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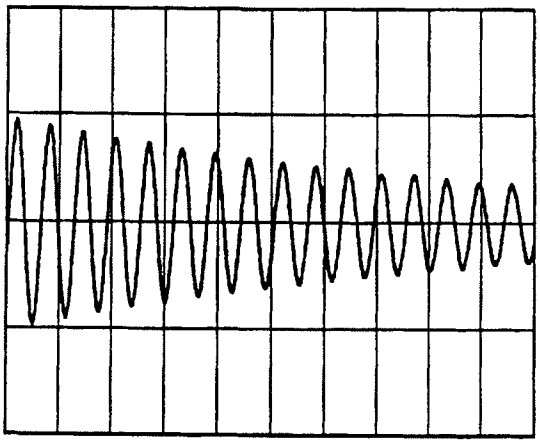
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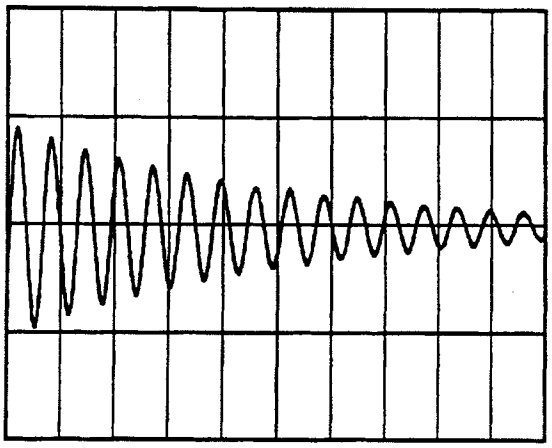
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Find the *Signal Processing Systems SPS390* at our website: ***Click HERE***

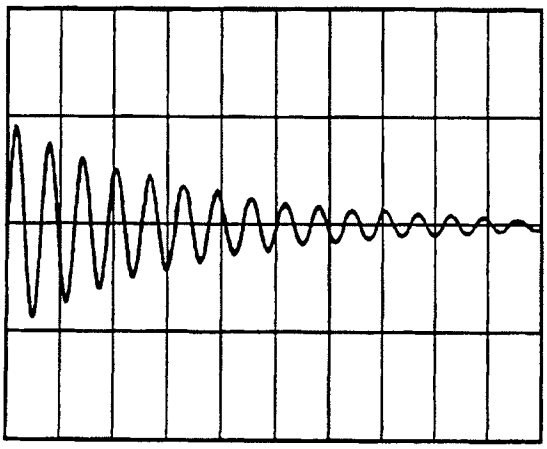
***Response Weighting With a TAU of 1. Last Cell Is Attenuated 1/e***



***Response Weighting With a TAU of 2. Last Cell Is Attenuated $1/e^2$***



***Response Weighting With a TAU of 3. Last Cell Is Attenuated $1/e^3$***

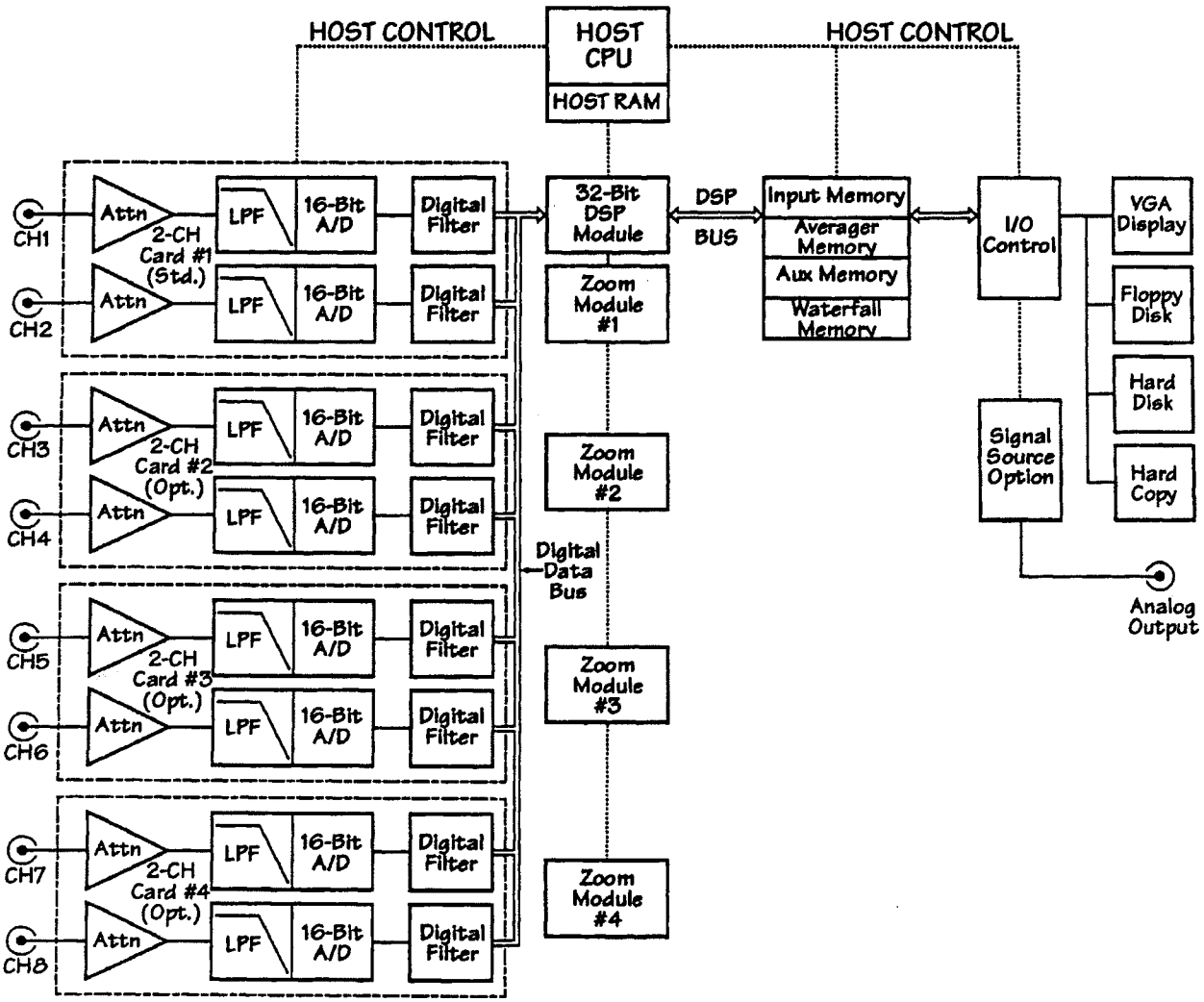




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## 4–2 FUNCTIONAL BLOCK DIAGRAM

### SPS390 Basic Block Diagram






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# 5–1 CLEANING

## 5–1.1 Chassis

**CAUTION**

**Always turn off the unit power and disconnect the power cord prior to cleaning any part of the chassis or components**

Use a clean, damp, lint-free cloth and avoid introducing any foreign matter or liquid inside the unit or inside externally exposed connectors.

### Cooling Fan Filter Cleaning

Avoid overheating of the unit by cleaning the fan filter periodically. The frequency of cleaning is dependent on the amount of dust and airborne particles present in the ambient operating environment.

Remove the filter clean dirt by rinsing the filter in water containing a mild detergent. Rinse the filter in clean water and thoroughly dry it before reinstallation on the unit.

### Display Screen Cleaning

Wipe the display glass with a clean, damp. Lint-free soft cloth. If a cleaner is necessary, use a commercial grade glass cleaner liquid. Avoid harsh chemical cleaners or any abrasive cleaner. Use a clean, lint-free cloth to dry the class after cleaning.



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## CAUTION

**Any existing configuration files with the same name as those listed in figure 5-1 will be destroyed. Protect any such files as necessary.**

Actual measurement results may be stored for archival purposes. This is not required but is a feature that would facilitate verification of field measurements should data validity questions arise. It is recommended that pre-measurement calibration verification data files be stored on properly formatted 3.5" removable-floppy disks and not on the internal hard drive.

If there is a requirement for an NIST or MIL SPEC calibration, contact Signal Processing Systems' customer service department.

