Measuring Power Amplifier Damping Factor

Damping factor is a commonly specified parameter of power amplifiers. Defined as the ratio of a reference impedance (usually 8 Ohms) to the output impedance of the amplifier, it is a measure of the amplifier's ability to source and sink load current without changing its output voltage. Real-world loads are typically reactive in nature. These energy storage elements create a back EMF which can force spurious currents back into the amplifier's output.
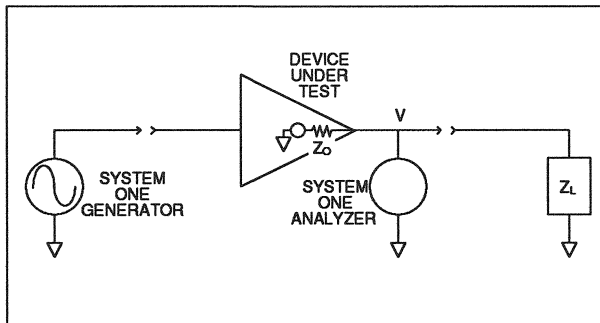

Figure 1 Test Setup, Basic Damping Factor Measurements

$$V_C = V_O \left[\frac{Z_L}{(Z_O + Z_L)} \right]$$

$$Z_O = Z_L \frac{(V_O - V_C)}{V_C}$$

$$DAMPING FACTOR \quad \zeta = \frac{Z_L}{Z_O}$$

$$\zeta = \frac{V_C}{V_O - V_C}$$

where Z_L = load impedance, Z_O = amplifier output impedance, V_O = open circuit output voltage, V_C = closed circuit output voltage

Figure 2 Damping Factor Formula Derivation

High damping factor (low output impedance) allows the amplifier to attenuate the disruptive effect of these return currents, thereby more accurately controlling the load.

Damping factor is measured by calculating the amplifier output impedance from measurements of the loaded and unloaded output voltage for a given frequency and load impedance. Refer to Fig. 1, for the test setup, and Fig. 2, for the relevant formula. Using System One, this can be accomplished with two successive frequency sweeps (one loaded, one unloaded) and a small BASIC program or other off-lir.: computation to calculate the damping factor from the data. The new data could then be regraphed with dimensionless units (X/Y) to display damping factor versus frequency.

Mike Elliott of Counterpoint Electronic Systems, Inc., Vista, CA, (America's Climatic Wonderland) suggested an alternative method which requires a second amplifier, but graphs the measurement in a single real-time sweep. Usually, the other channel of a stereo amplifier can be used for the second amplifier. See Figure 3 for a connection diagram and Figure 4 for the formula derivations. The test panel setups are shown in Figure 5. The reference load is bridged between the two amplifier outputs, and the ratio of the two output voltages is measured directly in TWO-CHANNEL mode. This is also a
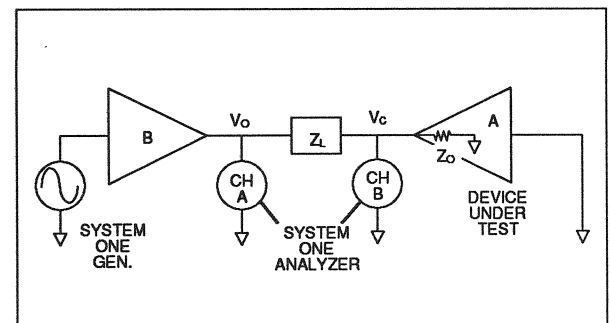

Figure 3 Test Setup, Real-Time Damping Factor Measurements

$$V_C = V_O \left[\frac{Z_O}{(Z_O - Z_L)}\right]$$

$$Z_O = Z_L \left[\frac{V_C}{(V_O - V_C)}\right]$$

$$\zeta = \frac{(V_O - V_C)}{V_C} = \frac{V_O}{V_C} - 1 \approx \frac{V_O}{V_C}$$

Figure 4   Formula Derivation, Two-Amplifier Measurement Method

very intuitive arrangement, since any voltage appearing at the output of amplifier A (device under test) is clearly an error caused by current from the load. It should be noted that most modern amplifiers have damping factors significantly greater than one. Hence, the damping factor is approximated simply by the ratio measurement itself (ignoring the "-1" term), with no external calculations required. In TWO-CHANNEL mode, the denominator of the ratio (in this case the output of amplifier A) is measured with the LEVEL voltmeter, which is fully specified down to 10mV and quite usable to 1 mV. The B amplifier should therefore be driven to a sufficiently high level output level (10 Volts, for example) so that the voltage at the output of the device under test (amplifier A) remains above 1 mV at all frequencies. This may be accomplished by changing the GENERATOR AMPLITUDE or adjusting the gain control of amplifier B. Note that LEVEL must be selected as DATA-2, even though the DATA-2 GRAPH TOP and BOTTOM are deliberately chosen so that the LEVEL measurement does not graph. This is a "workaround" to eliminate a "bug" discovered during preparation of this article.

