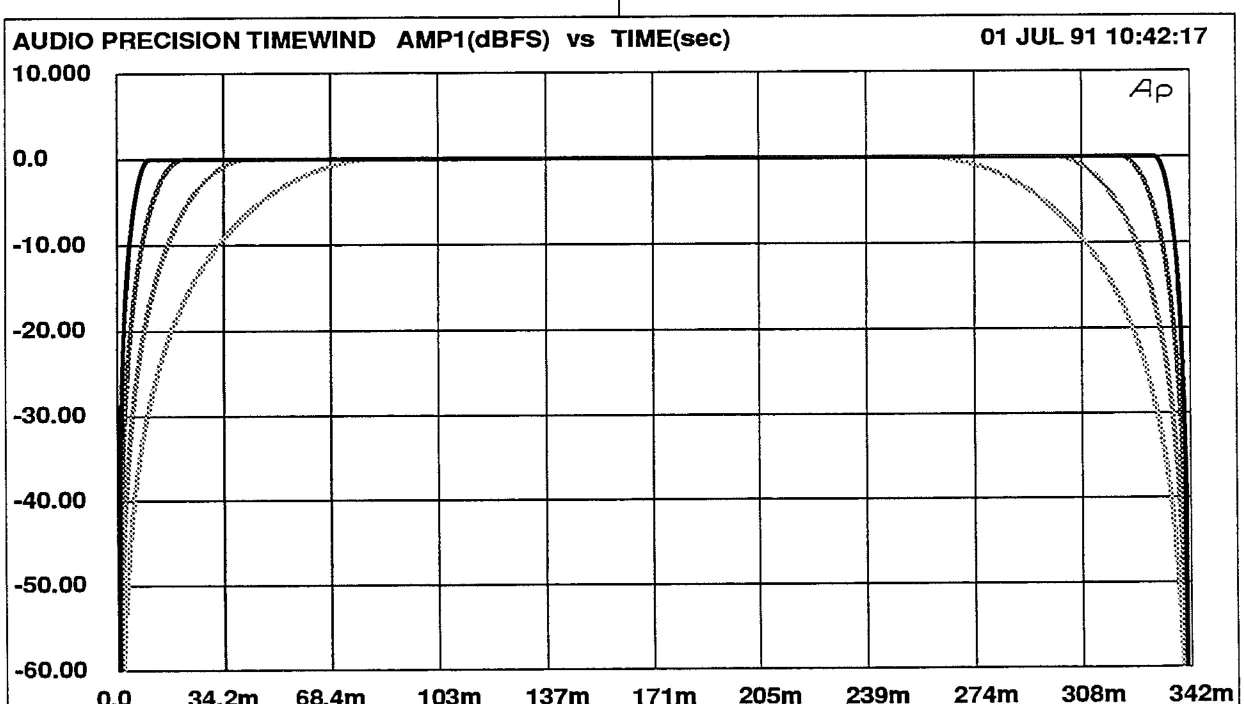

Figure D-13 Time Window Selections for Impulse-to-Frequency Transformation.. Typical Selections Are Short "Half-Window" at START, Longer "Half-Window" at STOP.

D.8.5. Time Windows for Time-to-Frequency Transforms

The FFT algorithm is used to transform a segment of the impulse response into the frequency domain in order to see the frequency response. This segment is selected from the original impulse response by setting to zero all data outside the region of interest. If signal in the data record being transformed does not naturally decay to zero at the beginning and end of the segment, there will be sharp discontinuities introduced by the selection of this segment for transforming. These discontinuities in the waveform produce large amounts of high frequency energy in the transformed result. This high frequency energy results in ripples on the displayed frequency response curve.

To alleviate this problem, a "window" may be applied to the data. The idea behind a window is to gradually taper the data at both ends of the record toward zero so that it will always make a smooth transition with the following and preceding repetitions of the record. This is accomplished by multiplying each point in the data record by a mathematical function which is near unity (1.000) in the center of the data record and small at the ends of the record. The simplest such function is an inverted cosine wave raised above zero with an added DC offset so that its negative peaks just reach zero. After multiplication by the window function the data record goes to zero at the ends and so smoothly meets each data record on either side of the one being transformed. Multiplying the data by the window function does alter the spectrum of the original signal. As might be expected by visualizing the envelope of the repeating windowed data record, the frequency response will be smoothed. However, the spurious high frequency components produced by the sharp discontinuities will have been eliminated. The raised cosine window described is called the HANN window after its inventor, Austrian meteorologist Julius von Hann. (It is often incorrectly called a Hanning window due to confusion with the "Hamming" window, named after its inventor Richard Hamming.)

The generic term window comes about because it restricts the view of the FFT to the central portion of the data record in much the same way that a window restricts the view of a person looking through it. There have been an endless variety of windows developed which trade off the spread in the spectral peak versus the ultimate attenuation of the spurious energy created by the ends of the data record. The sharper the roll-off in the skirts, the wider the peak must be in the passband. The bandwidth of the peak will be a specific number of bins for any given window. Increasing the length of the data record will reduce the bin width and therefore reduce the bandwidth in Hertz.

Transformation of impulse responses of loudspeakers is a special case, since the typical impulse has a fast rise and slow decay. Thus, it is desirable to use an asymmetrical window function which also has a fast rise and slow decay in order to taper values at the two ends of the selected portion to zero with minimal effect on the important information in the impulse. The window selection of MLS.DSP is therefore made up of two sections, each with its own panel selection field (see Figure D-3). The left-hand field on the TIME WNDW line selects among four raised half-cosine choices, each corresponding to the first 180 degrees of a cosine and thus rising from zero at the beginning of the selected portion of record to a value of 1.00 later in the record. The right-hand field on the same line selects among four raised half-cosine choices, each corresponding to the last 180 degrees of a cosine and thus falling from a value of 1.00 late in the selected portion to zero at the end of the record. The nominal time selections (<5%, <10%, <20%, <30%) refer to the percentage of the selected portion of the impulse across which the half-window makes its full transition. The "less than" sign (<) indicates that the actual percentage may be less than specified, since the actual number of samples for the transition from zero to unity (or vice-versa) will always be an exact binary power such as 4, 8, 16, etc. The DSP therefore rounds down from the selected value to the largest exact binary power within that value.

For example, assume a START time of 4 ms and a STOP time of 8 ms, resulting in a 4 ms time span. If the 48 kHz sample rate (20.8 microsecond sample

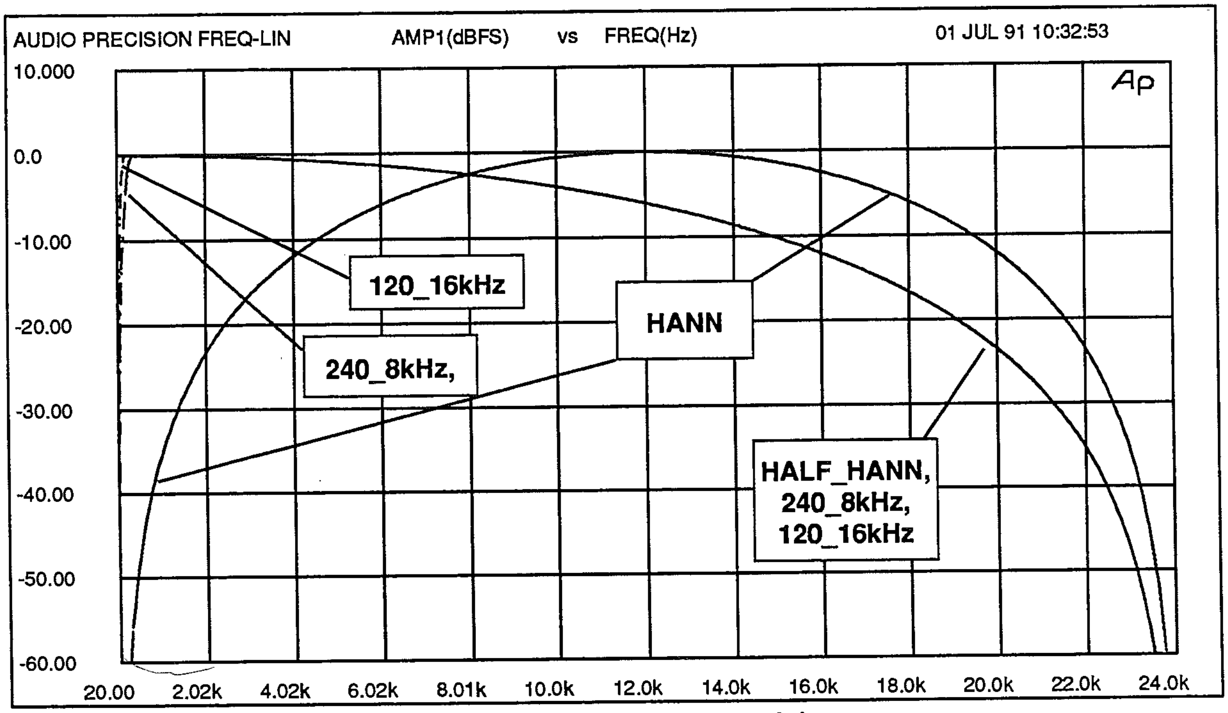

Figure D-14 Frequency Windows for ETim Curves, Linear Frequency Axis

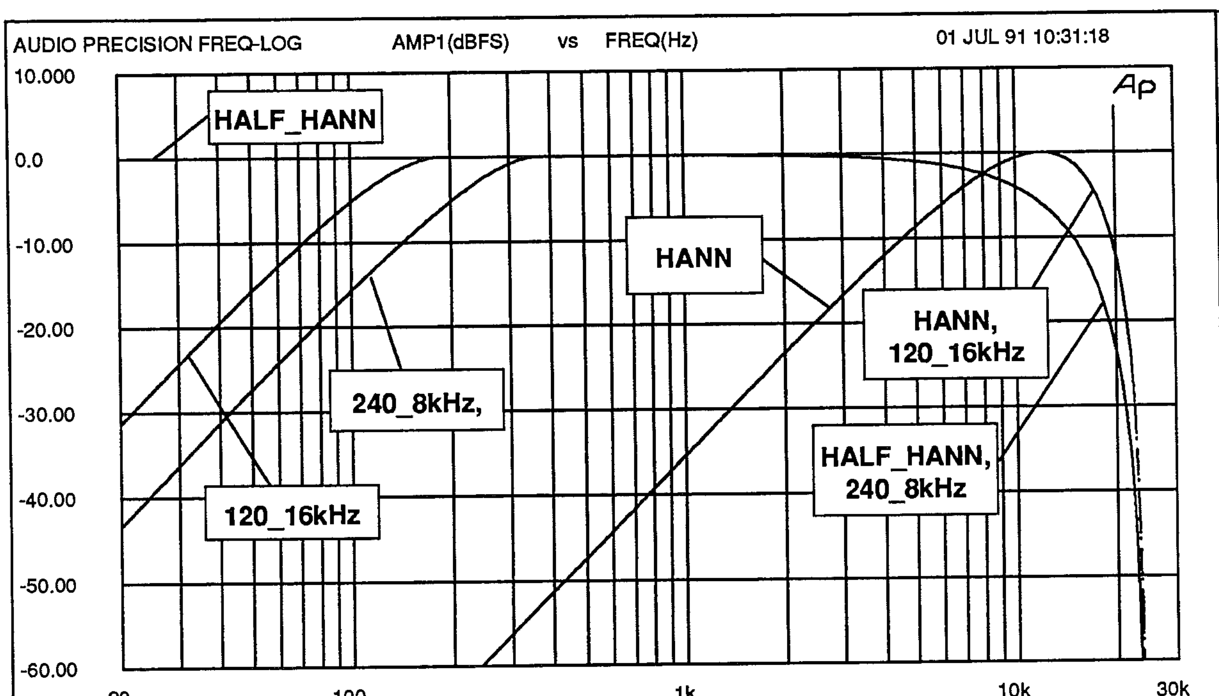

Figure D-15 Same Frequency Windows as Previous Figure, Logarithmic Frequency Axis period) is in use, there will be approximately 192 samples in the selected span. A START selection of 5% would nominally make the transition in the first 5% of the 192 samples, or 9.6 samples; the actual transition time will be rounded down to 8 samples as an exact binary power. Similarly, if 30% is chosen for the STOP selection, the nominal transition would take place during the final 30% of the 192 samples, or 57.6 samples. The actual transition time will be rounded down to 32 samples, the next lower exact binary power. All the data between the 8th sample from the beginning and the 32nd sample from the end will be unattenuated. See Figure D-13 for a graphical illustration of the four selections available both at the START and STOP ends of the record to be transformed.

For an excellent technical discussion of windows and their characteristics see F. J. Harris, "On the use of Windows for Harmonic Analysis with the Discrete Fourier Transform", Proc. of the IEEE, Vol. 66, No. 1, Jan 1978, pp. 51-83.

D.8.6. Frequency Windows for ETim Displays

The ETim selection causes the DSP to transform the impulse response into the frequency domain, apply a frequency-domain window and Hilbert Transform to the complex frequency response, and then inverse transform the data back to the time domain to produce a plot of the estimated arrival of energy versus time. This "energy-time curve" is an approximation to the actual energy arriving at the microphone, since this energy can never be known without the simultaneous measurement of the velocity and pressure of the sound. Since the Energy-Time curve is computed from the pressure response alone it can never be complete.