Figure 5-1. Test Configuration files

| 2 Channels<br>17 Files | 4 Channels<br>18 Files | 6 Channels<br>27 Files | 8 Channels<br>28 Files |
|---|---|---|---|
| test001.cfg | test001.cfg | test001.cfg | test001.cfg |
| test002.cfg | test002.cfg | test002.cfg | test002.cfg |
| test005.cfg | test005.cfg | test005.cfg | test005.cfg |
| test011.cfg | test011.cfg | test011.cfg | test011.cfg |
| test012.cfg | test012.cfg | test012.cfg | test012.cfg |
| test019.cfg | test019.cfg | test019.cfg | test019.cfg |
| test021.cfg | test021.cfg | test021.cfg | test021.cfg |
| test022.cfg | test022.cfg | test022.cfg | test022.cfg |
| test024.cfg | test024.cfg | test024.cfg | test024.cfg |
| test025.cfg | test025.cfg | test025.cfg | test025.cfg |
| test035.cfg | test035.cfg | test035.cfg | test035.cfg |
| test203.cfg | test035a.cfg | test035a.cfg | test035a.cfg |
| test203a.cfg | test203.cfg | test101.cfg | test101.cfg |
| test204.cfg | test203a.cfg | test102.cfg | test102.cfg |
| test204a.cfg | test204.cfg | test105.cfg | test105.cfg |
| test205.cfg | test204a.cfg | test111.cfg | test111.cfg |
| test206.cfg | test205.cfg | test112.cfg | test112.cfg |
| | test206.cfg | test119.cfg | test119.cfg |
| | | test121.cfg | test121.cfg |
| | | test125.cfg | test125.cfg |
| | | test135.cfg | test135.cfg |
| | | test203.cfg | test135a.cfg |
| | | test203a.cfg | test203.cfg |
| | | test204.cfg | test203a.cfg |
| | | test204a.cfg | test204.cfg |
| | | test205.cfg | test204a.cfg |
| | | test206.cfg | test205.cfg |
| | | | test206.cfg |



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| Step | Action |
|---|---|
| 5. | Load test setup **test011.cfg**. Start the averager by selecting the START function key (F2) on the SPS390.<br>Verify that all tonals other than the 1 kHz primary tonal are at a maximum level of -70 dB on all displayed channels. |
| 6. | Load test setup **test012.cfg**. Start the averager by selecting the START function key (F2) on the SPS390.<br>Verify that all tonals other than the 1 kHz primary tonal are at a maximum level of -70 dB on all displayed channels. |
| 7. | Load test setup **test019.cfg**. Set the synthesizer for 1.25 Vrms at 1 kHz.<br>a. Verify that all displayed channels show an overload indication (indicated by the corresponding Channel Overload Indicator going to reverse video).<br>b. Decrease the input level to 1.0 Vrms.<br>c. Verify that all channels are no longer in overload |
| 8. | Load test setup **test021.cfg**. Set the synthesizer for 38 kHz at 1 Vrms. For each of the displayed channels, one at a time, attach the synthesizer to one of the channels and shorting plugs to the other one or three channel(s).<br>Verify that all shorted channels have no tonal at 38 kHz that exceeds -70 dB. |
| 9. | Load test setup **test022.cfg**. Set the synthesizer for 95 kHz at 1 Vrms and attach it to the Channel 1 input BNC connector. Attach a shorting plug to the Channel 2 input BNC connector.<br>a. Verify that any tonal appearing at 95 kHz on Channel 2 is at a maximum signal level of -60 dB.<br>b. Attach the synthesizer to the Channel 2 input BNC connector and attach the shorting plug to the Channel 1 input BNC connector.<br>c. Verify that any tonal appearing at 95 kHz on Channel 1 is at a maximum signal level of -60 dB. |
| 10. | Load test setup **test024.cfg**. Set the synthesizer for 250 Hz at 0.60 Vrms. Remove all shorting plugs. Attach the synthesizer to the Channel 1 input BNC connector.<br>a. Verify that the display is updating (indicated by slight flicker and/or movement of the displayed trace).<br>b. Set the synthesizer to 0.55 Vrms.<br>c. Verify that the display is no longer updating.<br>d. Set the synthesizer to 1 Vrms.<br>e. Verify that the display is updating.<br>f. In the ACQUISITION menu, change the trigger source to external.<br>g. Verify that the "Data Not Available" message is displayed.<br>h. Connect a parallel connection from the Channel 1 input BNC connector to the EXT TRIG input.<br>i. Verify that the display is updating.<br>j. Remove the signal from the EXT TRIG input.<br>k. Verify that the display is no longer updating. |



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## 5–2.5 Testing Channels 5 through 8

To test channels 5 through 8, perform the following:

| Step | Action |
|---|---|
| 1. | If the SPS390 has no more than 4 channels installed, then skip this section. Otherwise, attach the synthesizer to the Channel 5 and Channel 6 input BNC connectors.<br>If the SPS390 has more than 6 channels installed, also attach the synthesizer to the Channel 7 and Channel 8 input BNC connectors. |
| 2. | Load test setup **test101.cfg**. Set the synthesizer for a 1 volt rms sine wave at 2000 Hz.<br>Verify that the display shows a signal at 2000 Hz and between -0.40 dBV and +0.30 dBV on all displayed channels. |
| 3. | Load test setup **test102.cfg**. Change the synthesizer to 10 kHz.<br>Verify that the display shows a signal at 10 kHz and between -0.40 dBV and +0.30 dBV on all displayed channels. |
| 4. | Load test setup **test105.cfg**. Change the synthesizer to 2 kHz.<br>Verify that the display shows a signal at 2 kHz and between -0.40 dBV and +0.30 dBV on all displayed channels. |
| 5. | Load test setup **test111.cfg**. Change the synthesizer to 1 kHz. Start the averager by selecting the START function key (F2) on the SPS390.<br>Verify that all tonals other than the 1 kHz primary tonal are at a maximum level of -70 dB on all displayed channels. |
| 6. | Load test setup **test112.cfg**. Start the averager by selecting the START function key (F2) on the SPS390.<br>Verify that all tonals other than the 1 kHz primary tonal are at a maximum level of -70 dB on all displayed channels. |
| 7. | Load test setup **test119.cfg**. Set the synthesizer for 1.25 Vrms at 1 kHz.<br>a. Verify that all displayed channels show an overload indication (indicated by the corresponding Channel Overload Indicator going to reverse video).<br>Decrease the input level to 1.0 Vrms.<br>Verify that all channels are no longer in overload. |
| 8. | Load test setup **test121.cfg**. Set the synthesizer for 38 kHz at 1 Vrms. For each of the displayed channels, one at a time, attach the synthesizer to one of the channels and shorting plugs to the other one or three channel(s).<br>Verify that all channels which are shorted have no tonal at 38 kHz which exceeds -70 dB. |
| 9. | Load test setup **test125.cfg**. Set the synthesizer for 1 kHz at 20 mVrms. Attach the synthesizer to the inputs of all displayed channels.<br>a. Verify that all channels show a 1 kHz tonal between 19.0 mVrms and 21.0 mVrms.<br>Use the Channel Full Scale Controls to change the input Full Scale for all displayed channels to each of the following levels: 50 mVrms, 100 |



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## 5–2.6 Testing System Features

To exercise the test system features, perform the following steps as applicable.

| Step | Action |
|---|---|
| 1. | Perform this step only if the SPS390 has the SSG option.<br><br>Load test setup **test203.cfg**. Disconnect the synthesizer and connect the SSG output to Channel 2. In the SETUP menu, select the Noise Gen Option. When the Noise Generator Setup menu is displayed, select OK. Start the noise function by selecting the NOISE START function key (F7) on the SPS390.<br><br>Verify, using the Spectrum display, that the overall value (OA) is between 0.95 Vrms and 1.10 Vrms. |
| 2. | Perform this step only if the SPS390 has the SSG option.<br><br>Load test setup **test203a.cfg**. In the SETUP menu, select the Noise Gen Option. On the Noise Generator Setup menu, select OK. Start the noise function by selecting the NOISE START function key (F7) on the SPS390.<br><br>Verify, using the Spectrum display, that the overall value (OA) is between 0.95 Vrms and 1.10 Vrms. |
| 3. | Perform this step only if the SPS390 has the SSG option.<br><br>Load test setup **test204.cfg**. In the SETUP menu, select the Sine Gen Option. On the Sine Generator Setup window, select OK. In the Spectrum display select the S function. In the AXES SETUP window ensure that the scale for the Y Axis Max is set to 1 and then select OK. Start the sine function by selecting the SINE START function key (F7) on the SPS390.<br><br>a. Verify that there is no overload for Channel 2.<br>b. Verify, using the Spectrum display, that only one tonal appears and that it has a minimum of 900 millivolts across the entire sweep.<br>c. Verify that the time required for the tonal to make one complete sweep is between 39 and 41 seconds. |
| 4. | Perform this step only if the SPS390 has the SSG option.<br><br>Load test setup **test204a.cfg**. In the SETUP menu, select the Sine Gen Option. On the Sine Generator Setup window, select OK. Start the sine function by selecting the SINE START function key (F7) on the SPS390.<br><br>a. Verify that there is no overload for Channel 2.<br>b. Verify, using the Spectrum display, that only one tonal appears and that it has a minimum of 850 millivolts across the entire sweep.<br>c. Verify that the time required for the tonal to make one complete sweep is between 78 and 82 seconds. |
| 5. | Load test setup **test205.cfg**. For all available channels attach one 50 ohm terminator per channel pair and connect Channel 1 to Channel 2, Channel 3 to Channel 4, Channel 5 to Channel 6, and Channel 7 to Channel 8.<br><br>Verify that all displayed DC amplitudes are between 175 millivolts and 225 millivolts. |

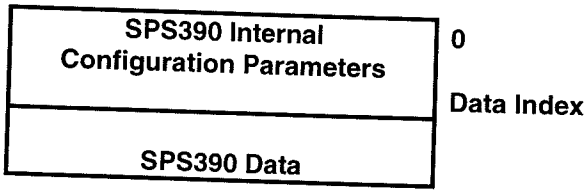


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# 6– EXTERNAL INTERFACE GUIDE

## 6–1 SPS390 INTERNAL FILE FORMAT

Each of the data files of the SPS390 have the following basic format, a section for the configuration parameter that determine the data characteristics and a section containing the raw data.