Figure 5   Panel Set-Up for Two-Amplifier Method

A typical damping factor versus frequency graph result is shown as Figure 6.
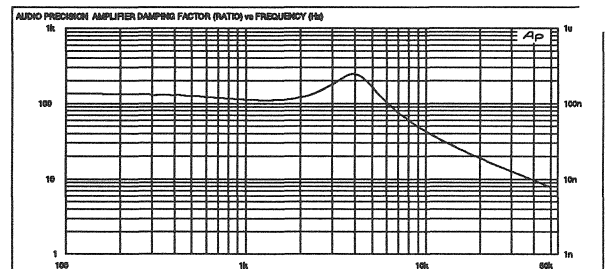
Tony Dal Molin

Figure 6   Example Graph of Damping Factor, Real-Time Two-Amplifier Method

Bi-Directional Printing

Some dot-matrix printers, including the Epson FX and LQ series, are strictly uni-directional in graphics mode. They print only while the print head moves from left to right across the page, then move the print head back to the left edge without printing. Other printers, including the Epson EX series, can operate in either uni-directional or bi-directional modes. Faster print-out will result with these printers in bi-directional mode, but the print quality is likely to be poorer since the vertical columns of dots produced when moving left-to-right may not exactly line up with the dot columns printed from right-to-left.

Version 2.00 of S1.EXE software permits the operator to select uni-directional or bi-directional mode, if his printer is capable of bi-directional printing. This selection is done when starting S1.EXE software, with the second digit of the /F (formatting) option (the first digit enables or disables form feeds and most formatting). If the /F option is not used, simply starting the software by:

S1 <Enter>

the software sends no mode command to the printer before each graph. Bi-directional printers are normal-

ly in bi-directional mode when turned on, so graphs will print bi-directionally. If the printer does not have bi-directional capability, it will of course print uni-directionally. Using the /F option with "1" as the second digit, such as:

S1 /F1,1 <Enter> or S1 /F0,1 <Enter>

has the same effect on printer directionality as not using the /F option at all. No code is sent to the printer, so it prints graphs in its normal power-on mode.

If the /Fn,0 option is used to force uni-directional printing by starting the software:

S1 /F1,0 <Enter> or S1 /F0,0 <Enter>

bi-directional printers will be forced into uni-directional mode in order to produce higher-quality (but somewhat slower) graph print-out. This option will have no effect on uni-directional printers since they are always uni-directional.

Tabular data is unaffected by the /F options and will be printed uni-directionally or bi-directionally depending upon the printer power-on settings or control panel.

A method for testing broadcast-quality audio transmission links has been standardized as CCITT specification O.33, based on the European Broadcasting Union (EBU) Recommendation R-27. The test signal sequence transmitted into the link consists of a one-second burst of FSK (frequency-shift keyed) information, followed by a variety of tones or silent periods. Total duration of the sequence is 30 seconds for monaural links and 32 seconds for stereo links. At the receiving end of the path, the FSK burst must be de-coded to determine which of 100 possible test sequences is to follow. Neither data communications link nor modems are required; the one-way FSK transmission down the audio link itself is the only control information. The receiving analyzer then, using the FSK burst as a synchronizing signal, measures and logs various performance parameters including absolute level, frequency response, distortion (at two frequencies), crosstalk and inter-channel phase if a stereo transmission is involved, amplitude linearity (to test compander accuracy), and noise. The FSK burst also carries information identifying the transmitting location. At the present time, only two test sequences (one monaural and one stereo) have been defined by the CCITT standard.

Audio Precision's German Distributor, RTW GmbH, has recently completed development of a hardware module and software programs to permit generation and measurement of the CCITT O.33 sequences by System One (SYS-22A). The special software causes the standard System One generator to generate the FSK sequence. The hardware module is a circuit board which mounts into the lower right-hand compartment of System One. This circuit board performs FSK de-coding and also provides the filters necessary for the distortion measurements. This hardware can be installed in new systems at time of order, or can be retro-fitted into any SYS-22A. It is not compatible with the early non-A versions (serial numbers below 20300). Figure 1 is an example of the report format following reception and measurement of the CCITT 01 (stereo) sequence.