The left TIME SWP field selects the window function applied to the frequency-domain data when computing energy-time curves. The available window choices are no_window, Half_Hann, Hann, 240_8kHz, and 120_16kHz. The Hann window, although used on other measurement equipment, yields inaccurate results because it excludes behav-

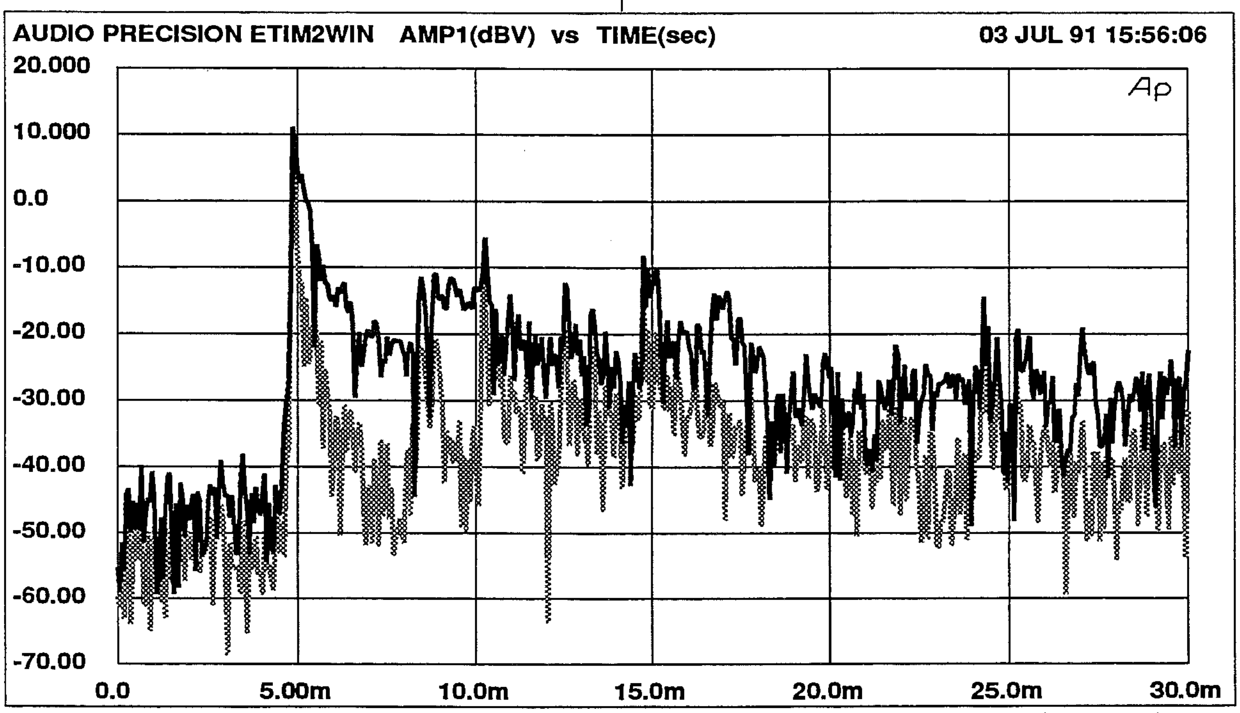

Figure D-16 Energy-Time Curves, Same Impulse Response with Hann Window (Lower, Lighter Trace) and 120_16kHz Window (Upper, Darker Trace). Note Hann Window Favors High Frequencies ior of the majority of most device's frequency range. The Half_Hann, 240_8kHz, and 120_16kHz are preferable for most applications. See Figure D-14 for illustrations of these frequency window choices plotted on a linear frequency scale, and Figure D-15 for the same windows plotted on a log frequency scale.

The first selection turns off windowing, allowing use of all frequency components of the arriving sound in the energy-time computation. The Half_Hann selection uses a raised half-cycle cosine window which does not attenuate low frequencies but attenuates high frequencies above one quarter of the sample rate (approximately 12 kHz). The Hann selection uses a standard raised cosine window as provided on measurement equipment from other manufacturers. It centers its analysis on the energy around one quarter of the sample rate (approximately 12 kHz) attenuating both low and high frequencies. This yields inaccurate results on most devices because it excludes the behavior of the vast majority of the device's frequency range.

The last two window selections have been developed by Audio Precision to reduce the inaccuracies of the Hann window at both low and high frequencies while still reducing the window ripples which would occur if no window was used. These windows gradually roll off the spectral data above and below the frequency values specified in their labels. The 240_8kHz window focuses the analysis between 240 Hz and 8 kHz by applying asymmetrical half-cycle raised cosine windows which attenuate below 240 Hz and above 8 kHz. The 120_16kHz window shifts these frequencies by an additional octave to improve measurements of wideband devices at the expense of slightly more window rippling.

Energy time curves from the same impulse response using the 120_16kHz (darker trace) and Hann (lighter trace) windows are shown in Figure D-16. Note the much more rapid fall-off of energy with the Hann window after the main impulse and each reflection, indicating that only very high-frequency energy (centered at 12 kHz) is being used in the computation. The 120_16kHz window is much more appropriate for full-range audio-frequency analysis.

For an excellent technical discussion of the effect of windows on energy-time-curve computation see Lipshitz and Vanderkooy, "Uses and Abuses of the Energy-Time Curve", Journal of the AES, Vol. 38, No. 11, November 1990, pp. 819-836.

D.9. Furnished Files

Two procedures, two stored impulse response waveforms, and a number of tests are furnished on the same diskette with MLS.DSP.

MLS-DEMO.PRO is a self-contained training demonstration of the MLS program, using stored impulse response waveforms. Thus, no loudspeaker or microphone is required with this procedure. It loads an impulse response (time domain) test with MLS.DSP attached, then downloads THREEWAY.WAV from disk. THREEWAY.WAV is the stored impulse response of a good quality three-way loudspeaker (woofer, mid-range, and tweeter) taken in a typical room with nine-foot (three meter) ceilings and the measurement microphone about five feet from the loudspeaker. The procedure prompts you to use the graphic cursors to locate the start of the impulse build-up (about 4.2 milliseconds for this impulse response) and the start of arrival of the first reflection (about 8.1 milliseconds for this stored impulse response). This expanded first-arrival signal is then regraphed (<F6>) in order to send the start and stop time information to the DSP module. Then, another test is loaded with frequency on the horizontal axis and both magnitude and phase displayed vertically, and the impulse is transformed (<F6>) and displayed.

MLS-DEMO.PRO ends at this point, but you may experiment with going back and forth between time domain and frequency domain tests, and selecting different portions of the impulse to be transformed. Use the <F6> key each time you load another test or change any panel setting. The impulse remains in DSP memory and will be repeatedly retransformed at each <F6> operation. It will also be instructive to LOAD WAVEFORM MID-INV.WAV and view it in the frequency domain. MID-INV.WAV was acquired with the same loudspeaker and same microphone location, but with the polarity of the mid-range driver inverted. The response curve dips and phase curve steps should be clearly visible.

MLS.PRO is set up for actual loudspeaker testing, and assumes a combination of sound pressure level, microphone sensitivity, and microphone amplification which produce approximately a one Volt magnitude impulse. The procedure prompts the user to determine the start time of first arrival, the start time of the first reflection, and the time of peak impulse and to enter these into the appropriate fields. Frequency and phase response is then displayed.

In addition to the log-vertical impulse display used in the two procedures, .TST files are furnished for energy-time curve display (ETIM.TST) and bipolar linear display of the impulse (IMPLSLIN.TST). Frequency-domain tests are furnished for both the typical usable anechoic portion of the spectrum above 300 Hz (RESP.TST and RESPHASE.TST, without and with phase displays) and for a full-range 20 Hz-20 kHz frequency response when a longer time span is analyzed (RESP-WR.TST and RESPHAWR.TST). Remember that the lowest frequency for accurate results is not determined by the SOURCE-1 START frequency of the response test currently in use, however, but by the time span between SOURCE-1 START and STOP times selected when <F6> was last pressed during a time domain display.

The detailed list of MLS files furnished is as follows:

1ST-3WAY.TST impulse response, start & stop times set for 1st arrival signal of THREEWAY.WAV

ETIM.TST ETim curve, first 50 ms of signal

IMPLSLIN.TST linear bipolar display of impulse, first 50 ms of signal

IMPLSLOG.TST log Peak display of impulse, first 50 ms of signal

MID-INV.WAV impulse response of NHT 3-way with mid-range driver out-of-polarity (was FRMINV.WAV)

MLS-DEMO.PRO demo procedure with prompts to find start and stop of 1starrival using THREEWAY.WAV, plots response & phase

MLS.DSP new program

MLS.PRO procedure to acquire signal, prompt operator to locate and enter start and stop time and peak impulse time, plots response & phase

RESP-WR.TST frequency response assuming long time span, 20 Hz-20 kHz

RESP.TST frequency response for typical anechoic portion, 300 Hz-20 kHz

RESPHASE.TST frequency and phase response, 300 Hz-20 kHz

RESPHAWR.TST frequency and phase response, 20 Hz-20 kHz

THREEWAY.WAV impulse response of NHT 3-way, normal polarity (was FR.WAV)

E. HARMONIC ANALYSIS PROGRAM HARMONIC.DSP

HARMONIC.DSP is a program for selective real-time measurement of the amplitude of audio signals. It consists essentially of a DSP-implemented bandpass filter followed by a DSP-implemented RMS detector. Since the program is real-time, its measurements may be observed in panel mode, bar-graph (F2) mode, and plotted as a DATA-1 or DATA-2 parameter onto an X-Y graph while some parameter is swept as SOURCE-1.

It may be used to measure individual harmonic distortion products, essentially unaffected by wideband noise. This technique contrasts with the THD+N (total harmonic distortion plus noise) function of System One's analog analyzer, which measures amplitude across the full audio spectrum following removal of the fundamental signal, and is therefore sensitive to noise in addition to distortion.

The expression "loading" a DSP program is used frequently throughout this manual. In fact, NAMES PROGRAM is the specific command required to download a .DSP program from computer disk to DSP unit. When a .TST file is saved to disk after using the NAMES PROGRAM command, the DSP program will also be automatically downloaded each time the .TST file is loaded thereafter (unless the DSP program is already in place from the previous test).

HARMONIC.DSP may also be used for swept spectrum analysis, measurement of specific intermodulation distortion products, depth of erasure measurements on analog tape recorders, crosstalk below noise level, and other "wave analyzer" or "selective voltmeter" applications.

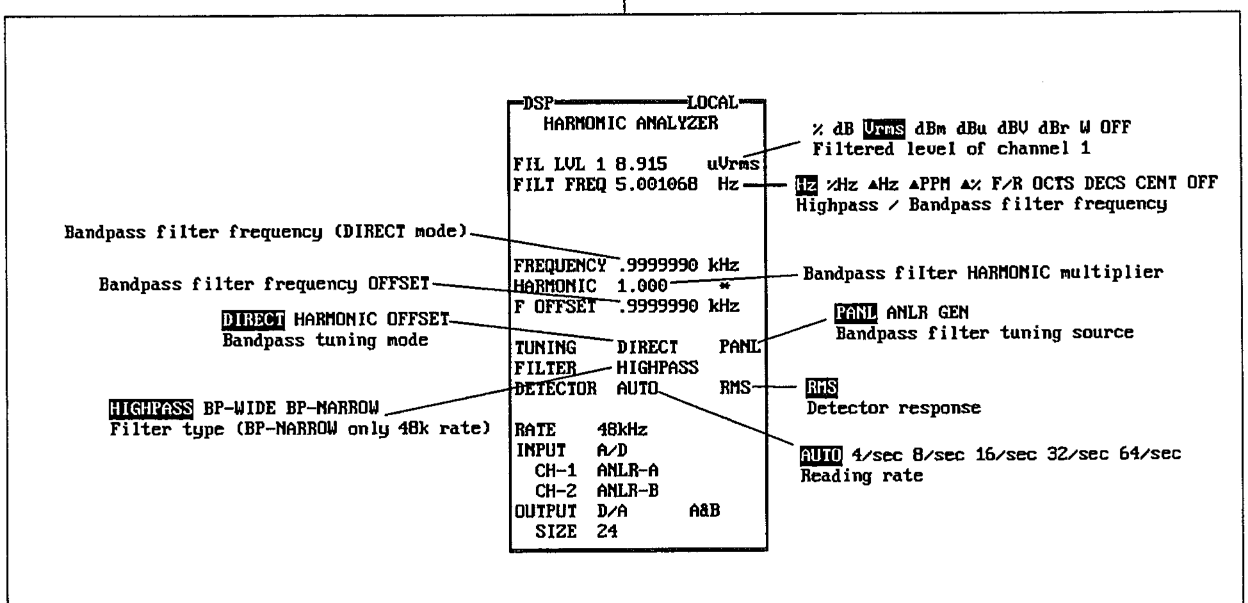

Figure E-1 DSP Panel with HARMONIC.DSP Loaded

Figure E-1 shows the DSP panel of S1.EXE software after the Names Program menu command has been used to select HARMONIC.DSP. The "exploded" call-outs show the choices available at the various selection fields. The FIL LVL 1 field at the top of the panel is a real-time numeric display of the amplitude of the Channel 1 input signal as modified by the DSP filter response. Figure E-2 shows the DSP HELP panel with HARMONIC.DSP loaded.

E.1. Filter Shape

HARMONIC.DSP may be operated as a bandpass filter or in a flat response mode. The flat mode, selected as HIGHPASS on the FILTER line, is intended as a real-time broadband amplitude measurement technique with the DSP module. This mode is useful to verify the presence of an input signal and to provide a quick check of levels. The purpose of the highpass filter in HIGHPASS mode is to reject any DC voltage present in the input signal. The turnover frequency of the highpass filter varies with the digital sample rate and is indicated in the FILT FREQ field at the top of the panel. Bandwidth extends to 80 kHz with the 192 kHz sample rate, with proportionately less bandwidth at 48 kHz. See Figure A-7 on page A-8 for a set of response measurements for each of the available sample rates.

Two widths of bandpass filter are obtainable at 48 kHz sample rate. These filters are selectable as BP-WIDE and BP-NARROW on the FILTER line. Figure E-3 shows typical frequency response measurement of both bandpass filter widths of HARMONIC.DSP. For comparison, the response of System One's analog bandpass filter is also shown. Note that only the BP-WIDE shape is available at the 192 kHz sample rate, even if BP-NARROW has been selected.

Figure E-4 is an expansion of the response of the two DSP filters and System One's analog bandpass filter over the top few dB. The HARMONIC.DSP bandpass filters are constant Q devices and thus have bandwidth that is essentially a constant percentage of center frequency. BP-WIDE has a Q (ratio

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═══════════════════════ DSP HELP SCREEN ═══════════════════════

This program measures the amplitude of analog input signals.  It is not
intended to handle digital signals.  The measured signal may be either
highpass or bandpass filtered.  If highpass filtered, the cutoff frequency is
dependent on sample rate and is not adjustable.  In bandpass filter mode the
the frequency may be controlled by the panel setting, it may track the analog
distortion analyzer bandpass / bandreject frequency, or it may track the analog
generator frequency.  The tracking may be at the same frequency, at any
multiple of the selected frequency up to 9, or at any offset above or below the
selected frequency.  The narrow filter does not function above 48 kHz sample
rate.  The program will default to BP-WIDE in this case.

Ratio units such as dB and % may only be used when channel 1 is driven from the
READING meter.  When measuring from the other sources use absolute units such
as Volts, dBm, W, etc.

The reading rates are only automatically selected when in bandpass mode.  The
highpass amplitude display mode will always default to 4/sec in auto.