Since the 3.3B release of the SPS390 software, the location of the block of data has been fixed to a certain value according to the file's type. The size of the configuration header is fixed by padding to the end.

| File Type | File Offset |
|---|---|
| Average (AVG) | 5120 |
| Extended recorded (XRC) | 5120 |
| Trace (TRC) Version 3.4 and Before | 558 |
| Trace (TRC) Version 3.5 and After | 676 |
| Waterfall (WFL) | 5120 |
| | |
| | |

### 6–1.1 SPS390 Interrupt and Address Usage:

The SPS390 uses the following hardware interrupts:

| Description | IRQ |
|---|---|
| Timer | o |
| Keyboard | 1 |
| Chained to 2nd int controller | 2 |
| COM2 (Mouse/Trackball) | 3 |
| COM1 | 4 |
| Front Panel Keypad (I/O board) | 5 |
| Floppy Disk Controller | 6 |
| LPT1 | 7 |
| Real Time Clock | 8 |
| Redirected from IRQ2 | 9 |
| Channel Overload (DSP board) | 10 |
| DSP32C (DSP board) | 11 |
| DSP32C (I/O board) | 12 |
| Math Coprocessor | 13 |
| Hard Disk | 14 |
| Unassigned | 15 |



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## 6–1.1 SPS390 Data Section

### 6–1.1.1 Average File Type: (*.avg)

The average data section is stored in internal ATT DSP32C floating point format in Vrms2 units ( see Decoding the Average (AVG) ).

The data is organized into sequence blocks of data each active channel. Note, only the active channels set in the Analysis Dialog are saved.

For example, an average file may contain four blocks of data for each active channel, 2, 5, 6, 7.

***Block of Channel 2 Data***
***Block of Channel 5 Data***
***Block of Channel 6 Data***
***Block of Channel 7 Data***

The format of each block is determined by the operating mode of the SPS390:

The **Sync Spectrum** mode stores the time block component of the acquired data. The size of this block is determined by the time block size set in the Analysis Dialog.

The **Spectrum, Sync Spectrum** and **Octave** mode only store the AVG spectrum (FFT) component of the acquired data. The size of this block is determined by the lines of resolution set in the Analysis Dialog.

The **Cross Spectrum** mode stores the AVG spectrum, and the CO and QUAD sections. The size of each sections is determined by the lines of resolution set in the ANALYSIS dialog. These sections are ordered into following manner:

AVG spectrum (FFT)
CO section for 1st reference
QUAD section for 1st reference
CO section for 2nd reference ( if MIMO is activated )
QUAD section for 2nd reference ( if MIMO is activated )



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### 6–1.1.3 Trace File Type: (*.trc)

The trace data is stored according to the trace type. The MODE of the SPS390 determined which display types that are available for the user to display and store to disk. The binary data stored in the trace files is in Vrms units.

Format for the SPS390 trace file with the following display types:

If the display type is any magnitude displays such as **Magnitude Spectrum** or **Transfer Function Magnitude**, the data consists of the Cross Spectrum magnitude information for the number of bins of resolutions specified in the ANALYSIS dialog.

If the display type is any phase displays such as **Phase** or **Transfer Function Phase**, the data consists of the Cross Spectrum phase information for the number of bins of resolutions specified in the ANALYSIS dialog.

If the display type is **Orbit** or **Nyquist**, the data consists of the 2 sets of values with each set having the number of bins of resolutions specified in the ANALYSIS dialog. For **Orbit**, the imaginary data set comes before the real data set. For Nyquist, the Y values come before the X values.

If the display type is an **Octave**, the data consists of the octave magnitude information for the number of octaves specified in the ANALYSIS dialog plus a extra floating point value for the Overall Magnitude that directly follows the trace data.

### 6–1.1.4 Waterfall File Type: (*.wfl)

The waterfall data file consists of a contiguous block of data from each active channel starting with the lowest channel and ending with the highest channel such as one to eight.

Each channel block is a sequence of waterfall records arranged according to their order of acquisition. However, if waterfall stop method is set to CONTINUOUS, the acquired waterfall records will wrap-around. Therefore, the first acquired record does not always correspond to Record #1.

The user can change the type of data allowed into the waterfall by going to the **Waterfall** dialog box and selecting a waterfall type. The size of each record is determined by number of lines of resolution set in the **Analysis** dialog plus 5 additional floating point values. These additional values are used to store the RPM, time-tag and overall RMS values as seen in the following table:

| Record | Byte Offset | Data |
|---|---|---|
| 0 to n | 0 to (n*4) | Vrms data |
| n + 1 | (n + 1) * 4 | Overall RMS for record |
| n + 2 | (n + 2) * 4 | Time ( divide floating point value by 100 to get into seconds ) |
| n + 3 | (n + 3) * 4 | RPM ( only present if tach activated ) |
| n + 4 | (n + 4) * 4 | Not Used |
| n + 5 | (n + 5) * 4 | Not Used |



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### DSP32 Floating Point Bit Format

sfffffff fffffff fffffff eeeeeeee

N = (-1)s * 1.f * 2(e-127)

### IEEE Floating Point Bit Format

seeeeeee efffffff fffffff fffffff

N = [(-2)s * 0.f] * 2(e-128)



The DSP ATT32C stores negative numbers in Two's Complement. To interpret negative numbers, first complement the bits and then add 1.

## 6–1.4 SPS390 Floating Point Format

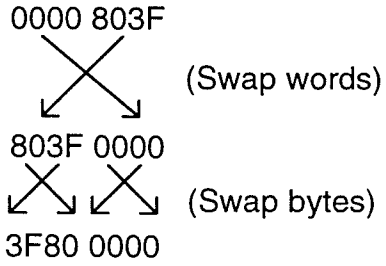
The data stored internally to the SPS390 uses the 32-bit floating point format that conforms to the Institute of Electrical and Electronics Engineers (IEEE) standard 754 format. However, when the SPS390 data is stored into a file, the data is byte and word swapped. This swapping occurs is a result of how the Intel-based X86 hardware stores data in internal memory. This is not a problem if the data is read by a machine having the same Intel architecture. However, if this data is exported to another hardware system (such as a RISC based machine like the 680x0), the data may have to be unswapped to interpret the data correctly.

### A example of how to unswap the floating point data:

This is a section of a hex dump of 32-bit floating point data stored in a MSDOS datafile. The data corresponds to the floating point numbers ( 0.0, 1.0, 2.0, 3.0, and 4.0).

| 0.0 | 1.0 | 2.0 | 3.0 | 4.0 |
|---|---|---|---|---|
| 0000 0000 | 0000 803F | 0000 0040 | 0000 4040 | 0000 8040 |

The correct IEEE representation of 1.0 is 3F800000. Therefore, the two 16-bit words need to be swapped and then, the bytes in each word need to be swapped.






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```
            }
                else
                {
                b4 = b4 | 0x80;          // Supply "1" for "1.f" by setting MSD
                iSign = 1;               // Set to positive
            }

            // Shift before adding the three mantissa bytes
                l4 = (long) b4 << 16;    // align number shifting over 16 bytes
            l3 = (long) b3 << 8;         // align number shifting over 8 bytes

                rMantissa = (l4 + l3 + b2) / pow (2, 23);  // 1.f
                rExponent = pow( 2.0f, b1 - 127 );

                // Number = sign * 1.f * 2 (e - 127 )
                rVrmsSqr = iSign * ( rMantissa * rExponent ) / 2;

                // Take square-root to change the Vrms units
                rVrms = (float) sqrt( (double) rVrmsSqr );

