

Yes, it's finally another issue of AUDIO.TST. No, you probably didn't miss any issues; this is the first since March, 1989. Other priorities involved in the introduction of the new DSP modules unfortunately prevented publishing other issues later in 1989. We plan to keep AUDIO.TST coming at a more regular rate in the future.

The new System One + DSP and System One Dual Domain units began shipments in October 1989. They have already gained wide acceptance, with well over 100 units having been shipped as of January. Many of the early applications are in A/D and D/A converter testing, digital mixing console and digital routing switcher tests, measuring digital audio channels of professional VTRs, and maintenance of all types of audio equipment in broadcast

DSP Report

Original Features, System One + DSP

and recording studio environments.

Earlier AUDIO.TST Newsletter articles gave an idea of some of the basic DSP features. In System One + DSP, these include FFT spectrum analysis, waveform display in digital storage oscilloscope mode, and individual harmonic analysis via a sharp digital filter tracking the incoming signal or any harmonic from second through ninth.

Original Features, System One Dual Domain

System One Dual Domain includes those same features, plus generation and analysis of digital audio signals directly in several formats. The digitaldomain analyzer includes a frequency counter plus two independent amplitude meters with RMS detectors. One meter also has the ability to insert highpass, steerable bandpass or bandreject, or noise filters, and to select a quasi-peak detector. System One Dual Domain can thus measure frequency response, THD+N versus frequency or amplitude, linearity and deviation from perfect linearity, weighted or unweighted noise and S/N ratio, quantization distortion, crosstalk, and frequency in any of the four possible combinations of analog and digital inputs and outputs. Several features have been added or enhanced since the last AUDIO.TST newsletter report. The remainder of this article describes additional capabilities being shipped as of February 1, 1990.

Bit Error Testing Program

BITTEST.DSP is a bit error test program useful in testing digital interface circuitry, digital transmission paths, and digital storage media. It both generates and analyzes walking bit patterns, pseudorandom signals, or a constant amplitude ("digital DC") signal, with any amount of time delay permissible between generator and analyzer. Received bit errors can be displayed in real-time, maximum, or totalize modes. The bit error test program additionally generates a sine wave and a low-amplitude staircase (ramp) signal, though it presently lacks the ability to analyze errors with these last two signals. The figure below shows the staircase signal as acquired on a second System One Dual Domain unit using an FFT pro-





Compressor Output (upper trace) with Sine Burst Input (lower trace), Conventional Linear-Vertical Display

gram in waveform display mode. The staircase signal continuously repeats across the 4 LSBs (16 lowest states) of the digital output. This signal is somewhat similar to the track 21 signal on the CBS Test CD.

Waveform Save and Load

Features have been incorporated into S1.EXE and both FFT programs to permit up-loading entire acquired waveforms from the DSP to computer disk, and later down-loading of disk-stored waveforms to the DSP for further analysis or display. These features also permit transfer of waveforms from one FFT program to the other, or combination of two waveforms acquired at different times or places into the two DSP channels for simultaneous display.

Waveform Envelope Mode

A new "MAXIMUM" waveform display mode has been added which sends to the PC for display *the absolute value* of the greatest positive or negative signal amplitude in each set of samples examined. Since absolute values are always positive, this mode permits signal display in LOG Volts or a dB unit to show wide-dynamic-range signals such as behavior of compressors with tone-burst stimulus. Both figures at the top of this page show the BUR-GEN waveform stimulus to a compressor (lower trace) and the output of the compressor (upper trace). The stimulus waveform amplitude between bursts is set to -20 dB. As viewed on any oscilloscope or in the normal waveform display mode of the FFT programs (left figure above), the overshoot at the





Same Waveforms, MAXIMUM Waveform Display Mode, Log Vertical Display

beginning of the burst and the gain recovery following the burst are visible, but not easily quantifiable. The dB display permitted by MAXIMUM mode (right figure above) makes it easy to see and measure the amount of overshoot and the release time of the compressor.