The filtered level readings use the amplitude field on the settling panel.  The
filter frequency display uses the frequency field on the settling panel.
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Figure E-2 DSP HELP Panel with HARMONIC.DSP Loaded

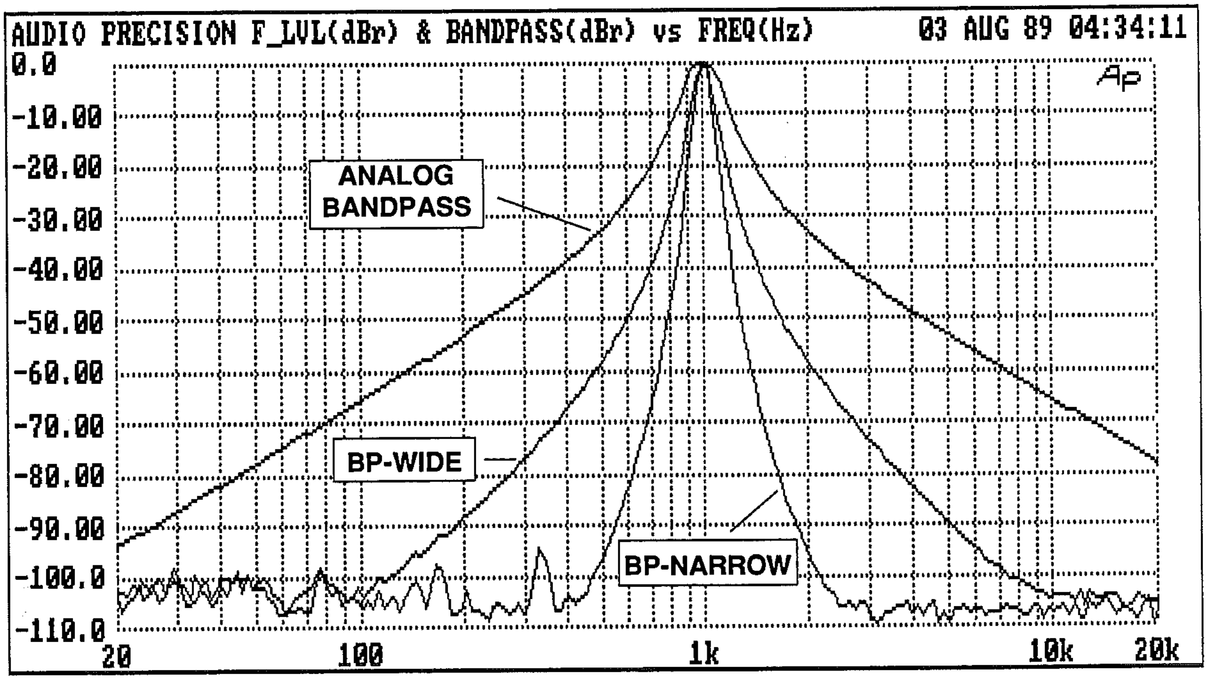

Figure E-3 Response Curves, Bandpass Filters of HARMONIC.DSP (48 kHz Sample Rate) and Analog Analyzer at 1 kHz.

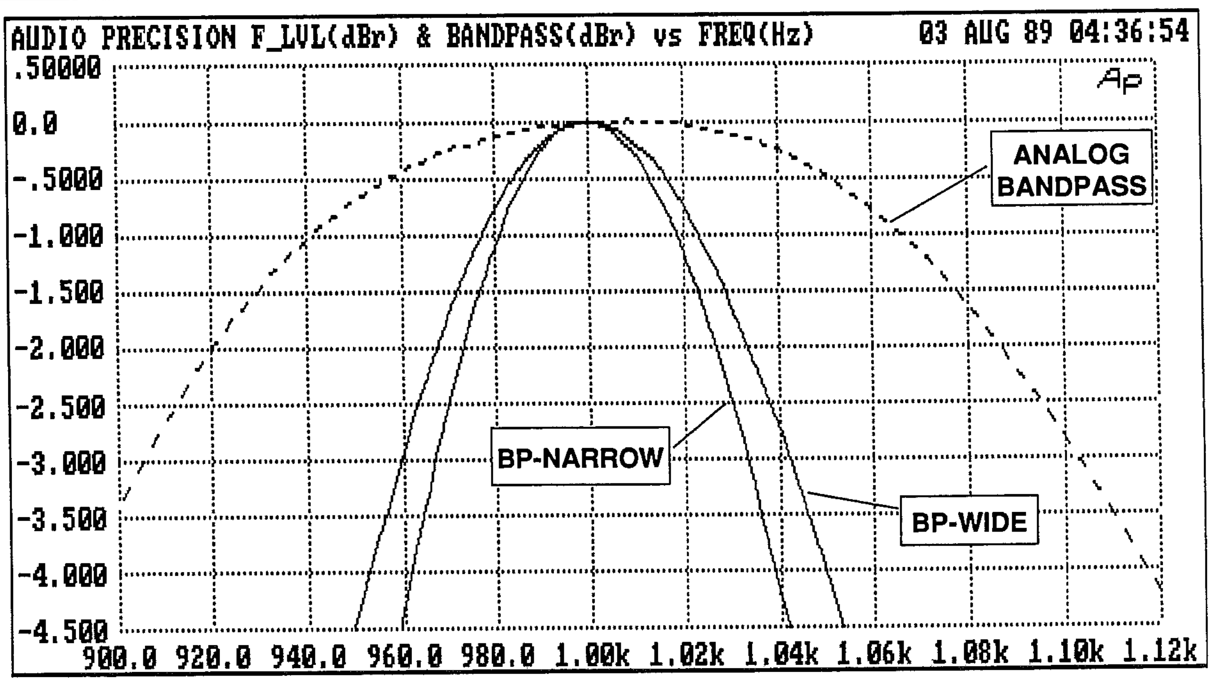

Figure E-4 Expanded Response Curves, HARMONIC.DSP Filters and Analog Analyzer BANDPASS Filter

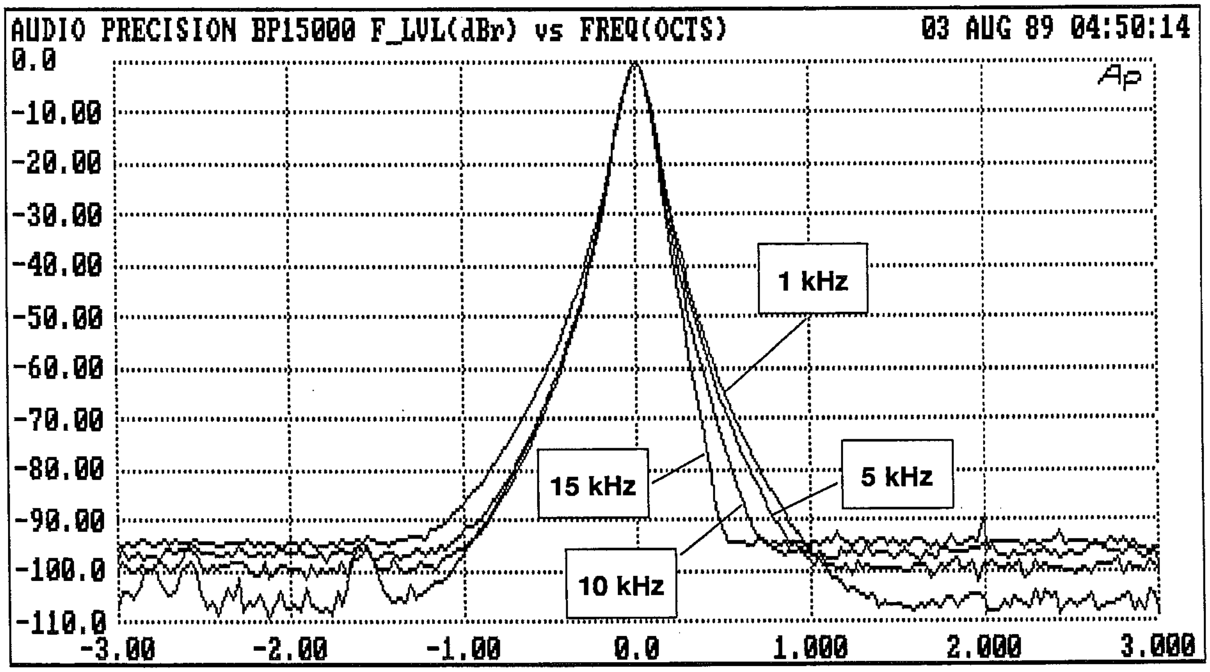

Figure E-5 Response Curves, BP-NARROW Selection of HARMONIC.DSP at Center Frequencies of 1 kHz, 5 kHz, 10 kHz, and 15 kHz at 48 kHz Sample Rate. Horizontal Axis is Frequency in Octaves Relative to Center Frequency

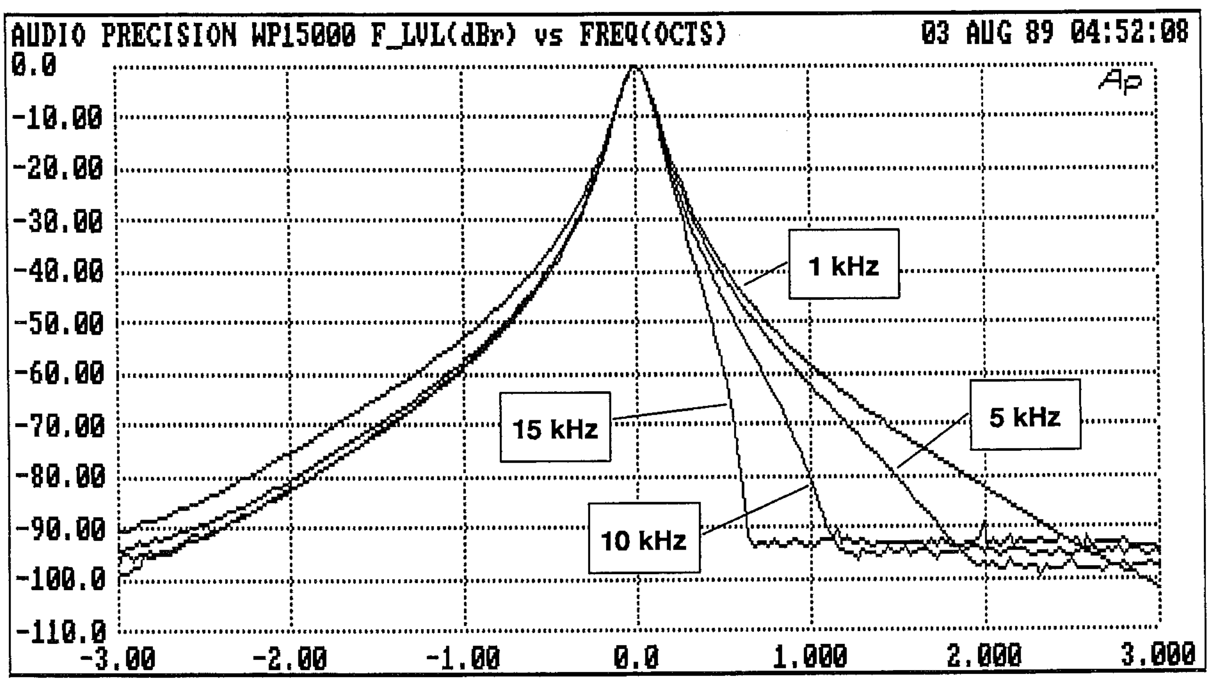

Figure E-6 Response Curves, BP-WIDE Selection of HARMONIC.DSP at Center Frequencies of 1 kHz, 5 kHz, 10 kHz, and 15 kHz at 48 kHz Sample Rate. Horizontal Axis is Frequency in Octaves Relative to Center Frequency of center frequency to -3 dB bandwidth) of approximately 12. The bandwidth is thus approximately 1/8 octave. BP-NARROW has a Q of about 15, or bandwidth of about 1/10 octave.

As the filter center frequency approaches one-half the selected digital sample rate, the shape of the filter changes. Figure E-5 shows four response curves of BP-NARROW for center frequencies of 1 kHz, 5 kHz, 10 kHz, and 15 kHz, all with the 48 kHz sample rate. They are overlaid on a horizontal axis in relative frequency units referred to the center frequency. It can be seen that the bandpass shape which is symmetric at low and moderate frequencies becomes unsymmetrical as the upper frequency limit is approached. Similarly, Figure E-6 shows BP-WIDE for the same four center frequencies. The change in symmetry is more pronounced on the wider filter.

E.2. Filter Tuning

The bandpass filter frequency is determined by the selections of tuning mode (DIRECT, HARMONIC, OFFSET), tuning source (PANL, ANLR, GEN), and numbers entered into the FREQUENCY, HARMONIC, and F OFFSET fields. The actual filter frequency resulting from the selections and numbers in these three fields is indicated in the FILT FREQ field near the top of the DSP panel. The absolute limits on filter center frequency are 80 Hz to 80 kHz with the 192 kHz sample rate and 20 Hz to 20 kHz with the 48 kHz rate.

There are three choices of TUNING SOURCE: PANL (a panel-entered fixed frequency), GEN (tracking the analog generator frequency), or ANLR (tracking the analog analyzer bandpass filter frequency). The analog analyzer bandpass filter frequency, in turn, may be steered in panel mode and during SOURCE-1 EXTERNAL FREQ sweeps by the frequency measured by the analog analyzer frequency counter. Thus, the ANLR selection of the HARMONIC.DSP program will cause the digital bandpass filter to be steered by the signal frequency present at the analog analyzer input in EXTERNAL FREQ sweep mode.

The actual bandpass filter frequency will be further determined by the TUNING mode selected (DIRECT, HARMONIC, or OFFSET). DIRECT mode uses the TUNING SOURCE frequency directly as the filter frequency. To sweep the digital bandpass filter across a frequency range, select PANL as the TUNING SOURCE, DIRECT as the TUNING method, and DSP FREQ at SOURCE-1 with the desired START and STOP frequencies (as limited by the sampling rate).

HARMONIC tuning mode passes an integral multiple of the TUNING SOURCE frequency, where the multiple may be entered in the HARMONIC multiplier field above F OFFSET. Harmonic numbers 1 through 9 may be entered. HARMONIC 1 is identical to DIRECT mode. Second harmonic analysis, for example, is chosen by HARMONIC TUNING mode with 2 entered as the HARMONIC multiplier above. Using GEN as the TUNING SOURCE and selecting a GEN FREQ sweep at SOURCE-1 would then permit a swept analysis of the second harmonic distortion of a unit under test. If the analog generator frequency is set to 1.000 kHz, the FILT FREQ field will indicate 2.000 kHz with GEN tuning and HARMONIC 2. Third harmonic distortion of the reproduce section of an analog tape recorder can be measured by selecting ANLR as the TUNING SOURCE, HARMONIC 3, selecting SOURCE-1 EXTERNAL FREQ, plotting DSP F_LVL at DATA-1, and playing a reference tape with a series of fixed frequencies at reference fluxivity level. Note that the 192 kHz sample rate is typically required in many swept harmonic distortion applications, since the 20 kHz upper frequency limitation with the 48 kHz sample rate is reached at moderate fundamental frequencies when a high HARMONIC order is specified.

OFFSET tuning mode permits measurement of a single intermodulation distortion product when the higher frequency tone of a SMPTE-like IMD signal is swept while the lower frequency tone is fixed. For example, with System One's IMD option SMPTE waveform selection chosen and a 500 Hz IMD frequency selected on the GENERATOR panel, the signal consists of a 500 Hz signal plus the main oscillator frequency. Non-linearity in the device under test will produce sidebands spaced 500

Hz above and below the main oscillator frequency. With OFFSET set to either +500 Hz or -500 Hz and GEN as TUNING SOURCE, the harmonic analyzer program will measure the amplitude of the selected IMD product, even as the main oscillator frequency is swept across a range above 2.5 kHz. If GEN is the TUNING SOURCE and F OFFSET is -500 Hz, the FILT FREQ field near the top of the panel will indicate 7.500 kHz when the analog generator is set to 8.000 kHz. Again, note that positive offset values place the filter frequency above the tuning source value, which will require selection of the 192 kHz sample rate for large values of OFFSET and/or as the tuning source frequency approaches 20 kHz.