                // Output routine goes here!!!
            }
            _close( iFile );
    exit( 0 );
}
```



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## 6–1.7 Configuration Header Description

Each SPS390 has a file header that contains the information about the data stored into that file. The averager (AVG), waterfall (WFL), and extended recorder (XRC) each have a detailed configuration header that stored both the data characteristics and acquisition and analysis settings. The following table contains a list of the useful variables and their corresponding offset into the data file.

| Field Name | Offset | Length | C Data Type | Notes |
|---|---|---|---|---|
| Acquisition Parameters | | | | |
| Channel 1 Information | | | | |
| mv/EU | 114 | 4 | float | |
| Full Scale Voltage | 118 | 4 | float | |
| DB Reference | 122 | 4 | float | |
| Reference Voltage for DB | 126 | 4 | float | |
| EU Label | 134 | 8 | char | |
| Channel 2 Information | | | | |
| mv / EU | 142 | 4 | float | |
| Full Scale Voltage | 146 | 4 | float | |
| DB Reference | 150 | 4 | float | |
| Reference Voltage for DB | 154 | 4 | float | |
| EU Label | 162 | 8 | char | |
| Channel 3 Information | | | | |
| mv / EU | 170 | 4 | float | |
| Full Scale Voltage | 174 | 4 | float | |
| DB Reference | 178 | 4 | float | |
| Reference Voltage for DB | 182 | 4 | float | |
| EU Label | 190 | 8 | char | |
| Channel 4 Information | | | | |
| mv / EU | 198 | 4 | float | |
| Full Scale Voltage | 202 | 4 | float | |
| DB Reference | 206 | 4 | float | |
| Reference Voltage for DB | 210 | 4 | float | |
| EU Label | 218 | 8 | char | |
| Channel 5 Information | | | | |
| mv / EU | 226 | 4 | float | |
| Full Scale Voltage | 230 | 4 | float | |
| DB Reference | 234 | 4 | float | |
| Reference Voltage for DB | 238 | 4 | float | |
| EU Label | 246 | 8 | char | |
| Channel 6 Information | | | | |
| mv / EU | 254 | 4 | float | |
| Full Scale Voltage | 258 | 4 | float | |
| DB Reference | 262 | 4 | float | |
| Reference Voltage for DB | 266 | 4 | float | |



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| Field Name | Offset | Length | C Data Type | Notes |
|---|---|---|---|---|
| Acquisition Parameters | | | | |
| Number of Analog Boards Present | 484 | 2 | integer | 0 to 8 |
| Analyzer Mode | 514 | 2 | WORD (unsigned integer) | 2 = Spectrum<br>4 = Sync Spectrum<br>8 = Cross Properties<br>16 = Octave<br>64 = Correlation |
| Number of Active Channels | 516 | 2 | integer | 1 to 8 |
| Active Channel List | 518 | 2 | integer | bit 1 = Channel 1<br>bit 2 = Channel 2<br>bit 3 = Channel 3<br>bit 4 = Channel 4<br>bit 5 = Channel 5<br>bit 6 = Channel 6<br>bit 7 = Channel 7<br>bit 8 = Channel 8 |
| Reference Channel List | 520 | 2 | integer | Same as Active List |
| Extended Recorder Size per Channel | 526 | 4 | long | Depends on Installed DSP Memory |
| Size of Time Blocks | 530 | 2 | integer | 512, 1024, 2048, 4096 |
| Number of Lines Resolution for FFT | 532 | 2 | integer | 200, 400, 800, 1600 |
| FFT Weighting Window | 534 | 2 | integer | 0 = Rectangle<br>1 = Hanning<br>2 = Harris Flat Top<br>4 = Response |
| Overlap | 536 | 2 | integer | 0 = None<br>1 = 25%<br>2 = 50%<br>3 = 75%<br>4 = 90%<br>5 = MAX |
| Response Tau | 538 | 2 | integer | 1 to 15 |
| Response Offset | 540 | 2 | integer | 10% to 80% |
| Octave Mode | 582 | 2 | WORD (unsigned integer) | 1 = 1/1 Octave<br>3 = 1/3 Octave |
| Octave Weighting | 584 | 2 | WORD (unsigned integer) | 1 = A<br>2 = B<br>3 = C<br>4 = D<br>5 = Flat |
| Octave Number of Bins of Resolution | 586 | 2 | WORD (unsigned integer) | |



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| Field Name | Offset | Length | C Data Type | Notes |
|---|---|---|---|---|
| Acquisition Parameters | | | | |
| Waterfall Time Increment | 626 | 4 | float | Delta Time |
| Waterfall % Amplitude | 630 | 2 | integer | % Amplitude Change |
| Averager Mode | 634 | 2 | integer | 0 = Linear<br>1 = Exponential<br>2 = Peak |
| Averager Stop Mode | 636 | 2 | integer | 0 = Count<br>1 = Time<br>2 = Manual |
| Averager Target Count | 638 | 2 | integer | |
| Averager Current Count | 640 | 2 | integer | |
| Averager Target Time | 642 | 4 | long | in Seconds |

## 6–1.8 SMS Data File Format

The SPS390 stores the SMS data into four different data files, Frequency Response Function (FRF), Coherence (COH), Auto Power Spectrum (APS), and Cross Power Spectrum (CPS).

The following table describes the SMS elements stores into the data file by SPS390:

| Description | Variable | Format | Value |
|---|---|---|---|
| General Header | header | headert | See *headert* structure |
| STAR Revision Code | revcode | int | 2831 |
| Header Length | headerlen | int | 16 |
| Measurement Header | msmheader | msmheadert | See *msmheadert* structure |
| Primary Measurement Type | datatype | int | 0 = FRF 12 = APS<br>2 = COH 13 = CPS |
| # of Complex Data Pairs | noelements | int | Analysis Resolution of 200, 400, 800, or 1600 lines |
| Measurement ID Label | measid | char[128] | Date and Time Label dd:yy:hh:mm:ss |
| Analyzer ID Label | analyzerid | char[24] | "SD390" |
| APS Amplitude Compensation | ispeak | char | 3 = Hanning compensation of Channel A<br>5 = Hanning compensation of Channel B(only used for datatype APS) |
| X Axis Range Information | xaxisrange | xaxisranget | See *xaxisranget* structure |
| Starting Frequency | xlow | float | Any valid frequency |
| Delta Frequency | deltax | float | Any valid frequency |



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### 6–1.8.1 How to calculate the "point" variable

point = (10 * abs(point_number) + direction) * sign(point_number)

Examples: 33Z = (10 * 33 + 3) = 36, -28R = -(10 * 28 + 4) = 284

For COH, FRF and CPS Channel 1 is the Reference Channel and Channel 2 is the Response Channel.

For APS Channel 1 and Channel2 are the same point.

For COH, FRF and CPS the complex data includes the real component followed by the imaginary component for each data point.

For APS, the imaginary component is 0, and the real component is the amplitude.

SMS Header C Data Structures

The SMS data files are defined by the following definitions and structures. Only the commented variables are used by the SPS390. All others variables are set to a NULL value.

```
#define MAXBSIZE 1601

typedef struct
{
        int   revcode;      /* STAR revision code */
        int   headerlen;    /* header length */
        int   entries;
        int   pline;
        int   sentries;
        int   spline;
        int   tentries;
        int   tpline;
} headert;