Before-and-After Tests

Signals can be acquired into the two DSP channels at different times, then displayed as two simultaneous FFT spectra or waveforms in split-screen fashion or overlaid. With the new SAVE WAVEFORM and LOAD WAVEFORM capability, these signals can be acquired months or years apart. Graphics cursors permit easy simultaneous examination of both channels to compare performance of two supposedly identical units, the same unit with and without dither, a prototype before and after design changes, etc.

Increased Spectrum Averaging

FFTGEN.DSP, one of the two furnished FFT programs, includes a mode for repeated signal acquisitions, FFTs, and transform averaging before display. Up to 1,024 acquisitions-and-averages may now be selected. Averaging has two advantages. One is to effectively reduce noise components, making coherent signals stand out more clearly. The two figures below show the residual distortion of the analog generator and analyzer of a System One in THD+N mode at one kHz. The first figure was a single acquisition and FFT. The fundamental is attenuated about 123 dB by the analog notch fil-



Residual Distorion of System One, 256 Times

ter. Second and third harmonics are visible at about -120 and -130 dB, and the noise floor at -140 to -145 dB obscures most lower-amplitude harmonics. The second figure is the result of 256 repeated acquisitions, FFTs, and averaging the FFTs. The noise floor has been reduced to -150 dB, allowing resolution of distortion products such as the 9th harmonic at -147 dB. This averaging mode also has interesting potential with voice and music program material, producing a graph of average spectral energy distribution over periods of up to five minutes.

House Sync

Installations with more than one digital audio device typically use a "house sync" bus as an external reference clock to assure that all digital signals are synchronous. The DSP module's rear-panel external sync input connector has been enabled by the latest DSP program revisions to provide house sync capability for System One Dual Domain-generated digital signals. This external sync capability also permits generating signals at the 44.056 kHz rate used in NTSC-video-related digital audio applications.

AES/EBU Status Bytes

The professional AES/EBU digital audio format and the similar SPDIF and EIAJ consumer formats contain defined status bytes in addition to the audio signal. These bytes control de-emphasis, copy prohibition, sample rate, etc. The System One Dual Domain user may set those bytes as he or she wishes on digital output signals in those formats.

General Purpose Serial Port

A rear-panel serial input/output port has been enabled by the latest DSP program revisions. Most recently-developed high-resolution (16-bit and more) A/D and D/A converters have serial rather than parallel connections. The System One Dual Domain serial port can be easily interfaced to these converters. The rear-panel parallel input and output ports are casily interfaced to parallel-connection converters and to other DSP units.

Two Dither Types

Rectangular probability distribution dither has been added as a user selection for digitally-generated signals, in addition to the original triangular probability distribution dither. Each has its own advantages in terms of S/N degradation versus reduction of noise modulation effects.



Data, Save Test! The SAVE DATA command of S1.EXE software converts any data presently in memory into ASCII values, then saves that ASCII file to disk under the name you supply. The results might look like this for a short, four-point (three step) sweep:

Hz	deg	dBu
20,	10.1750002,	4.52589495
200,	6.09999999,	7.25721348
2000,	-2.09999999,	7.54252103
19999.916,	-19.6000004,	6.84135187

As an ASCII file, the .DAT file can be viewed with almost any editor or the DOS "TYPE" command.

The SAVE TEST command saves the test data plus complete control information for System One hardware at the time of the test—generator level and impedance, analyzer function, detector type, filter selections, settling panel parameters, REGULA-TION mode parameters, etc. SAVE TEST also saves any comments you had in the EDIT COM-MENTS buffer at the time of saving. A .TST file is a binary file and cannot be usefully examined with an editor.

From those descriptions, it is easy to understand why some users select SAVE DATA since it seems it might be more economical of disk space. However, an analysis of structure and size of the two file types shows, in terms of actual space used on a typical hard disk, this is almost never true!