The software permits the user to define acceptable limits of performance for each measurement and will then highlight out-of-spec measurements on the computer screen and on the print-out to an attached printer. The software also permits totally automatic operation of the system. For example, it can be set up to make automatic transmission measurements during night or early morning hours when the link is not being used for normal programming. A test transmission can be made once or automatically repeated at defined intervals. The software can control SWR-122 switchers, so an automatic series of transmissions can be made from a network control point into a number of different transmission paths. At the reception and measurement end, the software permits saving measured data to disk and printing, or both. The full data or errors only may be printed.

The software furnished with the new transmission measurement capability not only supports the two defined standard (mono and stereo) test sequences, but also makes it easy for any user to create his own custom test sequences. Thus, a broadcaster can not only measure international standard test signals sent during the set-up period prior to international broadcasts, but may use the capability internally within his facility or network. He may choose to do faster testing or more thorough testing, as appropriate for his application. Creation of these special sequences is done via simple menus and interactive screen editors very similar to standard S1.EXE software, and requires no programming experience or knowledge of programming languages.

For more information and price quotations on the CCITT O.33 capability, contact Audio Precision or your Audio Precision International Distributor.

```
IDENTIFICATION = RTW        SEQUENCE = 01                    03.04.1990   10:52:19

Ref.Level Left      Right             Frequency  Level     Dist.
1020 Hz   [dBu]     [dBu]                [Hz]     [dBr]     [dB]
          -2.83     -2.82                1020     8.98      -74.5    A ak2
                                         1020     8.99      -74.6    B ak2
Frequency Left      Right     Phase    PAUSE
   [Hz]   [dBr]     [dBr]     [deg]       60      9.00      -75.3    A ak3
   1020   -11.97    -11.96    -0.1        60      9.01      -78.3    B ak3

     40   0.01      -0.03     -0.1     Frequency  Level     Crosstalk
     80   0.02       0.02      0.0        [Hz]     [dBr]     [dB]
    200   0.02       0.02      0.0       2040     -11.99    -62.18
    500   0.02       0.02     -0.1       2040     -12.00    -68.55
    820   0.01       0.01     -0.1
   2000  -0.00      -0.01     -0.2     Linearity  Left      Right
   3000   0.04       0.03     -0.3        [Hz]     [dBr]     [dBr]
   5000  -0.00      -0.03     -0.3       800      5.98      5.99
   6300   0.04      -0.01     -0.2       800     -6.05     -5.98
   9500   0.03      -0.03     -0.1       800      5.98      5.99
  11500  -0.02      -0.08      0.0
  13500  -0.01      -0.04      0.1     NOISE A        -77.8   dBqpsmax
  15000  -0.03      -0.06      0.1     NOISE B        -75.2   dBqpsmax
```

Figure 1 Example Report, CCITT O.33 Sequence 01 (Stereo) Audio Transmission Line Measurement

The System One generator has two classes of amplitude units; voltage & power. Volts RMS, Volts Peak, dBu, and dBV are voltage units. Watts and dBm are power units. The dBr (dB relative) unit may be either voltage or power, depending on what type of unit is in use at the dBr REF field near the bottom of the generator panel. If a Watts or dBm value is entered as the dBr REF, dBr will be a power unit; with any other unit at dBr REF, dBr will be a voltage unit.

Voltage Units—Open Circuit Voltage Calibration

When a voltage unit is used, the GENERATOR AMPLITUDE field directly specifies the generator open-circuit voltage (sometimes called emf, or electro-motive force). Unless the external load impedance is infinite, the actual voltage delivered to the load will be less than this open circuit voltage since the selected generator source impedance and the external load impedance act as a voltage divider. For example, if the 50 Ohm source impedance is in use and the generator output is connected to the analyzer input with the analyzer 100 kilohm input termination selected, the voltage across the load will be

$$\frac{100,000}{100,000 + 50} = 99.95\%$$

of the open circuit voltage. With the 50 Ohm source impedance in use, the voltage across the load will differ from the open-circuit voltage by no more than 0.1 dB if the load is approximately 5,000 Ohms or higher. For a 10,000 Ohm load (typical of bridging inputs on many professional audio devices), the loading of the 50 Ohm source will produce approximately an 0.05 dB error.