typedef struct
{
        int   datatype;     /* primary measurement type */
        int   misctype;
        int   noelements;           /* number or complex data pairs */
        float msmcalval;
        char caltrace[16];
        char measid[128];           /* measurement id label */
        char labletype[120];
        char xlabel[16];
        char ylabel[16];
        char msmdate[16];
        char msmtime[16];
        char analyzerid[24];        /* analyzer id label */
        int   noave;
        int   windowtype;
        char miscwindow[16];
        float wdwvalue;
        float spacing;
        int   surface;
        float minreal;
        float maxreal;
        float minimag;
        float maximag;
        float maxmag;
        int   mmdefined;
        char micpair;
        char ispeak;    /* APS amplitude compensation */
        float excamp;
```



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The following tables contain the values for various internal SPS390 configuration items:

| Display Types | Value |
|---|---|
| Compressed Time | 0 |
| Time Trace | 1 |
| Orbit | 2 |
| Magnitude Spectrum | 5 |
| Phase Spectrum | 6 |
| Real Spectrum | 7 |
| Imaginary Spectrum | 8 |
| TF Mag Accelerance | 11 |
| TF Mag Mobility | 12 |
| TF Mag Compliance | 13 |
| TF Mag Stiffness | 14 |
| TF Mag Impedance | 15 |
| TF Mag Dynmass | 16 |
| TF Phase | 17 |
| TF Phase Accelerance | 18 |
| TF Phase Mobilty | 19 |
| TF Phase Compliance | 20 |
| TF Phase Stiffness | 21 |
| TF Phase Impedance | 22 |
| TF Phase Dynmass | 23 |
| TF Coherence | 24 |
| TF Real | 25 |
| TF Imaginary | 26 |
| TF Nyquist | 27 |
| Cross Spectrum Magnitude | 28 |
| Cross Spectrum Phase | 29 |
| Cross Spectrum Real | 30 |
| Cross Spectrum Imaginary | 31 |
| Cross Spectrum Nyquist | 32 |
| Sync Spec Magnitude | 35 |
| Sync Spec Phase | 36 |
| Sync Spec Real | 37 |
| Sync Spec Imaginary | 38 |
| Octave | 39 |
| Auto Correlation | 40 |
| Cross Correlation | 41 |
| Multiple Display | 42 |

| Analyzer Mode | Value |
|---|---|
| Spectrum | 0x02 |
| Syn Spectrum | 0x04 |
| Cross Properties | 0x08 |
| Octave | 0x10 |
| Correlation | 0x40 |

| Waterfall Type | Value |
|---|---|
| Single | 0x01 |
| Cascade | 0x02 |
| Peak Hold | 0x04 |
| Profile | 0x08 |
| Profile OA | 0x10 |

| Math Functions Types | Value |
|---|---|
| None | 0 |
| Add Constant | 1 |
| Subtract Constant | 2 |
| Multiply Constant | 3 |
| Divide Constant | 4 |
| Inverse | 5 |
| Square | 6 |
| Square Root | 7 |
| Binary Add | 20 |
| Binary Subtract | 21 |
| Binary Multiply | 22 |
| Binary Divide | 23 |



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## 6–2.3 COMMAND topic

The **COMMAND** topic provides the ability to control the SPS390 analyzer from another applications. It provides the capability of simulating the SPS390 button presses and performs various operating tasks such as loading in a configuration file, starting the averager, and saving a trace to a file.

In Visual Basic, the **COMMAND** topic is executed via the ***LinkExecute*** command. To initiate a DDE link, set the LinkTopic with SD390 server and topic and call the LinkExecute with the item strings documented below.

In C, this topic is performed via the ***DdeClientTransaction*** using the ***XTYP_EXECUTE*** transaction type.

Topic Syntax: **COMMAND**

These are the group of DDE items that are used to control the SPS390 operations.

| | |
|---|---|
| Item Syntax: | **CTRL ACQU [ UPDATE \| HOLD ]**<br>**CTRL AVGR [ START \| STOP ]**<br>**CTRL WATR [ START \| STOP \| RESUME \| SINGLE ]**<br>**CTRL SSG [ START \| STOP ]**<br>**CTRL SRA**<br>**CTRL ARNG** |
| Description: | Set a specified SPS390 control or process to a desired setting. |
| Example: | Inform SPS390 to start to acquire data into the input buffer.<br>Server = SD390, Topic = **COMMAND**,<br>Item = "**CTRL ACQU UPDATE**" |

| | |
|---|---|
| Item Syntax: | **SAVE** <type> <filename> |
| Arguments: | type = { trace \| **CNFG \|AVGR \| WATR \| EXTR** }<br>filename = a string that contains a valid DOS filename with extension. |
| Description: | Save a specified area to SPS390 disk storage under a filename. |
| Example: | Server = **SD390**, Topic = **COMMAND**,<br>Item = "**SAVE CNFG testfile.cfg**" |

| | |
|---|---|
| Syntax: | **LOAD** <type> <filename> |
| Arguments: | type = { trace \| **CNFG \| AVGR \| WATR \| EXTR** }<br>filename = a string that contains a valid DOS filename with extension. |
| Description: | Load specified area from SPS390 disk storage under a filename. |
| Example: | Load configuration from a file.<br>Server = **SD390**, Topic = **COMMAND**<br>Item = "**LOAD CNFG testfile.cfg**" |



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Example: Retrieve the current status information.
Server = **SD390**, Topic = **STATUS**, Item = **INFO**

## 6–2.5 CHANNEL topic

This is a specialized group of DDE topics that provide machine diagnostic data information about analyzed spectrum data.

Topic Syntax: **CHANNEL_1 | CHANNEL_2 | CHANNEL_3 | CHANNEL_4**
**CHANNEL_5 | CHANNEL_6 | CHANNEL_7 | CHANNEL_8**
Item Syntax: **1XMAG**
Description: Get fundamental (1X) vibration in unscaled Vrms.



Use must have a complex waterfall and averager running.

**Example:** Get the 1X vibration from channel two.
Server = **SD390**, Topic = **CHANNEL**_2, Item = **1XMAG**

---

**Item Syntax:** **1XPHASE**
**Description:** Get fundamental (1X) phase in degrees .



Use must have a complex waterfall and averager running.

**Example:** Get the 1X phase from channel two.
Server = **SD390,** Topic = **CHANNEL_2,** Item = **1XPHASE**

---

Item Syntax: **MISALIGN**
Description: Get the misalignment (2X) vibration of a selected channel.



Use must have a complex waterfall and averager running.

Example: Get the misalignment vibration of channel two.
Server = **SD390**, Topic = **CHANNEL_2**, Item = **MISALIGN**



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This glossary contains definitions of common terms used in the collection and analysis of digital data relating to the dynamic properties of mechanical systems. Such data may include strain, vibration, sound pressure variations, system electrical characteristics, and other measurements well-suited to the SPS390. The glossary is based upon a list of terms and definitions compiled by the Institute of Environmental Sciences.

ACCELERANCE. *See* INERTANCE, MECHANICAL IMPEDANCE.

ACCELERATION SRS. *See* SHOCK RESPONSE SPECTRUM.

ACCURACY. The degree of conformity of a measure to a model, standard, or true value. Usually expressed as a percentage of a specified value. Should consider other dependent factors such as temperature, frequency, relative value (full scale vs. absolute), drift rates, range, etc.

ALIASING. When the sample rate $f$ is less than two times the highest frequency in the data, the frequency is ambiguously represented. The frequencies above $f/2$ will be folded back into the lower frequencies to produce erroneous results.

AMPLITUDE DENSITY FUNCTION. *See* PROBABILITY DENSITY FUNCTION.

AMPLITUDE DISTRIBUTION FUNCTION. *See* PROBABILITY DISTRIBUTION FUNCTION.

ANALYSIS RANGE/BANDWIDTH. *See* FREQUENCY RANGE.

ANTI-ALIASING FILTER. *See* FILTER, ANTI-ALIASING.

APPARENT WEIGHT/APPARENT MASS. A term sometimes used for the preferred term EFFECTIVE MASS (*See* MECHANICAL IMPEDANCE).

APERTURE ERROR. *See* ERROR, APERTURE.

ARGAND DIAGRAM. *See* NYQUIST DIAGRAM.

ARRAY. An ordered set of numbers; in a sampled data system discussed in this glossary (*See* SAMPLING) they are equally spaced in time or frequency.

AUTOCORRELATION FUNCTION. A measure of the similarity that a function has with a time-displaced version of itself, as a function of the time displacement or lag. When the lag is zero, the value of the autocorrelation is equal to the mean square value of the function. The sample autocorrelation function $\hat{R}_x(j)$ of the sequence $x_i$ is usually computed

$$\hat{R}_x(j) = \frac{1}{N-j}\sum_{i=1}^{N-j} X_i X_{i+j} \qquad j = 0, 1, \ldots, m$$

where

$x_i$ = the function x(t) sampled at the time $i\Delta t$
$\Delta t$ = the sample interval
$j\Delta t$ = the lag
N = the number of samples of x
$m\Delta t$ = the maximum lag

For a discrete "stationary" random quantity where N must be finite, the formula gives an estimate only with a statistical uncertainty which increases as N decreases. It is used to identify periodic components in the sample function $x_i$.



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BIAS. A tendency of a statistical estimate to deviate in one direction away from a true value.

BLOCK. *See* FRAME, MEMORY LENGTH.

BLOCK SYNCHRONOUS. A data sampling and/or processing technique in which the beginning or ending of a data block is synchronized with an external event.

BLOCKED CONSTRAINTS. *See* MECHANICAL IMPEDANCE /MOBILITY.

BOXCAR FUNCTION. *See* WINDOW.

BUTTERWORTH FILTER. *See* FILTER, BUTTERWORTH.

CALIBRATION FACTOR (also called SCALE FACTOR). Data acquired by an ADC is in terms of binary counts representing some fraction of a specified full-scale voltage input. The calibration factor is the multiplier that must be applied to the digitized (*See* DIGITIZE) value to convert it to engineering units. For example, an acceleration signal of 5-g/v is to be digitized by a 12-bit (11 bits + sign) ADC having a full input of 10 V. The Calibration Factor is

$$CF = 5\frac{g}{volt} \times \frac{10V}{2^{11}counts} = 24.414 \times 10^{-3} g / count$$

An ADC count of 1562 would then be equivalent to

g = (1562) x (24, 414 x 10 3) = 38. 135g

CENTER FREQUENCY. A characteristic of a bandpass or band reject filter or a constant percentage filter of these types, the center frequency is the geometric mean of the upper ($f_U$ ) and the lower ($f_L$) cutoff frequencies:

$$f_c = \sqrt{f_L f_U}$$

For a constant bandwidth filter, the center frequency is the arithmetic mean (*See* AVERAGE, ARITHMETIC) of the upper and lower frequencies:

$$f_c = \frac{1}{2(f_L + f_U)}$$

CHEBYSHEV FILTER. *See* FILTER, CHEBYSHEV.

CHI SQUARE. *See* GOODNESS OF FIT.

CIRCULAR CONVOLUTION (PERIODIC CONVOLUTION). The periodic convolution $z_i$ of two time sequences $x_i$ and $y_j$ (i = 0, 1, 2, . . . N-1) is defined as

$$z_i = \sum_{m=0}^{N-1} \overline{x}_m y_{i-m}$$

where $\overline{x}_m$ is a periodic extension of the sample set $x_m$, i, e., $\overline{x}_{i+kn} = x_i$ (n = 0, 1, 2., . . N-1) (k = 0, ± 1, ±2. . . ). The inverse discrete Fourier transform ($DFT^{-1}$) of the product of two DFTs is a periodic convolution of the original time sequences (*See* DFT and WRAP-AROUND ERROR) .

CIRCULAR CORRELATION. A correlation function computed in terms of a circular convolution.



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COMPLEX FUNCTION. A complex function is any mathematically defined relationship of the form

$y(x) = a(x) + ib(x)$ ,