The size of a .TST file with no data, no comments, and the standard AUDIO PRECISION graph title is 1,660 bytes with version 2.00A software. That 1,660 bytes principally consists of all the instrument and test condition setup data. Each data point in the test adds 13 bytes (except that the first point adds 20 bytes) to the file size. It does not matter whether both DATA-1 and DATA-2 are in use, since space is reserved for both. In contrast, a .DAT file contains a byte for every character including each space which separates the data values. A zero-data .DAT file is 64 ytes and every data point adds 63 bytes. Again, file size does not depend upon whether DATA-2 is in use.

and a .TST file length is approximately .TST = 1660+13n bytes

You can thus determine that a .DAT file is smaller for 32 data points or less, and a .TST file is smaller for more than 32 points. However, PC-compatible computers don't save just the exact file size. Hard disks on most PC-compatibles are divided into 2k (actually, 2048 byte) sectors. Save a one byte file or a 2048 byte file, and either will occupy a 2048 byte sector on the hard disk. So, here is how it works out: for 29 data points or less, both .DAT and .TST files are less than 2048 bytes and will occupy a single 2k sector. For 30 or 31 data points, the .DAT file will still fit in one 2k sector while the .TST file occupies two sectors. At 32 points the .DAT file also takes two sectors, and at 33 points and up th .TST file will be smaller than the .DAT file.

Finally, the big advantage of SAVE TEST rather than SAVE DATA is not just minimal disk space,

So, a little math says a .DAT file length is .DAT = 64 + 63n bytes



but the fact that the data is much less ambiguous since all the test conditions are saved with it. A .DAT file in dBr units gives you no idea what the dBr reference was. A .DAT file in dBm can't tell you what dBm impedance reference value was in use. You can't tell from a .DAT file what the generator amplitude was during a frequency sweep, which detector and filters were in use in the analyzer, or whether REGULATION or EQSINE modes were used. All that information is explicit in the .TST file format, directly readable in PANEL mode.

So why is there a SAVE DATA command at all? Purely for one reason—data transfer to other software programs. System One's .DAT file fits a standard format throughout PC software with its ASCII characters, comma delimiters between columns, and a carriage return-line feed at the end of every row. This format can be directly imported into Lotus 1-2-3 as a .PRN file and into most other spread sheet programs, into BASIC programs, or into almost any other software environment. Even if you do 1/lan to eventually export the data into Lotus 1-2-3, that still doesn't mean you must SAVE DATA at the end of the test. You can always use the more explicit (and usually more efficient) SAVE TEST command, then at any later time LOAD TEST, then SAVE DATA. In the interim your information will usually occupy less space on the disk and won't run the risk of becoming "orphan data" where you have the numbers, but no longer remember just what they mean.



Orders, Sales, Staff, Facility Growth

Thanks to the continuing support of our users, Audio Precision has just completed a very successful year in 1989. Our sales grew 44% over 1988; orders were up by 40%. 55% of 1989 orders came from International customers and 45% from U.S. domestic customers. The new DSP modules for System One were a substantial contributor to the order and sales growth.

That increased production also had its impact on our staff and facilities. Audio Precision now consists of 22 full-time employees plus some part-time help and

our continuing sub-contracted circuit-board assembly. In September 1989, we expanded into additional space at our same location. This expansion involved doubling space to 12,000 square feet (1,115 square meters). Further expansion of the Engineering staff is planned in 1990 in order to increase our rate of nc \prime product introduction. Manufacturing staff will increase as demanded by increases in order rate—test us to see if we can keep up!



Improving Speed Via Buffers Command Most users realize that the operating speed of System One depends upon their PC's processor type, clock rate, and presence of a math co-processor. An AT-compatible (80286-based) may typically run tests and procedures 30-40% faster than a PC-compatible (8088). The math co-processor (8087, 80287, etc.) has a major effect in dB conversion, log computations, and display speed; five to ten times improvement in the speed of an <F7> re-graph may result from use of the co-processor.