Power Units—Computer Needs More Information

The power units available for the generator are dBm & Watts; Watts units are rarely used as the generator amplitude unit. Neither the generator nor computer can directly measure or control generator output power, output current, or external load resistance. The computer and software have control only of generator open-circuit voltage. Therefore, in order for a value in dBm or Watts entered by the user into the generator AMPLITUDE field to be valid, the operator must first determine the value of the external load and enter that value into the W/dBm REF field near the bottom of the generator panel. The computer, knowing the value of source impedance presently selected on the generator, then uses the W/dBm REF field value to compute what the open-circuit voltage must be set to in order to produce a voltage across the load resulting in the specified power in the load. If the source impedance is changed to another available value, the computer re-calculates the equation and re-sets the open-circuit voltage so as to maintain the specified power in the load. For example, setting 0 dBm into a 600 Ohm load (0.7746 Volts across the load) from a 50 Ohm source load will produce a generator open-circuit voltage of

$$0.839V = 0.7746V \ \frac{600 + 50}{600}$$

Changing the source impedance to 600 Ohms causes the open-circuit voltage to go to

$$1.549 \ V = 0.7746 \ V \ \frac{600 + 600}{600}.$$

Loading and One vs Two Channels

In dBm mode for generator AMPLITUDE, the computer sofware assumes that both generator output connectors A and B are loaded with the external value specified in the W/dBm REF field if the A&B or A&-B generator output selections are made. It assumes only one output is loaded if either A or B output selections are made. Violating these assumptions will result in an amplitude calibraton error.

Why dBm?

In the early years of broadcasting and professional audio, audio circuits with matched terminations and maximum power transfer were the common case in studios and for audio transmission lines between facilities. Console and line amplifier output impedances, implemented with vacuum tube and transformer technology, were typically 600 Ohms. Equipment input impedances, again usually transformer-matched, were also typicaly 600 Ohms. Maximum power transfer takes place when the source and load impedances are matched. For such systems, the dBm unit (dB relative to one milliwatt) was appropriate since it is a power unit.

Audio Meters

However, actual power-measuring instruments are extremely rare in audio. Audio meters and distortions analyzers are *voltmeters*, measuring voltage across their input terminals. They do not know the power level, current value, nor source impedance across which they are measuring. Since the broadcasting and pro audio industry had "grown up" with 600 Ohm power-transfer systems in common use, audio test instrument manufacturers typically calibrated their voltmeters for this situation. Most audio test instruments and systems manufactured before approximately 1985 used the dBm unit on their meter scales and switch labels, calibrated with the assumption that the meter would always be connected across a 600 Ohm circuit when measuring dBm. Since the voltage across a 600 Ohm resistor is 0.7746 Volts when one miliwatt is being dissipated in that resistor, the meters were calibrated for a zero "dBm" indication with 0.7746 Volts applied. But, they were not measuring power; change the circuit impedance, and the meter is lying.

Voltage-Based Systems

M\)dern broadcasting and pro audio equipment normally has output impedances much lower than input impedances. Output impedance values from zero up to 50 Ohms are typical, and input impedances of 10 kilohms are typical. Such equipment, connected together, transfers negligible power due to the large impedance mis-match. However, nearly all the source voltage is transferred. As noted earlier, a 10 kilohm load reduces the open-circuit voltage from a 50 Ohm source by only 0.5%, or 0.05 dB. Thus, modern systems typically operate on a voltage transfer basis and the dBm, as a power unit, is not appropriate. A proper unit for voltage-based systems is the dBu (dB relative to 0.7746 Volts). The dBu is a voltage unit and requires no assumptions about current, power, or impedance. Those older audio meters calibrated in "dBm" are really dBu meters.

Old Habits Die Hard

Unfortunately, the "dBm" terminology has hung on long after its use is generally appropriate. Even some of the most-competent manufacturers of high-technology digital and analog professional audio equipment still use the dBm unit in their setup instructions. Users are told to apply an input signal of "+4 dBm" and then to adjust trim pots for an exact 0 VU indiction on a 24-track digital audio tape recorder, for example. Yet, the line input impedances of that tape recorder are 10 kilohms. What the manufacturer clearly wants is a +4 dBu input level (1.22 Volts). If we truly applied +4 dBm to that 10,000 Ohm input, the resulting 5.0 Volts would probably not even be within the trim pot adjustment range for 0 VU. So, a good general rule when working with modern audio equipment unless you know it to be terminated in 600 Ohms is to read the manufacturer's "dBm" as "dBu". Even if in doubt about the actual termination value, you won't go very far astray in using dBu with the System One generator's 50 Ohms source; 600 Ohms loads a 50 Ohm source by about 0.7 dB.