where

x = real variable,
a(x) = the real part of y(x), a = Re{y(x)},
b(x) = the imaginary part of y(x), b = Im {y(x) }, and
$i = \sqrt{-1}$

Complex functions are often represented in terms of their amplitude

$|y(x)| = [a2(x) + b2(x)]1/2$

and phase

$\varphi y(x) = \tan\text{-}1\ [b(x)/a(x)]$

CONFIDENCE BANDS. *See* CONFIDENCE INTERVAL.

CONFIDENCE INTERVAL. The range with a specified value of uncertainty within which the true value of a measured quantity will lie. For example, consider a random variable X with a true mean of x. Note that the true mean is never known, but can only be estimated. An estimate of the mean, $x_{bar}$, is then made with a 90% confidence interval of +a. The confidence statement means that if we made 100 such independent estimates of x, about 90 would be in the range of x±a, and about 10 would lie outside this range. Note that the confidence statement does not state how close a particular estimate x is to the true value x (a is a function of the numbered samples and the variance of the samples).

CONSTRAINTS, BLOCKED and FREE. (*See* MECHANICAL IMPEDANCE/MOBILITY.

CONTINUOUS SPECTRUM. The type of spectrum produced from nonperiodic data. The spectrum is continuous in the frequency domain (*See* LINE SPECTRUM, FREQUENCY DOMAIN).

CONVOLUTION. A mathematical concept. The convolution of continuous functions x(t) and y(t) is defined as

$$x(t) * y(t) = \int_{-\infty}^{\infty} x(\tau)\, y(t-\tau)\, d\tau$$

Let W(f),X(f), Y(f), and Z(f) be the Fourier transforms of w(t), x(t), y(t), and z(t), respectively. It can be shown that if z(t) = x(t)*y(t), then

Z (t) = X(f) · Y(f)

and if W(f) = X(f)*Y(f), then

x(t) · (t) = w(t).

If x(t) is the input to a linear system whose impulse response is h(t), then the output of the system is the convolution x(t)* h(t). (See CIRCULAR CONVOLUTION).

CORRELATION FUNCTION. *See* AUTOCORRELATION FUNCTION; CROSS CORRELATION FUNCTION.



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CROSS SPECTRAL DENSITY FUNCTION. A measure in the frequency domain of the similarity of two functions. It is usually computed from the Fourier transforms of two discrete functions $x_i$ and $y_i$ to

$$\tilde{G}_{xy}(k) = \frac{2}{} \tilde{G}_{xy} = \frac{2\Delta t}{N}[X_k^* Y_k]$$

where

| | |
|---|---|
| $\Delta t$ | = sample interval, |
| N | = the frame size of xi and Yi, |
| Xk* | = complex conjugate of Xk, and |
| Xk, Yk | = the discrete Fourier transform of xi, y |

$\tilde{G}_{xy}$ is a "raw" cross spectral estimate, The cross spectral density is then estimated by averaging M frames of $\tilde{G}_{xy}$. It can also be computed from the "cross-correlation function" as

$$G_{xy}(k) = 2F[R_{xy}(j)]$$

where

F[ ] = Fourier transform of quantity within brackets
Rxy(j) = cross-correlation of xi and Yi

For discrete "stationary" random quantities, the formulas give an estimate only with a statistical uncertainty (error) which increases as M decreases. Typically, the data is also multiplied by a window and this will affect the effective resolution of the estimate. (See FOURIER TRANSFORM WINDOW).

CUMULATIVE PROBABILITY DISTRIBUTION FUNCTION. *See* PROBABILITY DISTRIBUTION F UNCTION.

CURVE FITTING. The process whereby coefficients of an arbitrary function (usually a polynomial) are computed such that the function approximates the values in a given data set. A mathematical function, such as the minimum mean squared error, is used to judge the goodness of fit (*See* MEAN SQUARED ERROR, TEST).

CUTOFF FREQUENCY. The frequency at which the rolloff skirt of the filter shape is down from the nominal unity gain passband level by a specified amount.

DAMPING, Dissipation of energy.

DAMPING FACTOR. *See* DAMPING RATIO.



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DFT, Discrete Fourier Transform (*See* FOURIER TRANSFORM).

$DFT^{-1}$, Discrete Inverse Fourier Transform (*See* FOURIER TRANSFORM).

DIFFERENCE SPECTRUM, (*See* ERROR SPECTRUM).

DISPLACEMENT SRS, *See* SHOCK RESPONSE SPECTRUM.

DISTRIBUTION FUNCTION, *See* PROBABILITY DISTRIBUTION FUNCTION.

DOMAIN OFF, *See* FUNCTION.

DOUBLE LENGTH TRANSFORM. A 2N point DFT of sample set $x_i$ (i = 0, 1, ... N-1) extended with N zeros. This transform does not add any new information in the frequency domain, but is sometimes required to prevent circular convolution errors. Sometimes refers to a technique for obtaining the Fourier transform of a time series of length 2N by utilizing an N point fast Fourier transform algorithm (*See* FREQUENCY DOMAIN; CIRCULAR CONVOLUTION; FOURIER TRANSFORM).

DYNAMIC MASS. *See* MECHANICAL IMPEDANCE.

DYNAMIC RANGE. For spectrum measurements, the difference, in dB, between the overload level and the minimum detectable signal level (above the noise) within a measurement system. The minimum detectable signal of a system is ordinarily fixed by one or more of the following: noise level, low level distortion, interference, or resolution level. For transfer function measurements, the excitation, weighting, and analysis approaches taken can have a significant effect on resulting dynamic-range (*See* PROCESSING GAIN; TRANSFER FUNCTION).

EFFECTIVE MASS. A term sometimes used for the preferred term "dynamic mass" (*See* MECHANICAL IMPEDANCE; DYNAMIC MASS).

EFFECTIVE (NOISE) FILTER BANDWIDTH (Also called EQUIVALENT NOISE BANDWIDTH (ENBW)). The width of an ideal rectangular filter of unity gain having an area equal to the area under the square of the actual filter transmissibility characteristic.

EIGENVALUE/EIGENVECTOR. In structural dynamics the eigenvalues can be interpreted as the modal frequencies and eigenvectors as the "mode shapes." The basic concepts involved in the definition of eigenvalues and eigenvectors is as follows: Let A be an nxn matrix. If a non-zero vector X and a non-zero scalar $\lambda$ exist, such that

$$AX = \lambda X$$

then $\lambda$ is said to be an eigenvalue of A, and X an eigenvector. To obtain the eigenvalues, $\lambda i$, find the roots of the equation

$$\det(A-\lambda I) = O,$$

and to find the corresponding eigenvectors Xi solve

$$(A-\lambda i)Xi = O.$$

ENERGY SPECTRAL DENSITY. The square of the magnitude of the Fourier spectrum of a transient waveform. That is, if $F(\omega)$ is the Fourier transform of f(t), the energy spectral density, $A(\omega)$, is given by

$$A(\omega) = |F(\omega)|2.$$

Note that the energy spectral density is not necessarily related to physical energy (See FOURIER TRANSFORM).

ENSEMBLE AVERAGING. *See* AVERAGE, ENSEMBLE.



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FAST FOURIER TRANSFORM (FFT). One of a collection of algorithms for computing discrete Fourier transforms (DFTs) in an optimum fashion. The fast Fourier transform (FFT) algorithm reduces the number of complex multiplications required to compute the transform. This is accomplished by factoring the weighted trigonometric summations required in the computation of the discrete Fourier transform into a sequence of shorter weighted summations by taking advantage of the periodicities and symmetries of the periodic weighting factors and of the transform itself.

FFT. *See* FAST FOURIER TRANSFORM.

FILTER, ANTI-ALIASING. A filter used to remove the signals whose frequencies exceed one half the sample rate $f_s/2$. Ideal filters are not realizable so that the cutoff frequency, transition band, and out-of-band rejection need to be considered.

FILTER, BAND-PASS. A filter which passes all signal information between two cutoff frequencies and sharply attenuates all signal information outside that range.

FILTER, BAND-REJECT. A filter which attenuates all information between two cutoff frequencies and passes all signal information outside that range.

FILTER, BESSEL. An efficient stable filter characterized by excellent phase linearity at the expense of sharpness of cutoff and passband flatness. Commonly used for shock and other transient waveform phenomenon and often referred to as a linear phase filter.

FILTER, BUTTERWORTH. A filter having flat passband and moderately sharp cutoffs, but also moderately nonlinear phase response. Commonly used for acquisition of data from random processes.

FILTER, CHEBYSHEV. Similar to the Butterworth filter, it achieves faster rolloff near cutoff, but at the expense of significant passband ripple and a nonlinear phase response.

FILTER, DIGITAL. A weighted sum that can be defined as

$$g_m = \sum_{n=-N}^{N} b_n f_{m-n} - \sum_{n=1}^{N} a_n g_{m-n}$$

where

gm are the filter output samples,
fm are the filter input samples, and
bn and an are the filter weights or coefficients.

If all the weights a are zero, the filter is called a nonrecursive filter. If one or more of the weights $a_n$ are nonzero, the filter is called a *recursive* filter. If any of the weights $b_n$ are nonzero for $n < 0$, the filter is said to be *nonrealizable* since future values of the input samples, $f_m$ , will be required. If all the filter weights $b_n$ are zero for $n < 0$, the filter is said to be *realizable*. If we define a *digital unit impulse* as

$d = 0 \qquad n \neq 0,$

$dn = 1/\Delta t .$

where $\Delta t$ is the sampling interval, then the response of a digital filter to this impulse is called the *impulse response* of the filter. The impulse response for any nonrecursive filter will be finite for any finite N. Hence, nonrecursive filters are sometimes called *finite impulse response* filters. Recursive filters are called *infinite impulse response* filters because the impulse responses are not necessarily finite.

FILTER, GAUSSIAN. A low-pass filter with maximally linear phase response at the expense of sharpness of cutoff and passband flatness. Commonly used for shock and other transient waveform phenomenon (*See* FILTER LOW-PASS).



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FOURIER TRANSFORM. A bilateral transformation typically used to convert quantities from time domain to frequency domain and vice versa, usually derived from the Fourier integral of a periodic function when the period grows without limit, often expressed as a Fourier transform pair. In the classic sense, a Fourier transform takes the form of

$$F(\omega) = \int_{-\infty}^{\infty} f(t) e^{-j\omega t} dt$$

where

f(t) = continuous time waveform
(ω) = Fourier transform
ω = analysis frequency expressed in radians/sec

In the discrete or sampled sense (See DFT), this can be expressed as

$$F_k = \sum_{n=0}^{N-1} f_n e^{\frac{-j2\pi}{N} kn}$$

where

fn = samples of time waveform
n = running sample index
N = total number of samples or "frame size"
k = finite analysis frequency, corresponding to "FFT bin centers"
Fk = discrete Fourier transform, and can be associated with the frequencies
$\omega k = 2\pi k/(N\Delta t)$.

(*See* INVERSE FOURIER TRANSFORM, TIME DOMAIN, FREQUENCY DOMAIN, DFT, DFT $^{-1}$, FAST FOURIER TRANSFORM. )