E.3. Detector and Reading Rate

The detection portion of HARMONIC.DSP is always true RMS. The update rate may be selected from AUTO or fixed rates from 4/second to 64/second. The slower rates are required for accurate measurements at lower signal frequencies. The AUTO selection is designed to select the fastest rate which will give accurate results at the specific bandpass filter frequency currently in use. Thus, during a wideband frequency sweep, the reading and plotting rate will be noticeably slower at low frequencies. In HIGHPASS mode, the AUTO selection will produce 4 readings per second.

E.4. Input and Output Signal Selections

The DGEN selection of the INPUT line is not useful with HARMONIC.DSP since the program contains no digital signal generation capability. Likewise, the bottom two (OUTPUT) lines on the DSP panel have no effect. All analog inputs function as described in the DSP Introduction chapter, as do digital inputs in a Dual Domain unit. However, HARMONIC.DSP is not designed for making measurements on digital signals and no appropriate digital units of measure are provided.

E.5. Making Swept Tests

As is customary with S1.EXE software, most variable parameters can be swept and most measured values can be graphed during a test.

E.5.1. Sweepable Parameters

With HARMONIC.DSP loaded, the SOURCE-1 DSP selections are FREQ, HARM, and F_OFF. FREQ permits directly sweeping the bandpass filter when PANL is selected as the TUNING SOURCE. If a HARMONIC number above 1 is selected or if OFFSET mode is selected with a non-zero F OFFSET value, the actual filter frequency will be a multiple of or offset from the SOURCE-1 values.

Selecting HARM at SOURCE-1 permits sweeping the harmonic number in HARMONIC TUNING mode. For example, with a 1 kHz square wave signal and the 192 kHz sample rate, DSP HARM with a START value of 1 and STOP value of 9 will result in measurement of the fundamental and 2nd through 9th harmonics of the squarewave. The horizontal axis calibration will be harmonic number.

The F_OFF parameter permits sweeping the F OFFSET parameter in OFFSET mode of the program. For example, assume a DIN IMD test signal (SMPTE waveform mode of the generator) consisting of 500 Hz and 8 kHz signals. Select the SMPTE IMD analysis function of the analog analyzer, and select RDNG as the DSP CH-1 source. With OFFSET tuning mode, GEN tuning source, and a SOURCE-1 selection of DSP F_OFF with 500 Hz START, 2 kHz STOP, LIN, and # STEPS 3, the graph will consist of selective amplitude measurements at 500 Hz, 1 kHz, 1.5 kHz, and 2 kHz. These are the frequencies of 2nd order, 3rd order, 4th order, and 5th order IMD products.

E.5.2. Plotting Measurements

With HARMONIC.DSP, only one real-time measurement exists to be plotted. Select F_LVL at DATA-1 or DATA-2 in order to plot the amplitude measured by the program onto a graph. The units may be selected among %, dB, Vrms, dBm, dBu, dBV, dBr, and W. See the UNITS chapter of the System One User's Manual for definition of all these units except % and dB. Note that the references used for the dBm and dBr units are entered near the bottom of the ANALYZER panel.

The % and dB units are relative, and are defined only when RDNG (analog analyzer READING meter) is the source of signal. The references for the % and dB units depend upon the function selected for the READING meter. If the READING meter is in BANDPASS, THD+N, 2-CHANNEL, or CROSSTALK functions, the analog LEVEL voltmeter signal serves as the % and dB unit reference . With RDNG as the DSP input selection and W&F or one of the three intermodulation distortion functions (SMPTE, CCIF, or DIM), the % and dB unit reference is the defined reference for the particular measurement selection. This permits measurement of the intermodulation distortion products, calibrated directly according to the definition. When RDNG is the DSP input and the reading meter is in AMPLITUDE function or BANDREJECT function, the reading meter value is the % and dB unit reference.

The DSP bandpass filter center frequency, though not literally a measurement, can also be plotted at DATA-1 or DATA-2 by the F_FRQ selection. This may be convenient when either HARMONIC or OFFSET tuning modes are used, since the filter frequency is not then at the principal stimulus frequency.

E.6. Typical Applications

The DSP program diskette contains a number of sample tests, one of which uses HARMONIC.DSP. This test is K3VSFRQ.TST, to measure third harmonic distortion versus frequency on a three-head analog tape recorder. The generator sweeps across the audio band to 15 kHz. The analog analyzer in THD+N mode removes the fundamental frequency component. The DSP analyzer filter automatically tracks at three times the analog filter frequency, and this filter output is plotted. The 192 kHz sample rate is selected since the DSP filter must tune to 45 kHz at the maximum generator frequency. Both stereo channels are swept sequentially. A SETTLING DELAY of 90 milliseconds is stored in the setup, typical for many professional three-head tape recorders operating at 15 inches per second. This DELAY value must be at least the time delay between record and play heads at the selected speed. The test DELAYDUT.TST, stored on the diskette and described in the FFT chapter, can be used to measure the actual delay through any particular tape machine.

E.7. Furnished Files

HARMONIC.DSP version 2.10 revised program

K3VSFRQ.TST test file to measure 3rd harmonic distortion vs frequency on stereo device such as 3-head analog tape recorder

F. ADDENDUM—NEW GENANLR.DSP FEATURE

A new control field has been added at the right end of the DETECTOR line on the GENANLR.DSP panel to provide better user control over reading rate of the DSP meters.

The FREQ 1 meter reading rate is always slaved to the LEVEL 1 meter reading rate. When a fixed reading rate is selected in the first field on the DETECTOR line, the LEVEL 1 and FILT LVL2 meters update at the selected rate. If AUTO is selected, the update rate of the two meters depends upon several factors:

When the software panels are displayed, the LEVEL 1 reading rate is 4/second.

In bargraph (<F2>) display mode, the LEVEL 1 reading rate is 8/second.

If the BP-NARROW or BANDREJ filter is selected, the FILT LVL2 meter reading rate is determined by the frequency of the DSP BP/BR filter. The fastest reading rate which will produce readings of specified accuracy at the filter frequency is automatically selected.

If the "A"WTG or CCIR filter is selected, the FILT LVL2 meter reading rate is 4/second.

If none of these filters is selected, the FILT LVL2 reading rate is slaved to the LEVEL 1 reading rate, which in turn is controlled as described below:

During sweeps, the AUTO selection causes the reading rate to vary with frequency. The LEVEL 1 reading rate will be set to the fastest value which

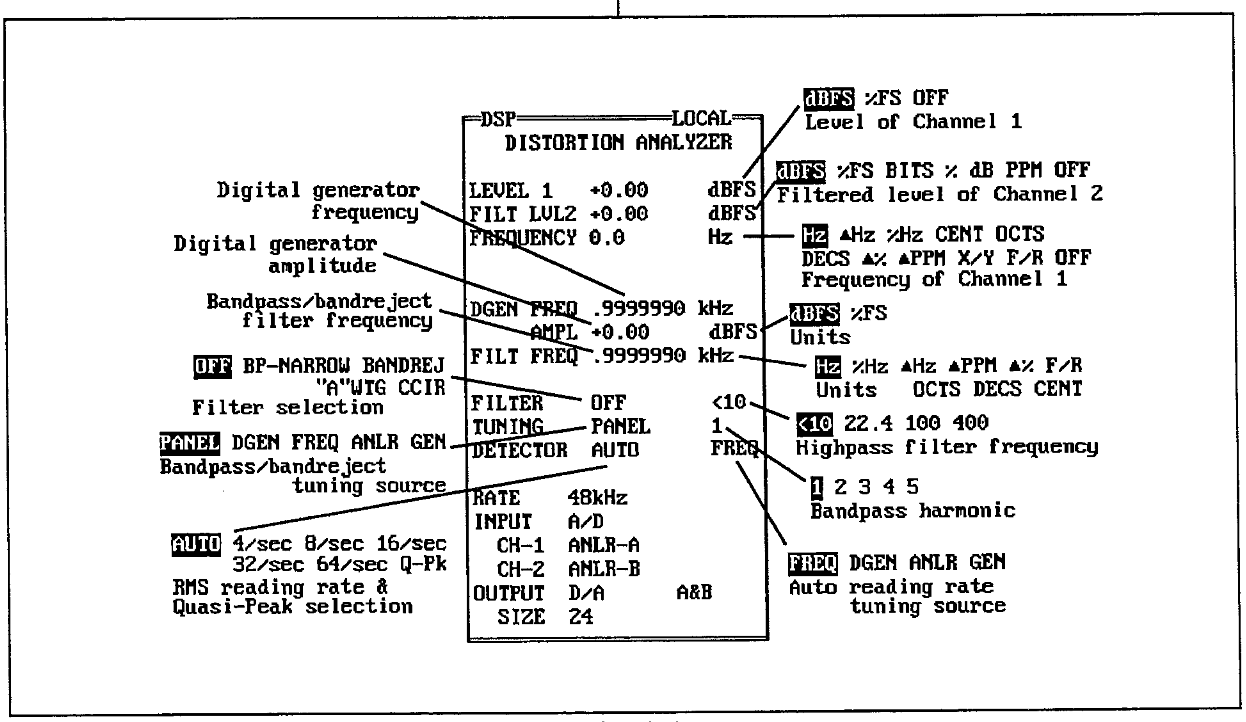

Figure F-1 DSP Panel With GENANLR.DSP Program Loaded will produce readings of specified accuracy at the present frequency of the device selected in the field at the right end of the DETECTOR line:

When FREQ is selected, the GENANLR FREQ 1 meter reading is the source of reading rate determination. This was the standard mode of earlier versions of GENANLR.DSP. The FREQ rate steering source will normally be selected for measurement of external digital signals such as from digital recordings. It should not be used if the amplitude of the digital signal is below about -60 dBFS or if the signal is noisy, since the FREQ 1 readings will then be unstable or missing.

When DGEN is selected, the GENANLR sinewave generator frequency is the source of reading rate determination. The DGEN selection should be used during GENANLR-generator-based sweeps, such as when measuring an external digital processor.

When ANLR is selected, the analog analyzer bandpass/bandreject filter frequency is the source of reading rate determination. The ANLR source should be used when measuring external signals which are available and connected in both digital and analog domains, if the analog domain signal has a cleaner or more reliable frequency measurement. The frequency counter in the analog analyzer normally steers the frequency of the analog BP/BR filter.

When GEN is selected, the analog generator frequency is the source of reading rate determination. The GEN source should be used during analog generator-based sweeps, typical when measuring A/D converters.

When a ratio unit (dB or %) is selected for FILT LVL2, the reading rates of both the LEVEL 1 and FILT LVL2 meter will be locked together at the slower of the rates for either meter, determined as described above.

F. DIGITAL ANALYSIS AND GENERATION PROGRAM GENANLR.DSP

F.1. Introduction

GENANLR.DSP is a general-purpose digital audio sinewave generation and measurement program. It is useful only with the System One Dual Domain (SYS-300 series) models. It emulates in DSP software most of the common analog measurement techniques for audio signals. Thus, it permits measurements on audio signals in the digital domain which can be readily compared to conventional analog audio measurements of other devices. GENANLR.DSP is not intended for measurements of analog-domain signals. Combined with the analog generation and measurement capabilities of System One, GENANLR.DSP permits measurement of A/D converters, D/A converters, and purely digital or purely analog audio products. With this program, it is possible to definitively measure the performance of converters and other digital audio devices without obscuring the desired parameters by an unnecessary conversion to or from the analog domain.

> The expression "loading" a DSP program is used frequently throughout this manual. In fact, NAMES PROGRAM is the specific command required to download a .DSP program from computer disk to DSP unit. When a .TST file is saved to disk after using the NAMES PROGRAM command, the DSP program will also be automatically downloaded each time the .TST file is loaded thereafter (unless the DSP program is already in place from the previous test).

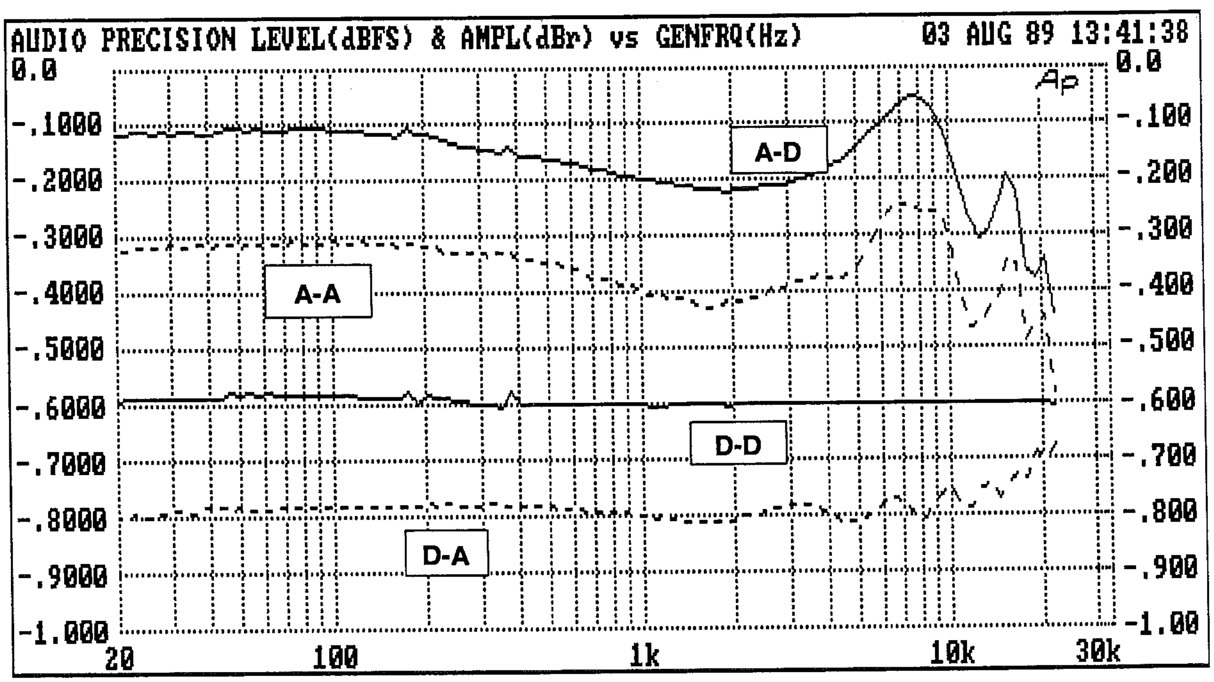

Figure F-1 Frequency Response of RDAT Recorder, Measured in All Four Domain Combinations

GENANLR.DSP is a real-time program. Signal is constantly generated and measurements can be constantly observed in panel or bargraph (F2) mode for adjustment purposes. Generated sinewave signals may be swept in frequency or amplitude as a SOURCE-1 parameter, and one or two measured parameters may be plotted onto X-Y graphs. Figure F-1 below shows the frequency response of an RDAT recorder measured in all four possible ways-analog to digital, digital to analog, digital to digital, and analog to analog. Figure A-4 on page A-5 shows THD+N vs frequency on the same recorder, also measured in all four possible modes.