Estimating Input Impedance (How Can I Find Out If My Circuit Is Terminated?)

The input impedance of an audio device can be quickly estimated by use of the generator 600 Ohm output impedance, GEN MONITOR connection, and dBr units on the analyzer LEVEL meter. With the generator output set for a nominal value such as 0 dBu from 600 Ohms, no load connected to the generator connectors, and GEN MONITOR selected, press the <F4> key to set the analyzer display to 0.00 dBr. Then, connect the unknown input to the generator output. If the external device input impedance is 600 Ohms, the analyzer reading will decrease to -6.02 dBr as half the generator open circuit voltage is dropped across the internal 600 Ohms source. If the drop is 0.06 dB or less, the input impedance is 10,000 Ohms or higher (bridging).

Analyzer Input Terminations and the GEN MONITOR Path

Confusion sometimes exists concerning the relationship between the System One analyzer's selectable input terminations and the GEN MONITOR internal cable path between generator output terminals and analyzer input. The GEN MONITOR path is never terminated in an impedance lower than 100 kilohms, regardless of analyzer input termination resistance selection; see Figure 1. The purpose of the analyzer input terminations is to properly terminate the output of the external device being measured. That termination should remain connected to the device being tested even while the analyzer is temporarily checking the generator output amplitude via the GEN MONITOR cable connection. The principal purpose of the GEN MONITOR path is to permit the analyzer to measure the actual generator output terminal voltage with as little upset of the present conditions as possible. Thus, the GEN MONITOR path to the analyzer input must present as little additional loading across the generator as possible, which means using the analyzer's highest (100 kilohm) input impedance when monitoring the generator.

If you wish to use the analyzer's input terminations in a self-tutorial examination of the impedance estimation and dBm setting concepts discussed above, you must therefore connect a cable between the generator output connectors and the analyzer front-panel input connectors. You may then, by various selections of generator output impedance and dBm and Volts (open-circuit) units on the generator, and various selections of dBu, dBm, and Volts units on the analyzer LEVEL meter and READING meter (in AMPLITUDE mode) plus terminated and 100 kilohm analyzer input selections, test the above concepts until they become intuitive.

EURO-Z Option

A special, optional version of System One hardware and software is available to comply with certain broadcasting standards in use in a number of European countries. Even though some of the generator output and analyzer input impedances are different in the EUR-Z option, the same mathematical concepts are used in the software and all the above discussion pertains.
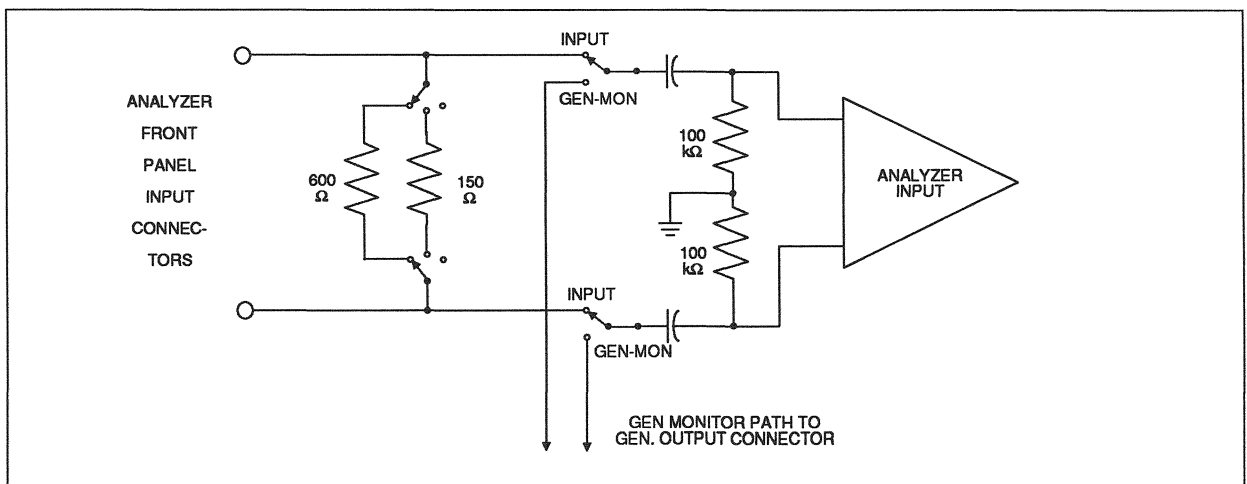