FRAME. Discrete set of elements (numbers) representing a time or frequency domain function. The numerical size of the element set is called the frame, block, or record size and is generally a power of 2, such as 64, 128, 256, etc., The term frame length, or block length, is used to describe the length of the element set in seconds or milliseconds and is equal to $N\Delta t$ where N is the frame size and $\Delta t$ is the sampling interval (*See* TIME DOMAIN, FREQUENCY DOMAIN, MEMORY LENGTH).

FREE CONSTRAINTS. (*See* MECHANICAL IMPEDANCE/MOBILITY.)

FREE RUNNING. Term used to describe the operation of an analyzer or processor which operates continuously at a fixed rate, not in synchronism with some external reference event. Analyzers, processors and computing systems are often thought to be operating in a triggered, block synchronous, or free running mode of operation.

FREQUENCY CELL. *See* FREQUENCY INCREMENT.

FREQUENCY DOMAIN. Any parameter which is expressed as a function of frequency.

FREQUENCY FUNCTION. *See* PROBABILITY DENSITY FUNCTION.



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GAUSSIAN DISTRIBUTION. The samples of a random variable x are said to belong to a Gaussian distribution, provided its probability density function is of the form

$$p(x) = \frac{1}{\sqrt{2\pi}\,\sigma} e^{-\frac{1}{2}\left(\frac{x-\mu}{\sigma}\right)^2}$$

where

σ = the standard deviation
μ = the mean

As a result of the central limit theorem, the distribution of a random variable which is the composite of a large number of independent random processes will approach a Gaussian distribution independent of the distribution of the individual random processes.

GAUSSIAN FILTER. *See* FILTER, GAUSSIAN.

GIBBS PHENOMENON. In representing a discontinuous signal, such as a square wave, as a finite Fourier series, the discontinuities exhibit a characteristic 9% overshoot. Phenomenon often appears in trying to synthesize a "perfect" frequency domain filter using FFT techniques, and is minimized by applying various windows to the data (*See* FOURIER TRANSFORM, FREQUENCY DOMAIN, FFT, WINDOW).

GOODNESS OF FIT TEST (CHI-SQUARED). A hypothesis test often used to test the equivalence of a "histogram" of experimental data and the "histogram" of some theoretical "probability density function". The number of observations in each subinterval of the "histogram" for the experimental data ($r_i$) is compared with the expected value from the theoretical "histogram" ($P_i$). The result is a probability statement concerning the likelihood that the true distribution of the experimental processes is the same as the theoretical distribution. A measure of the total discrepancy between the experimental and theoretical observations is formed.

$$X^2 = \sum_i (r_i - P_i)^2 / P_i$$

The theoretical distribution which minimizes X2 is assumed to be the most likely distribution of the experimental processes.

HARMONIC. A frequency component which is an integer multiple of the fundamental component of a complex spectrum. Harmonic components often represent unwanted distortion and are classified by their level and harmonic number (multiple of the fundamental), i, e, first, second, third harmonic.

HISTOGRAM. A plot showing the number of occurrences in each interval, used as a discrete approximation of a probability density function. An interval in the range of a variable is divided into subintervals; all the data are examined and the number of occurrences in each interval is then tabulated (*See* PROBABILITY DENSITY FUNCTION).

IMAGINARY. *See* QUAD.

IMPULSE FUNCTION. A mathematical definition of a time pulse with infinitesimal width δ, infinite magnitude 1/δ, and unity area. The major attribute of this function is its broad spectrum width which is, theoretically, infinite, making it attractive as a system excitation function to measure frequency response functions. In practice, this function can be approximated, but never fully achieved, due to the finite bandwidth and gain of instrumentation which is used to create and process the impulse.



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LEAST SQUARES PREDICTION. A least squares prediction is y(c, t) where c has been chosen to minimize the total square error E where

$$E^2(c) = \int_{-\infty}^{\infty} [x(t) - y(c,t)]^2 dt,$$

where x(t) is some desired function to be approximated by y(c, t), and where c is an adjusted parameter.

LINEAR PHASE FILTER. A filter where the phase shift is a linear function of frequency.

LINE SPECTRUM. The discrete frequency spectrum produced by the analysis of a periodic time function. Typically presented with fixed bandwidth resolution and normally contains neither broadband noise nor transient characteristics. Not necessarily given as a line or bar display.

LINEAR SYSTEM. A system is linear if for every element in the system the response is proportional to the excitation. This definition implies that the dynamic properties of each element in the system can be represented by a set of linear differential equations with constant coefficients and that, for the system as a whole, superposition holds.

MAXIMUM (MAXIMAX) SRS. *See* SHOCK RESPONSE SPECTRUM.

MEAN SQUARE ERROR. *See* ERROR, MEAN SQUARE.

MEAN VALUE. The average of all values.

MECHANICAL IMPEDANCE/MOBILITY. The dynamic characteristics of structures can be determined from frequency response functions in any general form of the complex ratio of a motion measurement (acceleration, velocity, or displacement) and a force measurement. For practical reasons most mechanical impedance/mobility measurements are limited to driving point and response measurements of rectilinear motion. The terminology recommended, as well as the resulting measurements, depend heavily on the boundary conditions and constraints of the system being considered. It is generally assumed that if the energy sources are present, they should be removed and replaced with the source impedance. The boundary conditions imposed on the exterior of the system as a result of its natural placement and usage should be clearly stated. The system constraints are those conditions that are a result of the choice of a matrix modeling and/or external subsystem connection points. The constraint conditions of all points of the system other than the driving point are implied by the definition of the type of measurement being made. If the constraint conditions of all other points in the system cannot be described in either of the following two ways they should be explicitly stated. A point in a system is defined as one of the set of measurement points in a distributed system, or at the nodes of a lumped system.

*Blocked Constraints* — The driving point is driven with a motion source. The response point is constrained with an infinite impedance and the restraining force is measured. This implies that the response point motion is zero except for the case where the response point and the driving point are the same. All other points on the system not constrained by explicitly-stated boundary conditions are constrained with an infinite impedance, i. e., their motion is zero. This set of constraints is usually very difficult to implement in a test laboratory, but is commonly used in modeling. This measurement is the measurement of blocked impedance.

*Free Constraints* — The driving point is driven with a force source, and the resulting motion is measured. This implies that the response point motion is measured when the external dynamic force at the response point is zero except for the case where the response point and the driving point are the same. All other points on the system not constrained by explicitly-stated boundary conditions are constrained with the zero impedance, i. e., the applied dynamic force is zero. This is easy to do; it is the measurement of a free mobility.

The recommended terminology for each of these measurements is given below, but because the listed terminology has not been used consistently in the past, a reader should exercise caution in interpreting reported data. The blocked impedance matrix [Z] (a matrix giving the relations between the complete set of measurement/drive points) is needed for modeling but is difficult, if not



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MODAL MASS. The generalized mass coefficient in a set of uncoupled differential equations of motion of a mechanical system. When a linear lumped parameter system has been transformed into a set of uncoupled differential equations of the form

$m_i x_i + c_i x_i + k_i x_i = F\, e^{j\omega t}$,

mi = the modal mass,

ci = the modal damping, and

ki = the modal stiffness

where

| | |
|---|---|
| xi | = generalized displacement coordinate of the ith mode |
| F | = generalized force |
| ω | = driving frequency |
| t | = time |
| j | = $\sqrt{-1}$ |

MODAL STIFFNESS. The generalized stiffness coefficient in a set of uncoupled differential equations of motion of a mechanical system (*See* MODAL MASS).

MODE SHAPE. The characteristic shape assumed by an elastic structure when vibrating at one of its natural frequencies. The mode shapes are proportional to the eigenvectors which satisfy the system equations of motion when evaluated at one of the roots of the characteristic equations (*See* EIGENVALUES).

MULTIPLE COHERENCE. *See* COHERENCE, MULTIPLE.

NICHOLS DIAGRAM. A plot showing magnitude and phase contours of a system transfer function where the ordinate is the logarithmic loop gain and the abscissa is the loop phase angle. Used in control systems and feedback system design.

NOISE. Any unwanted signal. Can also be used to denote a random signal source (*See* WHITE NOISE).

NONLINEAR SYSTEM. A system which is not linear (*See* LINEAR SYSTEM).

NONRECURSIVE FILTER. *See* FILTER, DIGITAL.

NONREALIZABLE FILTER. *See* FILTER, DIGITAL.

NONSTATIONARY. *See* STATIONARY.

NORMAL DISTRIBUTION. *See* GAUSSIAN DISTRIBUTION.

NORMALIZED ERROR SPECTRUM. *See* ERROR, NORMALIZED SPECTRUM,

NORMALIZED STANDARD ERROR. *See* ERROR, NORMALIZED STANDARD.

NYQUIST DIAGRAM. A polar display in which the real part of a measured frequency response function is plotted versus the imaginary part of the same frequency response function with frequency as the independent parameter. Often used in modal analysis and servo system design. Also referred to as an Argand Diagram (*See* FREQUENCY RESPONSE FUNCTION, MODAL ANALYSIS).

NYQUIST FREQUENCY. One-half the sampling frequency, that is 1/2 of $1/\Delta t$ (*See* SAMPLING INTERVAL). The Nyquist frequency establishes maximum frequency which can be present in the data and still avoid aliasing (*See* SAMPLING FREQUENCY, ALIASING).



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POLES. The roots (possibly complex) of the denominator of a ratio of polynomials (possibly complex). They are used in continuous or discrete system transfer function analysis and describe the impulse response characteristics of systems. *See* ZEROS.