Figure F-2 shows the DSP panel when GENANLR.DSP has been loaded via the Names Program menu command. The available multiple choices are shown in "exploded" fashion. Figure F-3 shows the DSP HELP screen. Figure F-4 is an equivalent functional block diagram of the analyzer portions of GENANLR.DSP

F.2. Digital Generator

GENANLR.DSP includes a digital sinewave generator program in addition to its digital analysis capabilities. The generator maximum frequency is limited to 22.5 kHz at the 48 kHz rate, 20.67 kHz at the 44.1 kHz rate, and 15 kHz at the 32 kHz rate..

The AMPLITUDE and FREQUENCY fields permit control of the signal generated. The sinewave is a full 24-bit amplitude resolution signal, with frequency resolution of (rate)/2^{24} where "rate" is the digital sample rate selected. With a 48 kHz sample rate, for example, the resolution is $\frac{48{,}000}{2^{24}}$ or approximately 0.0029 Hz.

The digital generator offers only two choices of units, %FS and dBFS. In both cases FS refers to digital full scale, a precisely defined quantity for digital audio signals. Positive digital full-scale is defined as 7FFFFF Hex, negative full scale is 800000 Hex, and the zero signal baseline is 000000 Hex.

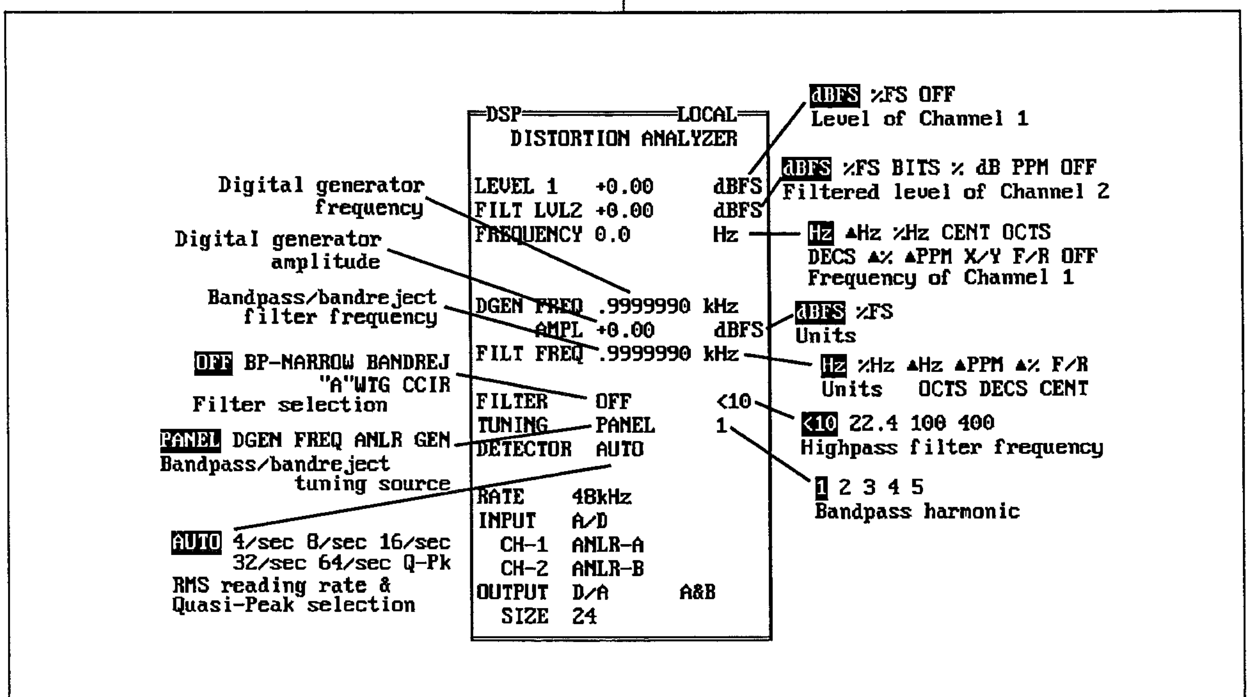

Figure F-2 DSP Panel With GENANLR.DSP Program Loaded

DSP HELP SCREEN

Digital Sinewave Generator and Distortion Analyzer

This program replicates the functions of the System One generator and analyzer for digital signals. It generates sinewave signals digitally and will simultaneously measure the input signals for level, filtered level and frequency.

The filter may be a bandpass filter for spectral analysis, a notch filter for THD+N measurements, or weighting filters for noise measurements. The bandpass and notch filters may be fixed at the panel frequency, may be tuned by the frequency counter, or may follow the analog or digital generator frequency.

To measure distortion set both DSP input channels to the same source channel (A or B) and set the filter to bandreject. To measure crosstalk set the input channels to opposite source channels and select the bandpass filter. Weighted noise measurements can be made on the source channel selected for the DSP channel 2 input by selecting the appropriate weighting filter. When the CCIR filter is selected with an RMS detector, the unity gain frequency is changed to 2kHz from the 1kHz used with the QPK detector.

The LEVEL1 display uses the LEVEL field of the settling panel, the FILT LVL2 display uses the AMPL field, and the FREQ display uses the FREQ field.

Figure F-3 DSP HELP Screen with GENANLR.DSP Loaded

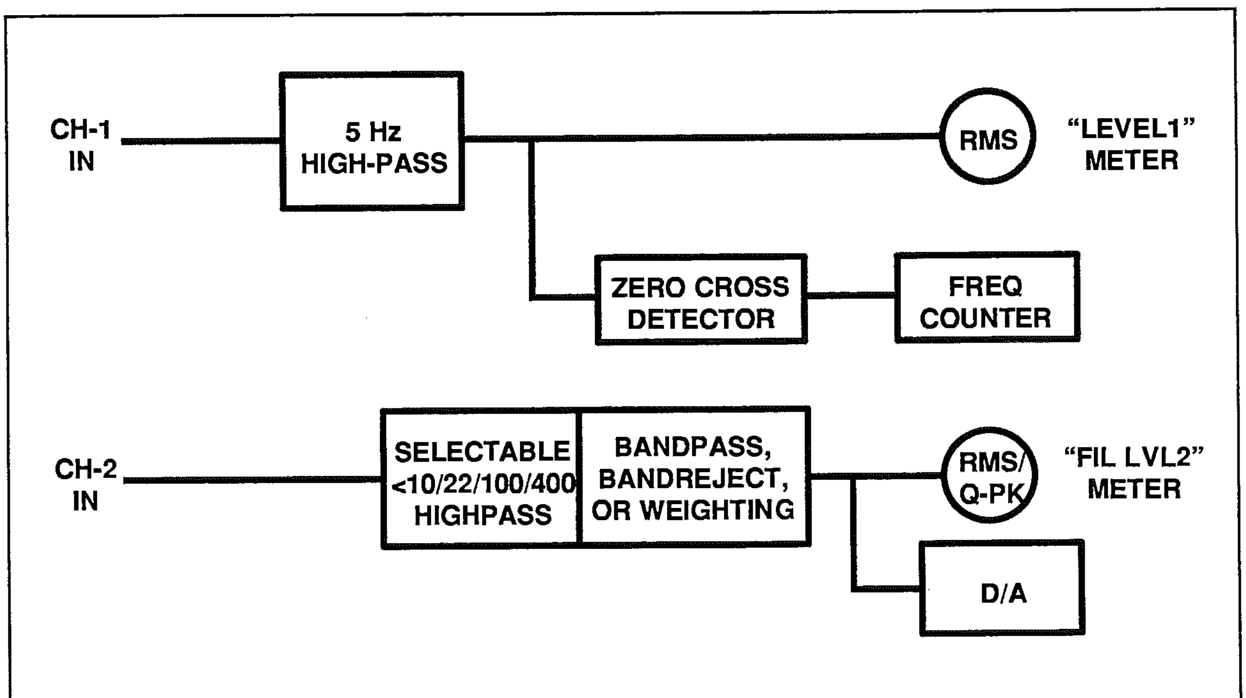

Figure F-4 Functional Block Diagram, Analyzer Portion of GENANLR.DSP

Amplitude calibration is in terms of a peak-equivalent sinewave (as are the non-sinusoidal waveform selections of the analog generator).

Caution should be exercised in not selecting output levels near maximum when significant amounts of dither are also selected. The sinewave signal and dither signal are independently generated. If the sum of the two exceeds digital full scale, clipping will occur.

F.3. Measurement Capability

The measurement portion of GENANLR.DSP consists of two amplitude measurement meters plus a frequency counter. One meter, LEVEL 1, is always broadband (except for a high-pass filter at 5 Hz or below), and is always driven by the CH-1 INPUT source selected near the bottom of the DSP panel. The second meter, FILT LVL2, may be broadband (but with a 5 Hz high-pass filter) or filtered and is always driven by the CH-2 INPUT source. The same source may be selected at both CH-1 and CH-2 to drive both meters simultaneously. The FREQUENCY counter is always driven from the CH-1 signal.

GENANLR.DSP functions only at the standard digital audio sample rates of 32 kHz, 44.1 kHz, or 48 kHz. The DSP module will synchronize to received rates within ±5% of whichever of those rates is selected. However, frequency measurements will be in error by any percentage error in the incoming sample rate.

F.3.1. Reading Rate

Detector update rate for both meters is selected in the DETECTOR field from choices of 4/second through 64/second plus AUTO. The AUTO automatically selects the fastest rate suitable for the measurement being made.

When in PANEL mode, the auto selection sets 4 readings per second to allow adequate time to view each reading. In bargraph (<F2>) mode, 8 readings per second is set to provide rapid feedback to the operator while adjusting the device under test.

During sweeps, the software looks at the filter selection to determine the appropriate reading rate. If one of the weighting filters ("A"WTG or CCIR) is chosen, the software assumes noise measurements are being made and selects 4 readings per second regardless of signal frequency. In bandpass and bandreject modes, the reading rate is based on the filter tuned frequency. In the flat (OFF) mode, the reading rate is based on the frequency measured by the DSP frequency counter.

The DETECTOR field also permits selection of a quasi-peak detector response (Q-Pk) for weighted noise measurements on digital domain audio signals, in conformance with CCIR Recommendation 468-4. When Q-Pk is selected, an RMS detector is used for LEVEL-1 reading and the quasi-peak detector for the FILT LVL2 reading. Both displays will be updated at the 4 readings per second rate since this mode is intended for noise measurements only.

F.3.2. Filter Responses

For the FILT LVL2 meter measuring the CH-2 signal source, filter are selectable in two panel fields following the FILTER label. The first field offers the selective and weighting filter choices of OFF (flat response), BP-NARROW, BANDREJ, "A"WTG, and CCIR. The second field allows selection of four high-pass filter values; <10, 22.4, 100, and 400 Hz. These high-pass values may be selected in conjunction with any of the filter shapes selected in the first field. All of the high-pass filter choices are two-pole (12 dB/octave) when selected in conjunction with the bandreject filter. When used with one of the weighting filters, they are four-pole (24 dB/octave) maximally flat designs. The 400 Hz high-pass becomes a steep ten-pole elliptical filter when the selective or weighting filter choice is OFF; the other high-pass filters remain four-pole. See the High Pass Filters section below for more details.

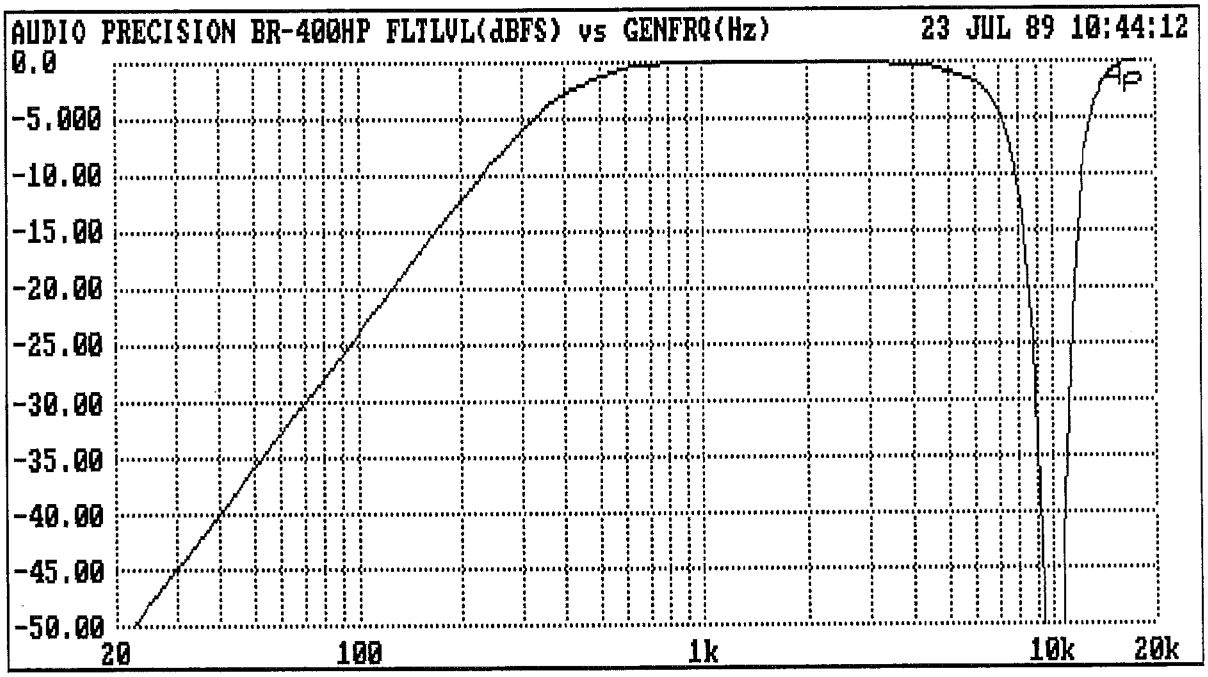

Figure F-5 Response of 400 Hz Highpass Filter Selection With Bandreject Filter Tuned to 10 kHz