Figure 1 Equivalent Circuit, Analyzer Input Terminations and GEN-MONITOR Path

Most System One users sooner or later discover the limitation that only 88 file names can be displayed on screen by any of the LOAD commands. This limitation most commonly appears with test (.TST) files. The only good solution to this problem is to segregate files into different sub-directories so that no more than 88 files of any one type are ever located in one sub-directory. For example, when first loading System One software onto a new computer, create a directory in the root called AUDIO (or SYSONE or AP or other appropriate name; in this article, we'll assume you used AUDIO). Copy the "S1.EXE" diskette, "Utilities and Equalization" diskette, and "Test and Procedures" diskette into this subdirectory. Then, use the MD (Make Directory) DOS command to make several more subdirectories below this directory. For example, make one called PERFCHEK (C:\AUDIO\PERFCHEK) and copy the "Performance Checks" diskette into it. If you have a DSP unit, make C:\AUDIO\DSP and copy the "DSP Tests and Programs" diskette into it. If you have the Compact Disc player testing applications note and diskette, make another directory named C:\AUDIO\COMPDISC and copy the app note diskette into it. Each time you undertake a project which is likely to result in a number of new files, make a new directory and move into it for all testing and file saving on that device.

Changing Directories

Now that your audio testing files are better organized, you will frequently need to change back and forth between these directories. If you remember the names of all your directories, are comfortable with DOS commands, and are a touch typist, you can easily do this with S1.EXE's DOS menu command and a few characters. For example, to move to the C:\AUDIO\DSP subdirectory from within S1.EXE, press <D> for DOS, then type:

CD \AUDIO\DSP <Enter>

The "CD" means "change directory"; "\AUDIO\DSP" is the full path name of the directory into which you want to move.

Simplifying It

To make subdirectory changing simpler, especially when there may be several System One users who do not remember all the various subdirectory names or are not proficient with DOS, you can create a number of procedures (.PRO files) to simplify the process. The procedure files will consist only of the change directory command and the full path name of the subdirectory to be moved into. Using the full path name guarantees that the procedure file will work from any subdirectory on the disk.

To make a procedure, use Util Learn mode. First, use the Edit Procedure command and delete any procedure presently in the editor. Then, use the Util Learn command to start "learn mode". Press <D> (for DOS). Type the command

CD pathname <Enter>

where "pathname" is the full path name of the directory to be moved into. For the principal audio testing sub-directory created above, the command would thus be

CD \AUDIO <Enter>

Then, use Util End to stop the learning process. Use the Edit Procedure command to view the results, which should look like this:

PROCEDUREv2.00
DOS CD \AUDIO /R
UTIL END

Use <Esc> to move back to the menu, Save Procedure, and supply the name "!AUDIO". Use of the exclamation point (!) character as the first character will cause this procedure to be sorted to the top of the list of procedures displayed when the Load Procedure command is used.

For your next procedure, you may either start again in Util Learn mode, or edit the middle line in Edit Procedure mode and then save the result. Editing in Edit Procedure mode is preferable, so that all the "!" procedures you make will be grouped into one subdirectory. If each is separately generated in Learn mode, each procedure will cause a sub- directory change as the procedure is being created with the result that each procedure will be saved in a different sub-directory. For the performance check directory, your finished procedure will look like this:

PROCEDUREv2.00
DOS CD \AUDIO\PERFCHEK /R
UTIL END

and it will be saved under the name "!PERFCHE.PRO". Make a similar procedure for each directory and name it with the directory name, except preceded with the ! character. For the example directories suggested above, you would end up with procedure files as follows:

!AUDIO.PRO
!DSP.PRO
!PERFCHE.PRO
!COMPDIS.PRO

Now, exit to DOS and use the DOS COPY command to copy all those .PRO files into every subdirectory you will be using. Since they all begin with the exclamation point, you can copy them all as a group by using DOS "wild cards". For example:

COPY !*.PRO newdestination <Enter>

From this point onwards, changing directories is as simple as

1. Using the Load Procedure (<L> <P>) command

2. Using the mouse or cursor keys to select the procedure corresponding to the directory you wish to move into; they will all be grouped at the beginning of the list of procedures due to the ! character, and will thus be easily distinguished from ordinary procedures

3. Pressing <Enter> and <R> <P> (or Run Procedure)

and you're in the new directory!