Not everyone realizes that a standard DOS configuration facility called BUFFERS, if not properly used, may slow down System One testing by a margin of two-to-one or worse. A BUFFER is a section of memory set aside for disk operations. When a disk read or write is called for, DOS checks to see if the information is already in a buffer; if so, no disk access is needed. The default value for number of BUFFERS is only two or three, which is quite inadequate for optimum operating speed with S1.EXE software. The symptom of an inadequate number of buffers is very slow access operation to a hard disk, similar to the operation of a nearly-full hard disk or one with badly-fragmented files. If a simple SAVE TEST operation requires several seconds of hard disk activity, check to be sure that you have an adequate number of buffers. Check by using S1.EXE's EDIT COMMENTS facility, an editor, or the DOS "TYPE" command to examine your CONFIG.SYS file, found in the root (C:\) directory of your hard disk or on your bootable DOS diskette. If there is no BUFFERS= statement (or no CONFIG.SYS file), you are operating with the default two or three buffers.

An optimum value for S1.EXE and many other programs is about 10-15. More than 20 buffers may actually begin to slow the system. Each buffer will reduce the amount of memory available to other applications by 528 bytes. To set buffers to the desired value, simply include a command such as:

BUFFERS=10

in your CONFIG.SYS file. You can edit your CON-FIG.SYS file (or create one if none exists on your computer) using the EDIT COMMENTS facility of S1.EXE.



Important Save Graphics Information

The version 2.00A User's Manual should have been more explicit in the HARD COPY chapter, describing use of the new SAVE GRAPHICS command plus PLOT.EXE or POST.EXE to obtain high-resolution graphs from plotters and laser printers.

When S1.EXE software has been started with the /G command line option to enable graphics reporting, a graphics display list called APLAST\$\$.GDL is created on disk each time a graph is displayed. Using the SAVE GRAPHICS command and furnishing a file name then copies that graphics display list disk file to your new name.

The .GDL file created by SAVE GRAPHICS will contain the last graph displayed. If, for example, you have just started S1.EXE software, loaded a test (but not displayed the graph) and immediately used SAVE GRAPHICS, the .GDL file will consist of the Audio Precision logo seen each time the software starts. To load a test and immediately send the graph to a .GDL file, you must first use <F7> (or the equivalent RUN GRAPH menu command) to graphically display the data before using SAVE GRAPHICS. The new graphics information is written into the graphics file, replacing whatever was previously in that file, whenever an <F7> or <F9> key operation causes a graph to be displayed.

Version 2.00A Software Upgrade Shipped

More than 1,500 copies of version 2.00A of S1.EXE were shipped to U.S. customers and International Distributors between January 31 and February 3, 1990. This free upgrade consists of a new 306-page User's Manual and four diskettes. The four diskettes include the S1.EXE v2.00A program diskette, a Tests and Procedures diskette, a Performance Check diskette, and a Utility diskette. Only one version of S1.EXE is now required, rather than the separate CGA and Hercules versions of previous revisions. S1.EXE now automatically determines the type of computer display system and installs itself accordingly. It supports the full resolution and larger color palettes of VGA and EGA systems in addition to Hercules and CGA. Owners of System One + DSP and System One Dual Domain will also receive a version 5 DSP program diskette and the 72-page DSP User's Manual.

In addition to new high-resolution multi-color display capability, v2.00A includes user control of print-screen graph height and width, HP LaserJet printer support, portrait or landscape mode graphs, graphs to HPGL pen plotters (multi-color) and Laser-Jet or PostScript laser plotters with resolution unlimited by the computer display system, graphics cursors, sub-procedures and a GO TO command, a procedure feature permitting menus within procedures for user test-type selection, an averaging mode for noise reduction, REGULATION by varying generator frequency, improved MAXIMUM and MINI-MUM algorithms in REGULATION mode, improved status display on the HELP screen, improved memory-usage options, log graph end points of 3 and 8 in addition to 1, 2, and 5, and a variety of other improvements.

If you haven't received your free upgrade yet, contact Audio Precision or your Audio Precision Distributor.

Why doesn't COMPUTE NORMALIZE work for phase data?