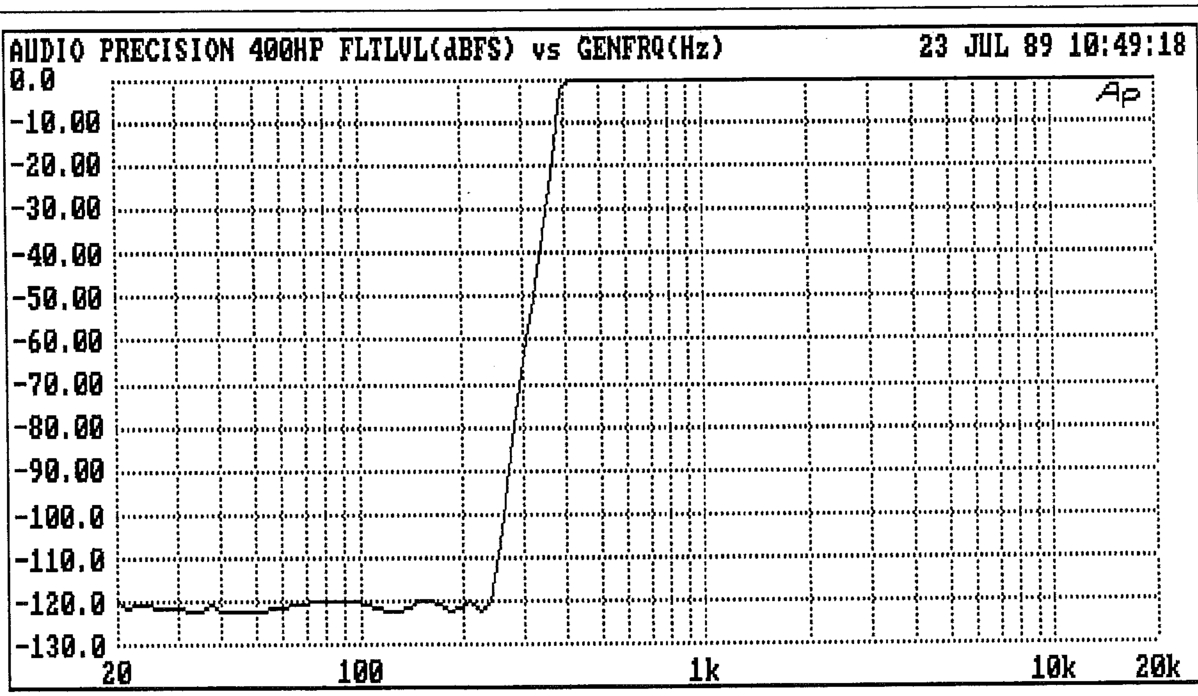

Figure F-6 Response of 400 Hz Highpass Filter Selection with Main Filter Off

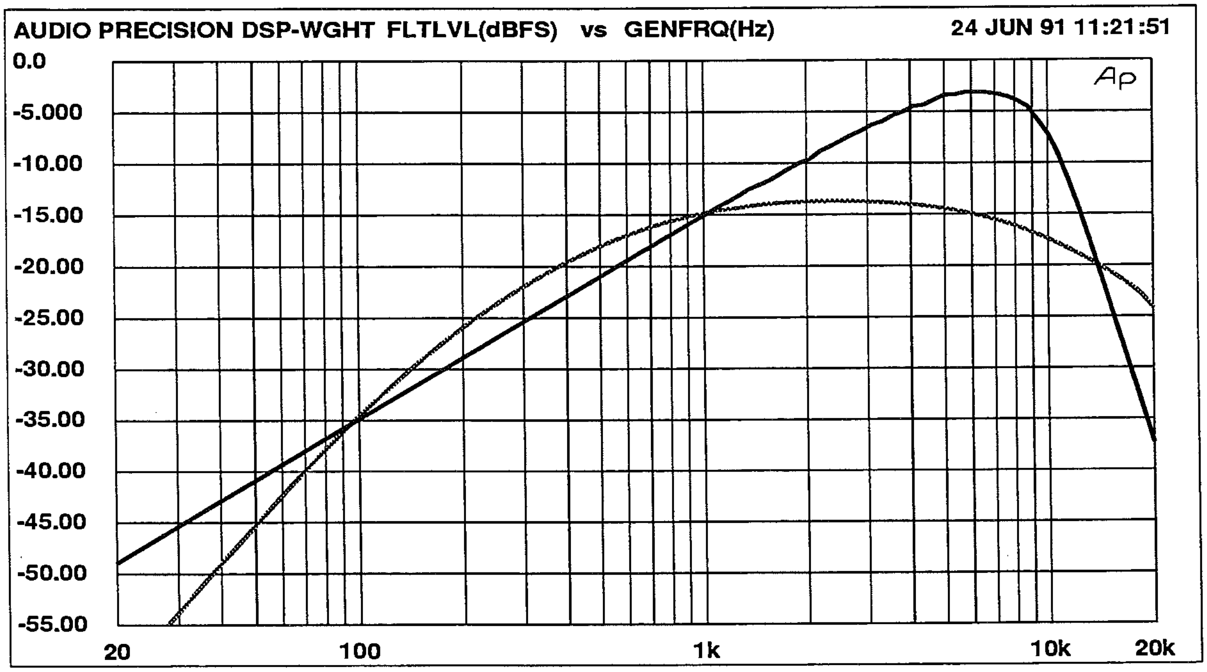

Figure F-7 CCIR (Q-Pk) and A-Weighting Curve Responses, GENANLR.DSP Program, Generator Amplitude -15.0 dBFS. With RMS Detector Selection, CCIR Unity-Gain Frequency Shifts to 2 kHz.

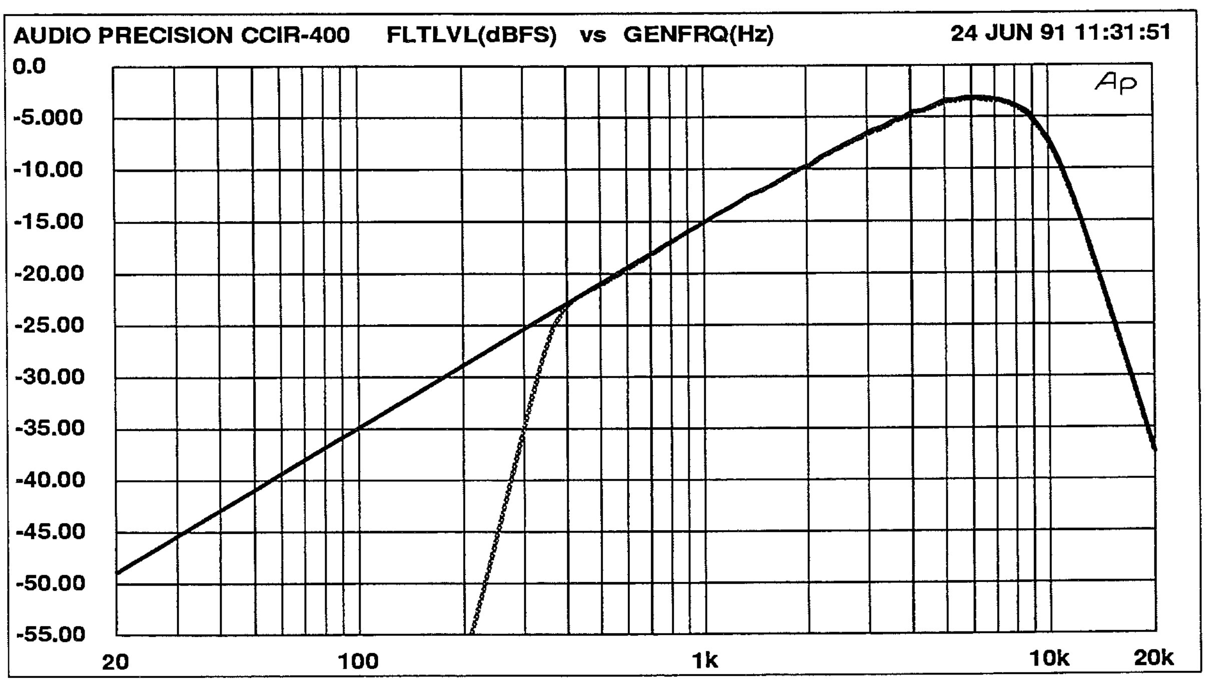

Figure F-8 CCIR (Q-Pk) Response with and without 400 Hz Highpass Filter, Generator Amplitude -15.0 dBFS

F.3.2.1. Selective and Weighting Filters

The bandpass filter (BP-NARROW) can be tuned to any center frequency from 0.04% to 40% of the sample rate; for example, from 20 Hz to 19.2 kHz at the 48 kHz rate. The filter is a 10-pole design with a nominal 1/13 octave bandwidth (Q of 19, -3 dB bandwidth of approximately 5.2% of center frequency). Figure F-9 shows the frequency response curve of this filter and the BANDREJ (bandreject, or notch) filter superimposed. The response will become non-symmetrical as the center frequency approaches 20 kHz, identically to the BP-NARROW choice of the HARMONIC.DSP program described in the previous chapter. The BP-NARROW filter is provided for any type of selective amplitude measurements such as level linearity, AC mains hum analysis, individual harmonic distortion, etc.

The BANDREJ filter tunes from 0.1% to 40% of the sample rate; for example, from 50 Hz to 19.2 kHz at the 48 kHz rate. It is intended principally for THD+N (total harmonic distortion plus noise) and quantization noise measurements on digital audio signals. This selection also includes a 2-pole highpass filter at approximately 5 Hz included to block DC. Selecting either the 22.4, 100, or 400 Hz high-pass filter in the second field moves the two-pole (12 dB/octave) highpass filter to the -3 dB frequency selected. The BANDREJ with 400 Hz high-pass combination is intended for THD+N measurements on signals with large amounts of power mains-related noise. See Figure F-5 for a response measurement of this filter combination with the notch filter set to 10 kHz.

The "A" and "CCIR" choices are DSP-implemented weighting filters for noise measurements. The A-weighting filter is more commonly used in North America, while the CCIR weighting curve is in common use in Europe. CCIR Recommendation 468-4 spells out the use of this filter plus a quasi-peak detector for weighted noise measurements. The CCIR filter and Q-Pk detector of GENANLR.DSP conform to this standard and thus permit measurements in the digital domain which can be correlated to analog audio measurements. When the CCIR filter is used, the gain depends on the type of detector employed. If the quasi-peak detector is selected, the CCIR filter maximum gain will be 12.2 dB as specified in CCIR Recommendation 468-4. This results in a 1 kHz unity gain frequency. If the RMS detector is selected, the unity gain frequency moves to 2 kHz as normally used in CCIR/ARM measurements. This is equivalent to the CCIR_2K selection on the analog analyzer. See Figure F-7 for superimposed response graphs of the DSP A-weighting and CCIR weighting filters.

F.3.2.2. High-Pass Filters

The 400Hz selection, when the first field is set to OFF, is an extremely sharp, high-rejection 400 Hz high-pass filter (ten-pole elliptical). See Figure F-6 for a response curve of this filter. The 400Hz filter selection permits quantization noise and distortion measurements of A/D converters and digital systems. Quantization noise and distortion measurements are made by driving a digital system at its full input amplitude range with a low-frequency sinewave whose frequency is not integrally-related to the sampling rate. This signal exercises the A/D converter while the filter attenuates the fundamental and all harmonics below approximately 220 Hz by at least 120 dB, but passes wideband noise above 400 Hz unattenuated.

All of the high-pass filters are two-pole (12 db/octave) designs when the bandreject filter is selected. When used with one of the weighting filters, they become four-pole designs. See Figure F-8 for response curves of the CCIR filter (Q-Pk detector selection) with the 400 Hz high-pass filter simultaneously selected. With the first filter field of the GENANLR panel set to OFF, the 22.4 Hz and 100 Hz filters remain four pole while the 400 Hz filter becomes the steep ten-pole design described above.

F.3.3. Bandpass and Bandreject Filter Tuning

The tunable bandpass and bandreject (BP-NARROW, BANDREJ) filters can be tuned in several methods selected on the TUNING line. The available TUNING modes are PANL, DGEN, FREQ, ANLR, and GEN. The bandpass filter may also be automatically tuned to the 2nd, 3rd, 4th, or 5th harmonic of the FILT FREQ value by selecting 2, 3, 4, or 5 instead of 1 in the field at the right end of the TUNING line.

PANL mode controls the filter frequency from a user entry in the FILT FREQ field three lines above the TUNING field. The bandreject filter may only be tuned to the fundamental tuning source frequency. PANL should be the selected tuning mode when it is desired to directly sweep the BP-NARROW or BANDREJ filter center frequency by the DSP FILFRQ selection at SOURCE-1 or SOURCE-2 of the SWEEP (F9) DEFINITIONS panel.

DGEN mode steers the filter frequency along with the digital generator frequency. Thus, a manual entry into the GEN FREQ field will simultaneously set the DSP generator and filter frequencies to the same value. Sweeping DSP GENFRQ at SOURCE-1 or SOURCE-2 will cause the FILT LVL2 measurement filter to track the digital generator. With the BANDREJ filter selection and both CH-1 and CH-2 measuring the same signal, this can produce a swept measurement of THD+N versus frequency of a digital audio device. See Figure F-10 for the setup panels for such a test. With the BP-NARROW filter selected, CH-1 measuring the driven channel, and CH-2 measuring the non-driven channel, a sweep of crosstalk/separation of a digital device will result.

The FREQ selection on the TUNING line causes the BP-NARROW or BANDREJ filter to be steered to the value currently being measured by the DSP frequency counter and displayed near the top of the DSP panel. Thus, as an incoming digital domain audio signal from a CD, tape, or digital transmission is received, the filter is automatically tuned to the signal frequency. With the BANDREJ filter, this mode will produce real-time display of THD+N of digital domain signals.

The ANLR selection refers to the frequency of the DIS-1 bandpass-bandreject filter in the analog analyzer. The analog filter, in turn, is steered by the analog analyzer's frequency counter in PANEL or SOURCE-1 EXTERN FREQ sweep modes and will

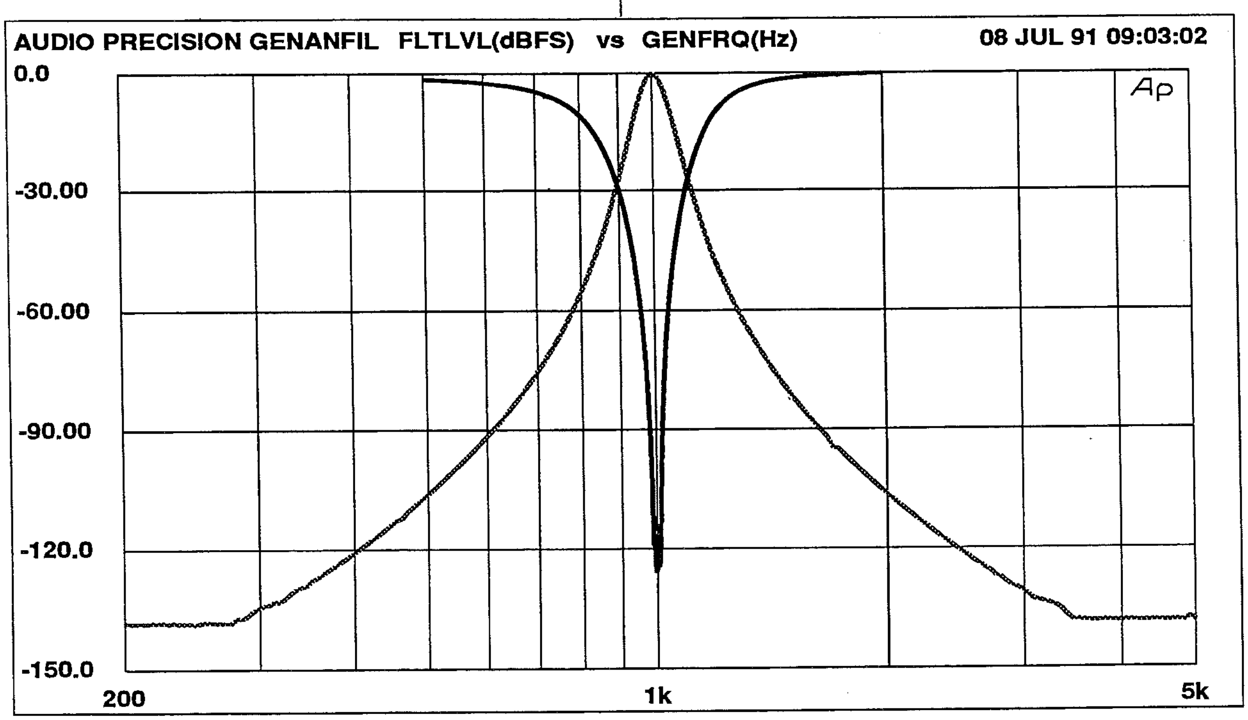

Figure F-9 Bandpass and Bandreject Filter Resonse, GENANLR.DSP track the analog generator during SOURCE-1 GEN FREQ sweeps except in the IMD modes of the System One analyzer.