The measured power of gaussian (white) noise is proportional to the bandwidth of the measuring instrument. Measured power is reduced 2:1 (-3.01 dB) when the measurement bandwidth is cut in half, or 10:1 (-10.0 dB) for every 10x reduction in measurement bandwidth. If measurements are made as voltage, the voltage varies proportionally to the square root of the bandwidth. Thus, a 2:1 bandwidth reduction produces a 1.41:1 voltage reduction (still -3.01 dB). When selective measurements are made of gaussian noise, the measuring bandwidth must be known and computational corrections must be made in order to express the measurement in terms such as noise per unit bandwidth.

FFT Measurement Bandwidth

The effective bandwidth of an FFT analyzer such as System One + DSP or System One Dual Domain depends upon two factors; FFT resolution and the window function. FFT resolution depends upon the data record length and the sample rate. The number of bins (spectral lines) is one-half the number of signal samples used in the FFT transformation. Transforming 16,384 samples (the MAXIMUM selection if full memory is installed), for example, will produce 8,192 spectral lines or bins. All bins are equal width, and they are evenly distributed from dc (zero Hz) to 1/2 the sample rate (Nyquist frequency).

For a 48 kHz sample rate and a 16,384 sample data record length, for example, the 8,192 bins are evenly spaced from 0 to 24,000 Hz and the width of each bin is consequently 2.93 Hz. If the record length is changed from MAXIMUM to 4096, only 2048 bins will result and the bin width (at the 48 kHz sample rate) becomes 11.72 Hz. Gaussian noise measured with a 4096-sample record length will therefore produce a 4x higher power (+6.02 dB) than in MAXIMUM record length, since the bin width is 4x greater. If the record length stays at MAXIMUM (16,384 samples with full memory) but the sample rate is reduced to 1 kHz, the bin width becomes 0.061 Hz. Gaussian noise in this case would produce 48x less power than with the 48 kHz rate and MAXIMUM record length. At the 1 kHz sample rate, however, the system bandwidth is only about 350 Hz.

Noise Voltage Per Root Hz Bandwidth

To convert the actual measurement to the equivalent value for another bandwidth, it must be corrected for the ratio of the two bandwidths. To express a noise voltage in "Volts per √Hz bandwidth" when measured with the 2.93 Hz bandwidth of MAXIMUM record length, full memory, and the 48 kHz sample rate, divide the measured voltage by the square root of 2.93:1.

Window Functions and Noise

Use of a window function is based on the fact that an FFT algorithm assumes data before and after the transformed record to be an infinite series of exact replicas of the data record. This is rarely the case with coherent signals, since the signal period rarely exactly matches the record length. A window is a weighting function used to multiply each sample of the amplitude record. The window value is unity (1.000) at the center of the record, tapering off to a small value at the beginning and end. The resulting low amplitude at the ends of the record means that the records could be "spliced" end-to-end into an infinite series. The window thus reduces sharp discontinuities, but has the effect of broadening somewhat the signal spectrum. When noise is measured with an FFT analyzer using a window function, the result is "bin spillover"; noise energy in adjacent bins also has some effect on the amplitude of the measured bin.

When there are no coherent, discrete signals, or when such signals are only slightly above the noise level, FFT measurements can be made without use of a window function. There is then no "spillover" of energy from adjacent bins, and the noise bandwidth of each bin equals the theoretical value. Thus, using the FLAT window selection (no window; this name will be changed from FLAT to NONE at the next FFT program revision) produces noise measurements which do not need correction. Use of the HANN window produces voltage readings 1.22 times higher, and the BH4 window selection produces voltage readings 1.41 times higher, than with no window. If absolute noise values are desired, the voltage readings obtained with either of these windows must therefore be corrected by dividing by 1.22 (HANN) or 1.41 (BH4).

Spectrum Averaging

For best accuracy of noise amplitude measurements, use of the FFTGEN.DSP program and spectrum averaging is recommended. Spectrum averaging reduces the variance in noise and produces more repeatable and accurate values.