Compute Normalize vs Compute Delta



The COMPUTE NORMALIZE menu utility is highly useful to "push" amplitude data up or down on a graph so that it goes through a point you specify. Here is a common applications example: you have just measured frequency response of a CD player, using a test disc. The left channel data (graphed as DATA-1) goes exactly through zero dB at 1 kHz because you pressed the <F4> key while a 1 kHz signal was playing, just before the response sweep. But, the right channel (DATA-2) is above or below zero, depending upon the channel imbalance of the CD player. So, you use <Esc> to go to the menu, select COMPUTE NORMALIZE, and enter the numerical arguments "2,1k,0" meaning normalize DATA-2, do the normalization at the horizontal value of 1 kHz, and push the data up or down so that it goes through the 0 dB target at the horizontal value of 1 kHz. Now you can press <F7> for a regraph and both DATA-1 and DATA-2 nicely go through 0 dB at 1 kHz.

If you ever try COMPUTE NORMALIZE with phase data, you'll be disappointed. The data will go through the value you specify at the horizontal point you specify, but the overall phase curve shape will change with surprises at all other horizontal values. Why this happens, and what to do instead, is the subject of this article.

COMPUTE NORMALIZE works by dividing and multiplying, and it works on data in the "base unit". In all amplitude measurements, the base unit is Volts. In phase measurements, the base unit is degrees. When you made your CD player response measurement graphed in dBr (or any other decibel unit), S1.EXE software converts the base unit voltage numbers to dB for display. When you gave the COMPUTE NORMALIZE instructions, here is what S1.EXE did: 1. Converted your 0 dB target into a value in the base unit, Volts.

2. Obtained the original data value at 1 kHz in the base unit, Volts. If the data does not contain a point exactly at the requested horizontal (SOURCE-1) value, it will interpolate between the two horizontal points which bracket that value. If there is no data bracketing the value, it will display an error message.

3. Divided the user-entered target value by the data value.

4. Multiplied every data point (in base units) in the file by the result of the division in step 3.

So, what looks like a simple dB addition or subtraction with amplitude data is really obtained by division and multiplication. Now you can see why this won't work with phase data.

Let's say the original phase data varied between +20 and -70 degrees across the audio band, with a +5 degree value at 1 kHz. You wanted to normalize the phase curve to 0 degrees at 1 kHz. COMPUTE NORMALIZE will divide your 0 degree target by the +5 degree actual data value, get 0 for the answer, multiply all data points by 0, and you're surprised. Or, if you asked for a 10 degree target, the division quotient will be 2 and the resulting phase data after multiplication will vary between +40 and -140 degrees—a completely different curve shape.

The solution for phase normalization with S1.EXE is use of the COMPUTE DELTA utility. COM-PUTE DELTA is a subtraction utility, and it works in the units currently displayed. For the case just described of phase normalization by 5 degrees, you'll need to create a single-point file with the value 5 degrees, and save it to disk. To do this, load (or create) a .TST file with SOURCE-1 as a frequency parameter such as GEN FREQ, and with DATA-1 selected as ANLR PHASE (or LVF-1 PHASE with 1.60 or earlier software) in degrees. Then use <Esc> EDIT DATA to enter the data editor. Leave the top header row of text intact (Hz, deg, OFF), but delete any data below that line by use of the <F6> key. Then, enter any frequency (the value won't matter) in the first column, a comma, and 5 (or the phase offset value you need) in the second column. The result will look like this:

Hz deg OFF 100,5

It's not important that the numbers line up beneath the headers, but the comma is critical as a column delimiter. Now, use <Esc> to get back to the menu, then SAVE TEST and supply a file name such as 5DEG.

Now, it's time to re-load the phase-vs-frequency test whose data you wanted to move down by 5 degrees. Use <F7> to graph it to be sure it's the file you want. Then, <Esc> NAMES DELTA and use the cursor to select your file name (5DEG.TST in this example), followed by <Enter> to make the selection. Now, use COMPUTE DELTA. The software will try to interpolate the DELTA file data for every frequency in your phase sweep, but since the DELTA file only contains one value, it will come up with 5 degrees as the result of each "interpolation". So, it will subtract 5 degrees from every data point. Use <F7> to re-graph the data and you should find your original phase curve shape, but now offset vertically to go through the desired target.