The GEN selection at TUNING refers to the frequency of the analog generator. This mode is convenient for making THD+N measurements in the digital domain versus frequency, as the analog generator drives the input of an A/D converter.

F.3.4. Analog Output of Digital Input Signals

When the output signal is routed to a digital interface, the ANALOG OUTPUT D/A connector is driven by the output of the DSP filter (FILT LVL2). This function is analogous to the MONITOR OUTPUT READING connector of the analog analyzer. It permits audible or oscilloscope monitoring of the signal and, in BANDREJ modes, of the distortion products. Gain is automatically selected in the DSP depending upon digital signal level, so low level signals and distortion products can be properly monitored without significant additional distortion being added by the D/A converter.

F.3.5. Units of Measure

The LEVEL 1 meter offers only two choices of units, %FS and dBFS. In both cases FS refers to digital full scale, a precisely defined quantity for digital audio signals. Positive digital full-scale is defined as 7FFFFF Hex, negative full scale is 800000 Hex, and the zero signal baseline is 000000 Hex. Amplitude calibration is in terms of a peak-equivalent sinewave (as are the non-sinusoidal waveform selections of the analog generator).

In addition to these same two absolute units, the FILT LVL2 meter also offers four relative units: BITS, %, dB, and PPM. In all four cases, the relative units refer to the present reading of the LEVEL 1 meter. Thus, if the LEVEL 1 meter is measuring one input channel (CH-1 A, for example) and the FILT LVL2 meter is measuring the opposite channel (CH-2 B), a relative unit can be selected for FILT LVL2 and the display will indicate stereo separation or crosstalk; see Figure F-11. If the same signal is selected at both inputs (CH-1 A and CH-2 A, for example) and the BANDREJ filter is selected in the FILT LVL2 path, a relative unit selection will result in a display of THD+N measured in the digital domain. The dB and % units are direct computations. PPM stands for parts per million; 0.0001% or -120 dB is one PPM. The BITS unit is derived from dB at the scale factor of 6.02 dB per bit.

F.3.6. DSP Frequency Measurements

The FREQUENCY display field near the top of the DSP panel is driven from a DSP-implemented frequency counter. The measured frequency may be displayed (or plotted at DATA-1 or DATA-2) in relative units in addition to Hz. All the relative units refer to the REFS Freq field on the analog ANALYZER panel.

F.3.7. Input Signals

Only System One Dual Domain (SYS-300 series) can acquire and perform analysis of digital audio signals directly in the digital domain.

The INPUT fields near the bottom of the DSP panel permit selection of the signal type and source. If A/D is selected as INPUT, one or two channels of analog signal will be converted to digital samples by the two 16-bit A/D converters in the DSP unit and presented to the DSP chip for analysis. Since the GENANLR.DSP program provides only measurement units expressed in digital domain values, analysis of analog domain signals is generally better performed with System One's analog analyzer.

If SERIAL, PARALLEL, or DGEN are chosen as the source on the INPUT line, the choices at CH-1 and CH-2 become A, B, or NONE. When SERIAL is selected, the UTIL SERIAL-DSP MODE menu command permits selection among three serial interfaces—AES/EBU, SPDIF/EIAJ, and a general-purpose SERIAL interface. AES refers to the AES/EBU format at a front panel female XLR connector. SPDIF selects the SPDIF/EIAJ (Sony Philips Digital Interface) front-panel digital input. SE-

RIAL refers to the general-purpose serial input on the 15-pin D subminiature connector on the DSP rear panel. The parallel input connector and general-purpose serial input/output connectors are located on the instrument rear panel. A and B refer to the two multiplexed channels available in any of these digital formats. These are often assigned, respectively, as left and right channels of audio in a stereo recording or transmission. DGEN refers to the digital generator output. DGEN may be thought of as functionally similar to the GEN MONITOR path between the analog generator and analog analyzer. The front-panel AES/EBU and SPDIF/EIAJ connectors are somewhat different and function differently on units with serial numbers below and above SYS1-32214.

F.3.8. Analog Audio Outputs

For most hybrid digital-analog measurement applications, any required analog audio signals should be provided from System One's analog generator. This analog generator features distortion products typically 120 dB below the fundamental across the mid-frequency audio range and is suitable for critical tests of the A/D converters up to and including 20 bits. However, it is also possible to obtain analog output from the digital generator if desired. When the OUTPUT D/A selection is made near the bottom of the DSP panel, analog output from the digital generator is available via an internal 16-bit D/A converter. Distortion of the D/A converter is typical of 16-bit converters at about -85 to -90 dB, considerably inferior to the analog generator. The channel selections (A, B, A&B) on the OUTPUT line of the DSP panel have the same effect when D/A is selected, since there is only a single DAC output.

F.4. Typical Applications and Sample Tests

The DSP program diskette also contains a number of sample tests and procedures using the GENANLR.DSP program. These tests include stereo frequency response and stereo THD+N versus frequency measurements for all four possible combinations of analog and digital interfaces.

All are stored with the 48 kHz sample rate selection, but can be changed as desired. If the 32 kHz sample rate is used, the upper sweep frequency limit on the SWEEP (F9) DEFINITIONS panel of the response and THD+N tests should be changed to a value no higher than 15 kHz.

All are stored with SERIAL digital interface selected. When S1.EXE software is loaded, the default serial interface sub-set selected is AES-EBU. This may be changed to SPDIF (EIAJ) or general-purpose SERIAL by the UTIL SERIAL MODE menu command. PARALLEL interface mode is selected on the DSP panel.

F.4.1. Frequency Response Tests

The response tests are named A-A-RESP.TST, A-D-RESP.TST, D-A-RESP.TST, and D-D-RESP.TST. All are set up as single-sweep stereo tests, since both digital and analog analyzer have two-channel generator outputs and separate, independent two-channel voltmeters.

Analog generator output conditions on the A-A and A/D tests may need to be changed for the particular device being tested. All are stored with unbalanced 25 Ohm output and a one Volt rms generator dBr reference level, as may be appropriate for a consumer RDAT recorder with variable record level control. Professional devices are likely to have balanced inputs with full-scale values on the order of +18 to +24 dBu. Using A/D-RESP.TST, the full-scale analog input value may be determined by observing the digital output readings on the DSP panel as the analog generator amplitude is adjusted. This can easily be done experimentally with both the GENERATOR and DSP panels on screen. To do this from the normal (<Ctrl><Home>) panel orientation of GENERATOR-ANALYZER-SWEEP (F9) DEFINITIONS, move the panel cursor to the center (normally ANALYZER) panel, then press <Ctrl><PgDn>. Since the DSP panel is the last of the nine panels, these keystrokes replace the current

```
                                        ═DSP══════════════LOCAL═╗ ╔SWEEP (F9) DEFINITIONS═╗
                                          DISTORTION ANALYZER     DATA-1     DSP    FLTLVL
                                        LEVEL 1   -0.01    dBFS   GRAPH TOP  +0.00    dBFS
                                        FILT LVL2 -120.97  dBFS     BOTTOM   -130.00  dBFS
                                        FREQUENCY 455.9984 Hz       # DIVS   0        LIN

                                                                  STEREO     DSP    FLTLVL
                                        DGEN FREQ 456.0013 Hz     GRAPH TOP  +0.00    dBFS
                                             AMPL +0.00    dBFS     BOTTOM   -130.00  dBFS
                                        FILT FREQ .9999990 kHz      # DIVS   0        LIN

                                        FILTER    BANDREJ         SOURCE-1   DSP    GENFRQ
                                        TUNING    DGEN       1      START    50.0000  Hz
                                        DETECTOR  AUTO              STOP     10.0000  kHz
                                                                    # DIVS   0        LOG
                                        RATE    48kHz
                                        INPUT   SERIAL            # STEPS    30
                                          CH-1  A                   TABLE    OFF
                                          CH-2  A
                                        OUTPUT  SERIAL    A&B     DISPLAY    MONO-GRAPH
                                          SIZE  24
PANEL DGEN FREQ ANLR GEN                                 To change setting, use SPACE bar.
Bandpass/bandreject tuning source                        To return to menu, press the Esc key
```

Figure F-10 Setup Panels, THD+N (dB Below Digital Full Scale) vs Frequency

panel with the DSP panel. Then, adjust the generator amplitude until the digital output levels from the device as displayed on the DSP panel are 0 dBFS. Press <F3>, which will store the current value of analog generator output as the new generator dBr reference. You can then manually enter this value of generator dBr reference in all the tests which use the analog generator (A-A-RESP.TST, A-D-RESP.TST, A-A-THD.TST, and A-D-THD.TST).

The frequency response tests are all stored to sweep at an amplitude one dB below full scale, since a digital system cannot be driven above 0 dB and there may be some frequencies with more gain than at the 1 kHz reference frequency.

F.4.2. THD+N vs Frequency Tests

The distortion tests are named A-A-THD.TST, A-D-THD.TST, D-A-THD.TST, and D-D-THD.TST. See Figure F-10 for the key DSP and SWEEP DEFINITIONS panel settings of D-D-THD.TST. Since both the analog and digital analyzers can only measure THD+N on one channel at a time, these tests are all set up in STEREO mode at DATA-2. They thus perform two frequency sweeps, measuring channel A on the first sweep and channel B on the second.

A-A-THD.TST and A-D-THD.TST are stored to sweep at -1.0 dBr to allow for systems with analog-input frequency response peaks above the 1 kHz value. Better THD+N numbers can be obtained by operating at a higher input if this 1.0 dB margin is excessively conservative for the particular device being tested. The exact input value for 0 dBFS output can be determined experimentally as described above under Frequency Response Tests. Generator output impedance and balance-unbalance selections may also be changed as required. The digital output tests, D-A-THD.TST and D-D-THD.TST, run at 0 dBFS.

The THD+N tests using digital measurement (A-D-THD.TST and D-D-THD.TST) operate only to a lower frequency limit of 50 Hz, limited by the digital bandreject filter frequency specifications.

```
=DSP==============LOCAL==
  DISTORTION ANALYZER

LEVEL 1    -0.01      dBFS
FILT LVL2 -134.97     dBFS
FREQUENCY 455.9984    Hz


DGEN FREQ 456.0013    Hz
     AMPL +0.00       dBFS
FILT FREQ .9999990 kHz

FILTER    BP-NARROW
TUNING    DGEN        1
DETECTOR  AUTO

RATE      48kHz
INPUT     SERIAL
  CH-1    A
  CH-2    B
OUTPUT    SERIAL   A
  SIZE    24

A  B  NONE
DSP Channel 2 Input Source
```

Figure F-11 DSP Panel, Crosstalk/Separation Measurement in Digital Domain

F.4.3. Crosstalk-Separation Tests

For inter-channel crosstalk and stereo separation tests with digital domain devices, one channel must be driven while a selective (bandpass filtered) amplitude measurement is made on the other channel. The bandpass filter must track the frequency of the generator in use. See Figure F-11 for the DSP panel settings for separation measurements on a digital input-output device. The digital generator drives channel A while the FILT LVL2 meter, in BP-NARROW mode, measures channel B. The TUNING DGEN 1 selection causes the filter to track the digital generator fundamental frequency. For a swept, graphed measurement, the SWEEP DEFINITIONS panel settings would be exactly as in Figure F-10.

F.4.4. Linearity Tests

Linearity tests and procedures will also be found on the diskette for analog input-digital output and digital input-analog output devices. These are labeled A-D-LIN.PRO, A-D-LIN.TST, D-A-LIN.PRO, and D-A-LIN.TST. Loading and running the procedure will cause the test of the same name to be run. The test drives both stereo inputs of the device in an output versus input amplitude sweep across a 120 dB dynamic range in 2 dB steps. Stimulus frequency is 500 Hz, and output is selectively measured.

D-A-LIN.TST makes its amplitude sweep downwards from 0 dBFS and therefore will work with converters of any resolution, but is stored with dither at the 16-bit level. To run it with an 18-bit converter, for example, the dither level (SIZE) should be changed to 18 and the test re-saved. Correct dither amplitude is critical to the test function. With no dither, linear operation will be truncated as the signal passes below the LSB amplitude. To test at another frequency, change both the DSP generator frequency and the analog analyzer BP/BR filter frequency to the new value and re-save the test before running the procedure.

The procedure will first set the analyzer dBr reference level to the measured output of channel A, perform amplitude sweeps and measurements in sequence on both channel A and channel B, use COMPUTE LINEARITY to fit a best straight line to the -10 dBr to -40 dBr portions of the measured data on each channel, change the graph coordinates to display deviation from perfect linearity, and re-graph the data.

A-D-LIN.TST is stored as a downwards amplitude sweep from 0 dBr with the generator dBr reference 1 Volt rms. Generator output is set up as unbalanced. The output impedance and balance-unbalance selection should be changed to appropriate values for the particular device to be tested. The generator dBr reference should be changed to the analog full-scale input for the device as described above under frequency response tests. SAVE TEST to store this modified setup.

The procedure will then run similarly to D-A-LIN.PRO. Amplitude sweeps will be made sequentially on both channels, the COMPUTE LINEARITY function used, graph coordinates changed, and deviation from perfect linearity displayed.

Selective (BP-NARROW) amplitude measurement is critical to linearity measurement. For D/A testing, the BANDPASS function of the analog analyzer must be used and RDNG plotted. The analog bandpass filter must be fixed at the digital generator frequency. For A/D testing, the FILT LVL2 meter of GENANLR must be used and plotted, in BP-NARROW mode. The DSP filter tuning source should GEN, the analog generator frequency. Note that the FILT LVL2 meter always measures CH-2, so select the digital channel appropriately in the INPUT CH-2 field near the bottom of the DSP panel.