Screen Display Points vs FFT Bins

The screen display of noise amplitudes with System One FFT programs will be further affected by the #STEPS parameter of the SWEEP (F9) DEFINITIONS panel. Assume a MAXIMUM record length with full DSP memory, resulting in an 8,192 bin FFT. If #STEPS is set to 8192, each bin value will be sent as a point to the PC for plotting and each represents accurately the noise at each bin frequency, subject to the window corrections and bandwidth relationships described above. When the #STEPS is smaller than the number of FFT bins between the START and STOP frequencies, the DSP sorts through a set of adjacent FFT bins and sends the highest value of that set to the PC to be plotted. If #STEPS is set to 405, for example, there will be one value sent from the DSP to the PC for each set of approximately 20 bins. The DSP will search through the first 20 bins and send the highest value to the PC to be plotted as the first point; the highest value in bins 21 through 40 will be the second point plotted, etc. This "peak picking" routine guarantees that no high-amplitude coherent signal will be skipped over. With noise signals, this process prejudices the screen display toward higher values. For absolutely-calibrated measurements of noise, the #STEPS value should be set equal to the number of bins in the FFT (one-half the record length), the START frequency should be set to 0 Hz, STOP frequency to the Nyquist frequency (one-half the sample rate), and a LIN horizontal display should be used. If measurements are desired only over a narrow spectral range, choose the SOURCE-1 START and STOP frequencies ap-

7

propriately and make the #STEPS at least equal to the number of bins between those frequencies. There is no harm to using a #STEPS value larger than the number of FFT bins; the result is simply that each bin will be plotted as several horizontal points of the same amplitude value.

Selective Measurements via Analog Bandpass Filter

System One's analog bandpass filter is a constant-Q (constant percentage bandwidth) design. Its equivalent noise bandwidth is 1/3 octave, or approximately 26% of center frequency. For example, the noise bandwidth is 260 Hz at a 1 kHz center frequency. Thus, to correct a 1 kHz noise voltage measurement to Volts/$\sqrt{Hz}$, divide by the square root of 260.

Can S1.EXE Use Memory Above 640k?

Certain capabilities of S1.EXE software depend upon the amount of computer memory available. Specifically, the number of test points in a sweep which can be re-graphed via <F7> or saved to disk depends upon the memory available to S1.EXE when it loads. With approximately 590k bytes available, for example, S1 will normally load with capability for up to about 950 points. With only 490k bytes available, a normal S1 load will provide only about 190-point capability. For most non-DSP applications of System One, 100-200 points in a test are sufficient and memory is seldom an issue. Utilizing the full horizontal resolution of the computer display is often desirable when making FFT measurements, however; this requires about 500 to 550 points.

Since nearly all modern computers have 640k or more of memory, why should we have a problem providing S1.EXE with the 530k to permit 500-point tests? First, DOS itself fills a significant amount of memory when the computer is booted; about 60k is used by DOS version 3.1, for example. Secondly, many PC users have installed programs called TSRs (for Terminate and Stay Resident). Typical of these are the many menu-type programs which shield the user from having to type DOS commands. These programs provide simple and attractive menus and panels for changing directories, erasing files, adding new sub-directories, etc. These programs can be convenient, but they may occupy from 50k to 100k bytes of memory at all times. With one of these programs and several smaller TSR utilities loaded in addition to DOS, there may be insufficient memory available for S1.EXE to do what is needed.

With 286 and 386-based computers with several megabytes of memory now common at reasonable prices, users often ask whether S1.EXE software can access that additional memory. The direct answer is "no"; S1.EXE does not use any memory above 640k. But, with 386-based computers and one megabyte or more of memory, there is an approach which can significantly increase the amount of memory available to S1.EXE. This approach requires the purchase of either the 386MAX software package from Qualitas, Inc., 8314 Thoreau Drive, Bethesda, MD 20817 [telephone (301) 469-8848] or the QEMM package from Quarterdeck Office Systems, 150 Pico Boulevard, Santa Monica, CA 90405 [telephone (213) 392-9851]. Both are readily available at software stores and by mail.

Instructions with either of these packages describe how to configure CONFIG.SYS and AUTOEXEC.BAT files which execute automatically each time the computer is booted. These programs then have the ability to load most drivers and TSR utility programs into "memory holes" with addresses between 640k and 1 megabyte. Thus, you can have the advantage of convenient and powerful TSR utilities without using up the valuable DOS-addressable memory below 640k, leaving that lower-address memory available to S1.EXE. S1.EXE will then load into the available remaining space to provide maximum capabilities.

POST.EXE Modifications for Ventura Publisher Compatibility

Users of the Ventura Publisher desktop publishing program have reported problems when importing into Ventura Publisher the "encapsulated PostScript" graphic files created with the POST.EXE utility furnished by Audio Precision. The cause of these problems has been isolated and POST.EXE has been revised for compatibility with Ventura Publisher.

No features were added or changed in POST.EXE in this revision.

If you have experienced problems importing the .EPS files produced by POST.EXE, call or write Audio Precision or your Audio Precision International Distributor for a free copy of version 2.03 of POST.EXE.