F.5. Furnished Files

A-A-RESP.TST test file to measure stereo frequency response through device with analog inputs and outputs. Does not require DSP

A-A-THD.TST test file to measure THD+N vs frequency on both channels of stereo device with analog inputs and outputs. Does not require DSP

A-D-LIN.PRO procedure to measure amplitude linearity of stereo device with analog inputs and digital outputs. Uses A-D-LIN.TST, COMPUTE LINEARITY

A-D-LIN.TST test file to measure amplitude linearity of stereo device with analog inputs and digital outputs. Used in A-D-LIN.PRO

A-D-RESP.TST test file to measure stereo frequency response of device with analog inputs and digital outputs

A-D-THD.TST test file to measure THD+N vs frequency on both channels of stereo device with analog inputs and digital outputs

D-A-LIN.PRO procedure to measure amplitude linearity of stereo device with digital inputs and analog outputs. Uses D-A-LIN.TST, COMPUTE LINEARITY

D-A-LIN.TST test file to measure amplitude linearity of stereo device with digital inputs and analog outputs. Used in D-A-LIN.PRO

D-A-RESP.TST test file to measure stereo frequency response of device with digital inputs and analog outputs.

D-A-THD.TST test file to measure THD+N vs frequency on both channels of stereo device with digital inputs and analog outputs

D-D-RESP.TST test file to measure stereo frequency response of device with digital inputs and outputs

D-D-THD.TST test file to measure THD+N vs frequency on both channels of device with digital inputs and outputs

GENANLR.DSP version 2.10 revised program

G. DIGITAL ERROR RATE MEASUREMENT PROGRAM BITTEST.DSP

G.1. Introduction

This program generates digital test signals and measures the returned digital signals for bit errors. BITTEST operates at 48 kHz, 44.1 kHz, and 32 kHz sample rates. The test signal may be a pseudo-random sequence, constant valued samples ("digital dc"), a sinewave of selectable amplitude and frequency, or walking bit patterns. Generated word width is always 24 bits. The measurement may display both real-time received data and errors in the received data sequence. Any amount of delay between transmitted and received signals is permissible, allowing testing of devices and transmission links with large amounts of delay or even recorder-reproducers. Measurement word width is selectable for all signals. No dither is added to any of the signals.

The expression "loading" a DSP program is used frequently throughout this manual. In fact, NAMES PROGRAM is the specific command required to download a .DSP program from computer disk to DSP unit. When a .TST file is saved to disk after using the NAMES PROGRAM command, the DSP program will also be automatically downloaded each time the .TST file is loaded thereafter (unless the DSP program is already in place from the previous test).

The extensive signal generation and error measurement capability of BITTEST is useful for investigating the integrity of digital audio data links, recorders, etc. It is also invaluable for design test of digital interfaces. Each waveform in the program

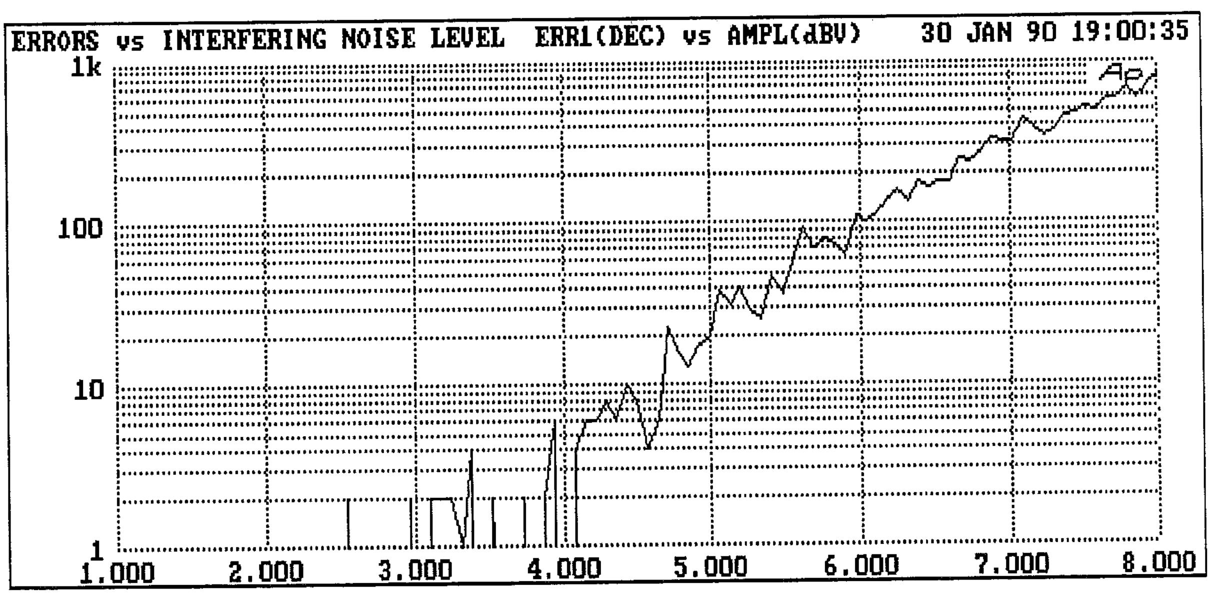

Figure G-1 Errors vs Noise Level Coupled into AES/EBU Interface

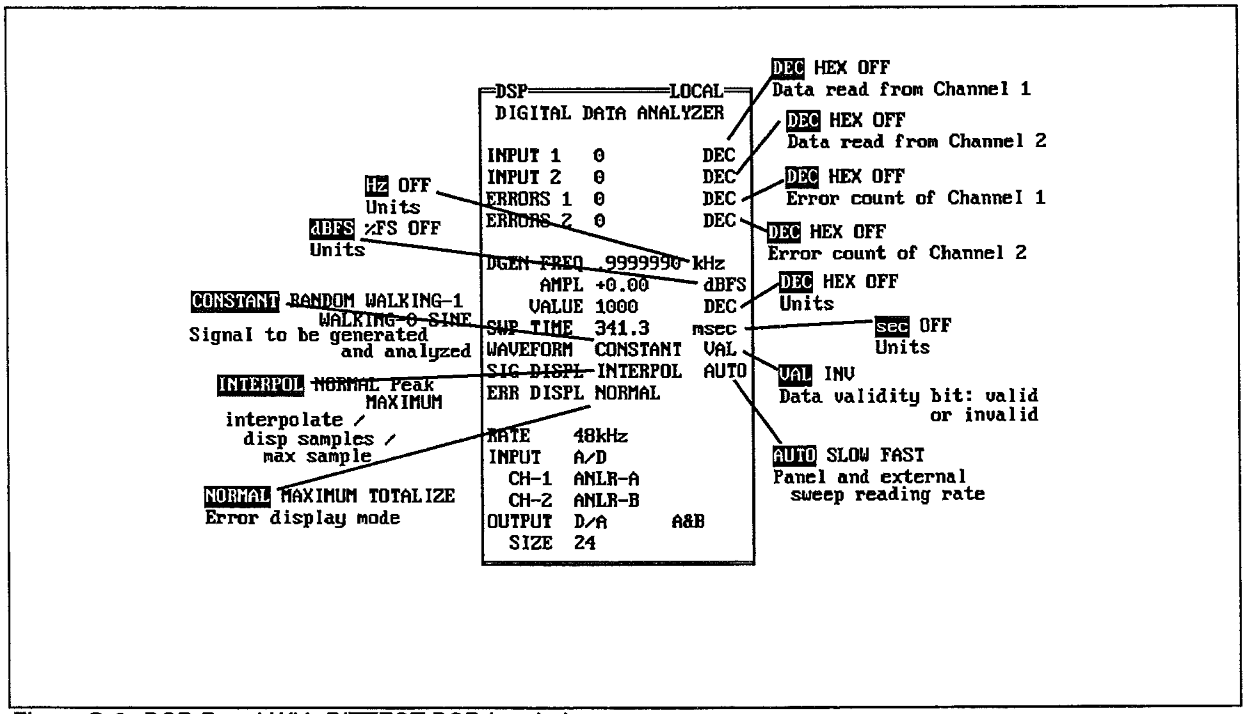

Figure G-2 DSP Panel With BITTEST.DSP Loaded

DSP HELP SCREEN

Digital Data Generator and Error Rate Analyzer

This program generates digital data and tests the received data for errors against the pattern sent. This pattern may be a constant, a psuedo-random sequence, a walking bit pattern, a sinewave, a ramp, or a repetitive pulse.

The number of errors per measurement interval may be displayed, the MAXIMUM of these values may be displayed or a running total may be accumulated in TOTALIZE mode. To reset the TOTALIZE or MAX-HOLD error count press enter on the field. No errors are reported with ramps or pulses.

The SIZE selection sets number of input bits analyzed, the output word width is always 24 bit. The two DSP input channels (1 and 2) should normally be set to source channels (A and B).

When using this program SETTLING is automatically defeated.

Figure G-3 DSP HELP Screen with BITTEST.DSP Loaded has a specific testing application. BITTEST is not intended for use with analog input or output signals and such use will produce meaningless results.

BITTEST.DSP is a real time program, but also contains the ability to capture signals and display them in the time domain (digital storage oscilloscope mode). Signal is constantly generated and measurements can be continuously observed in panel or bargraph (F2) modes. This is particularly useful for making adjustments on devices under test. Some of the test signals use the rear-panel TRIG OUT BNC connector to trigger an oscilloscope or logic analyzer. Signal parameters, where adjustable, may be swept as a SOURCE 1 parameter, and one or two measured parameters may be plotted on X-Y graphs. Figure G-1 shows the measured error rate of a digital interface as a function of the interfering noise level artificially introduced by a System One random noise generator.

Figure G-2 shows the DSP panel when BITTEST.DSP has been loaded via the Names Program menu command. The available multiple choices are shown in "exploded" fashion. Figure G-3 shows the contents of the Help DSP screen when BITTEST is loaded.

G.2. Signal Generation

Digital-domain signals of five different waveforms may be generated by BITTEST.DSP. The output signal is always of 24-bit resolution.

G.2.1. Constant Mode

The CONSTANT mode generates a continuous stream of data samples at the same fixed value. This is the digital equivalent of a DC voltage source. The data word to be output is set by the DGEN VALUE field on the panel and may be the source parameter in a sweep. The value may be entered in decimal or hexadecimal notation; use a leading zero if the first hex character is alphabetic. The value entered is with respect to the word length entered in the SIZE field near the bottom of the panel. Entering a DGEN VALUE of 1 with SIZE 16, for example, produces a constant stream of samples with the binary value

0000 0000 0000 0001

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

0000 0000 0000 0001 0000 0000.

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

0000 0000 0000 0000 0000 0001.

CONSTANT mode aids in the investigation of data-dependent errors in digital systems. No TRIG OUT signal is generated in this mode. The DGEN FREQ and DGEN AMPL fields have no effect in CONSTANT mode.

G.2.2. Random Mode

The RANDOM mode outputs a pseudo-random sequence uniformly distributed between plus and minus full scale. The amplitude is not adjustable. This signal is intended for error rate testing of communications links, AES/EBU interfaces, and digital recorders. No TRIG OUT signal is output in this mode. The DGEN FREQ, DGEN AMPL, and DGEN VALUE fields have no effect in RANDOM mode.

G.2.3. Walking Bit

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

G.2.4. Sinewave

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

G.3. Validity Bit

The validity bit on the AES/EBU and SPDIF/EIAJ interfaces may be set or cleared as desired to test the behavior of equipment in response to this bit. When the field is set to VAL the validity bit will indicate a normal signal. When the field is set to INVAL the validity bit will indicate an interpolated signal. Some equipment is designed to mute when this occurs. The selected validity bit will be generated in every sample until changed on the BITTEST panel.

G.4. Signal Analysis

Received data is displayed on the INPUT 1 and INPUT 2 fields of the panel, truncated to the word length set in the SIZE field. The readings are updated at the rate selected on the panel (AUTO-SLOW-FAST) until an error occurs. When an error occurs (see below), the INPUT 1 or INPUT 2 field display will "freeze" the data containing the first error detected on that channel. HEX displays do not work correctly for large negative numbers (above 22 bits). DECimal should be used where possible.

Received data is also measured to determine if it matches the data transmitted. Only the number of bits selected in the SIZE field will be analyzed. This comparison is done with algorithms which are insensitive to delay between the send and receive sections. The number of errors in the received data per measurement interval are counted for each channel. The ERR DISPL field selects the type of analysis to be performed. In the NORMAL mode the error counts during the last measurement interval (1/4 second in SLOW or 1/32 second in FAST display mode) are displayed directly in the ERRORS 1 and ERRORS 2 fields of the panel. By selecting MAXIMUM, the maximum error count during any measurement interval will be held in the display. A running total of all errors may be accumulated by using TOTALIZE mode. To reset the TOTALIZE or MAXIMUM error counts press <Enter> with the cursor on the ERR DISPLAY field.

All modes make error measurements based on the signal selection. Thus, RANDOM must be selected when measuring a digital reproducer playing a recording previously made with the RANDOM signal from BITTEST, SINE must be selected to measure errors from a sinewave recording, etc. For error checking with a CONSTANT mode signal, CONSTANT mode must be selected and the DGEN VALUE field must further have the specific value entered which was used when the recording was made. The random, sine, and walking bit modes compare each successively received sample with the algorithm used to generate the signal. Two samples are required before the error analysis synchronizes with the signal, so errors may be indicated during the initial measurement interval even though no error actually exists.

The SIZE field sets the number of input bits to be analyzed in all modes. It affects both data and error displays. The SIZE field also defines word length for the DGEN VALUE entry.

The two DSP input channels (1 and 2) should normally be set to source channels (A and B). If the same channel (A or B) is selected for both channel 1 and 2 the readings will be redundant.

G.4.1. Error Detection Flag

Whenever an error is detected the DSP will pulse the AUXILIARY OUTPUT (pin 11) on the rear panel SERIAL INTERFACE connector. This allows triggering an oscilloscope to observe received data errors. Since the error indication does not occur until after the end of the data word containing the error, a digital storage oscilloscope or logic analyzer is recommended. That instrument should be set in pre-trigger mode to acquire data before the trigger event.

G.5. Waveform Display

BITTEST.DSP can acquire digital-domain signals into the DSP buffers and then display those signals in the time domain (storage oscilloscope mode). Select DSP TIME at SOURCE-1 and either the DSP input signals or the error signals at DATA-1 and DATA-2. See the FFT chapter of this DSP manual for general information on time-domain displays, relationships between DSP samples and SOURCE-1 # STEPS, graphic aliasing factors, and the function of the NORMAL-INTERPOL-PEAK-MAXIMUM choices on the WAVEFORM line of the DSP panel. The SAVE WAVEFORM and LOAD WAVEFORM commands are also functional with BITTEST and permit retaining waveforms on computer disk with later download for further analysis.

G.6. Graphing Data Patterns

Data patterns and error patterns may be graphed versus time or versus some swept parameter. Figure G-1 is an example of errors graphed versus noise amplitude combined with the digital signal into an AES/EBU interface. The swept, horizontal axis parameter may be any of the System One analog, digital, or dc (via DCX-127) variable output capabilities. By selecting EXTERNAL TIME at SOURCE-1, "chart recordings" may be made of data or errors versus time.

G.7. Supplementary Information

For more information on testing digital interfaces, see the paper "Measuring AES/EBU Digital Audio Interfaces" by Richard C. Cabot, Journal of the Audio Engineering Society, Vol. 38, #6, 1990 pp 459-467.

DSP PROGRAM INDEX

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