POWER. A function which varies as the square of a measured quantity. Power in this sense is not necessarily physical power.

POWER SPECTRAL DENSITY (PSD) FUNCTION. Also called the AUTO SPECTRAL DENSITY. A real-valued continuous function of frequency, presented with frequency on the horizontal axis and density on the vertical axis, which is a special case of a "cross-spectral density function" where the signal is compared with itself. The following is a method for measuring the PSD, which can lead to a definition of PSD: a signal is passed through a contiguous set of bandpass filters with an effective noise bandwidth B. The mean square output of each filter divided by the bandwidth B is measured with an effective averaging time T. A plot of this normalized mean square output as a function of the filter center frequencies is an estimate of the PSD. The PSD can be defined as the limit of this process where the time T is increased without limit and the bandwidth B is reduced to zero.

An equivalent definition is: the Fourier transform of the autocorrelation function. It is also the expected value of the squared magnitude of the Fourier transform of a sample time history divided by the effective (noise) filter bandwidth. Note that for a finite time history the value of the power spectral density can only be estimated. The statistical error and the effective filter shape are essential parts of that estimate.

PRIMARY SRS. *See* SHOCK RESPONSE SPECTRUM.

PROBABILITY DENSITY FUNCTION. An accumulation of data samples plotted as a function of an independent variable of the samples, such as amplitude, and normalized for a specified integral value is termed a histogram. For a time-varying function x(t), the probability density, p(x), can be defined as the limiting case of a histogram as

$$p(x) = \lim_{\substack{\Delta x_i \to 0 \\ N \to \infty}} \frac{N_i}{N \Delta x_i}$$

where

| | |
|---|---|
| Δxi | = the quantization intervals of x |
| N | = the total number of samples |
| Ni | = the number of samples within Δxi . |

A term, p(x), the probability density function, is also called the frequency function of a sample distribution. To find the probability that a value falls within a specific range x1 to x2, we integrate through the range. Symbolically,

$$P(x_1 < x < x_2) = \int_{x_1}^{x_2} p(x)dx.$$

Since it is certain that every measurement must yield some real value, we must have

$$P(-\infty < x < \infty) = \int_{-\infty}^{\infty} p(x)dx = 1.$$

Note that the probability density function is always positive, being the derivative of the probability distribution function.



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RANDOM INPUT, UNCORRELATED. A random input is uncorrelated for a lag t if for every lag greater than t the autocorrelation function is zero. White random noise (equal energy at all frequencies, a constant PSD) will be uncorrelated for all lags greater than zero. (*See* AUTOCORRELATION, WHITE NOISE, PSD.)

RANDOM VIBRATION. Vibration whose instantaneous value cannot be predicted with complete certainty for any given instant of time. Rather, the instantaneous values are specified only by probability distribution functions which give the probable fraction of the total time that the instantaneous values lie within a specified range.

Notes: "random" means not deterministic, "white" means uncorrelated (flat PSD), "Gaussian" describes the shape of the probability density function, and "noise" usually means not the signal, but can mean the random process. These are all different, though related.

RANGE OF F. *See* FUNCTION.

REAL. In a complex signal, the component that is in phase with the excitation. In frequency domain analysis, it is the magnitude of the COSINE terms of the Fourier series, "Coincident, Co", as in CO-QUAD analyzer (*See* FREQUENCY DOMAIN).

REAL FUNCTION. *See* FUNCTION. A function whose range is the subset of real numbers.

REALIZABLE FILTER. *See* FILTER, DIGITAL.

REAL-TIME ANALYSIS. Analysis for which, on the average, the computing associated with each sampled record can be completed in a time less than, or equal to, the record length. In digital analyzers, the functions accomplished during the computing time should be specified; e.g., Fourier transform, calibration, normalizing by the effective filter bandwidth, averaging, display, etc. (*See* FOURIER TRANSFORM).

RECURSIVE FILTER. *See* FILTER, DIGITAL.

REFERENCE SPECTRUM. The desired PSD or SRS of a control signal.

RESIDUE. A coefficient in the numerator of the partial fraction expansion of a transfer function. The residue can be associated with the mode shape.

RESIDUAL SRS. *See* SHOCK RESPONSE SPECTRUM.

RESOLUTION. The discernible difference between one value and adjacent values in a measurement. In a digital signal analyzer it usually refers to the smallest time or frequency increment that can be discerned. In frequency domain measurements the frequency resolution is also called delta f ($\Delta f$) and is equal to the analysis bandwidth divided by the number of spectral lines measured. Since only periodic signals can be resolved to within
$\Delta f$, the "effective noise bandwidth" of a digital analyzer is probably a more meaningful measure of resolving ability in the frequency domain (*See* FREQUENCY DOMAIN).

RESONANCE. The enhancement of the response of a physical system to an excitation, The resonant frequencies are usually defined as those frequencies where a small change in the frequency of excitation in either direction will cause the system response to decrease. The term resonant frequency is also used (sometimes erroneously) to denote the imaginary coordinate of the poles of a transfer function, the undamped natural frequencies of a system, and the damped natural frequencies.

RINGING, FILTER. *See* FILTER, RINGING



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SHOCK RESPONSE SPECTRUM (SRS). The maximum responses of a series of uniformly damped single-degree-of-freedom (SDOF) systems to an applied transient waveform, plotted as a function of the natural frequencies of the SDOF systems.

*Primary SRS* -- Maximum responses are determined *during* the application of the non-zero input waveform.

*Residual SRS* -- Maximum responses are determined during the period *after* the input waveform has been removed,

*Maximum (Maximax) SRS* -- Maximum responses are determined from the maximum values of both the primary and the residual SRS.

*Acceleration SRS* -- Maximum responses are measured in acceleration units.

*Velocity SRS* -- Maximum responses are measured in velocity units.

*Displacement SRS* -- Maximum responses are measured in displacement units.

SIDE LOBE. A secondary response separated by a notch from the main or desired response. Usually refers to a filter spectral shape, particularly in digital filters that have complex structure (many notches and peaks) in the filter transition band (*See* FILTER, DIGITAL).

SIGNAL ANALYSIS. Process of extracting information about a signal's behavior in the time domain and/or frequency domain. Describes the entire process of filtering, sampling, digitizing, computation, and display of results in a meaningful format (*See* TIME DOMAIN, FREQUENCY DOMAIN).

SIGNAL-TO-NOISE RATIO. A measure of signal quality. Typically, the ratio of voltage or power of a desired signal to the undesired noise component measured in corresponding units.

SIGNATURE ANALYSIS. The method whereby a physical process or device is identified in terms of the frequency characteristics of the signal it generates.

SIMULTANEOUS SAMPLE & HOLD. In data acquisition systems, the technique of using separate sample and hold amplifiers for each channel. This allows simultaneous sampling on all channels, thereby eliminating any skew due to use of a multiplexer (*See* SKEW).

SINGLE DEGREE OF FREEDOM. *See* DEGREES OF FREEDOM.

SKEW. Difference between sample times of sampled data in different channels of a data acquisition device. This results from multiplexing a single sample and hold amplifier among the channels (e.g., if 10 channels are scanned in 5 ms, there is a 0.5 ms skew between channels). Can also be due to head misalignment in a tape recorder.

SKEWNESS. The third moment of the probability density function of a random variable. It is a measure of the asymmetry of the function about its mean. A Gaussian distribution has zero skewness.

SMOOTHING. The general process of removing noise from data. Typically, it is performed by filtering a signal (provided that noise components are different in frequency than the desired signal).

SPECTRAL DENSITY FUNCTION. *See* POWER SPECTRAL DENSITY FUNCTION.

STABLE AVERAGING. *See* AVERAGE, STABLE.

STATIONARY (and NON-). A condition of a random process identified by the invariance of its statistical moments with time; if only the first two moments (mean and mean square) do not change, then the random process is weakly stationary; if all moments vary, the process is nonstationary. The nth moment of a random process X is defined as the expected value of $X^n$.



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TRANSIENT. A finite duration change from one steady state condition to a second steady state condition. Usually the initial and final steady state conditions are zero.

TRANSIENT ANALYSIS. The analysis of the data when the excitation of a system is of finite duration; can be used to study the change from one steady-state to a second steady-state condition.

TREND. The deterministic signal when a long-period deterministic signal is added to a time history. Trends are usually unwanted signals which are removed from the data approximating them with polynomials.

TRIGGER. A stimulus that initiates a process.

TRUNCATION ERROR. *See* ERROR, TRUNCATION.

VARIANCE. The square of the standard deviation. The variance of a time history, x(t), is given by

$$Var(x) = \sigma_x^2 = \lim_{T \to \infty} \int_{-\infty}^{T} [x(t) - \mu_x]^2 dt$$

where

μx = the mean value of x(t)
σx = the standard deviation of x(t).

VARIANCE REDUCTION FACTOR. The ratio of the variance of a statistical estimate computed with overlap processing to the variance computed without overlap processing.

VECTOR. A quantity that has magnitude, direction, and sense. A vector is commonly represented by a directed line segment whose length represents the magnitude and whose orientation in space represents the direction and sense.

VISCOUSLY DAMPED. Damping that is proportional to velocity.

VELOCITY SRS. *See* SHOCK RESPONSE SPECTRUM.

WEIGHTING FUNCTION. *See* WINDOW.

WHITE NOISE. Uncorrelated, flat PSD noise (*See* RANDOM VIBRATION).



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WRAP-AROUND ERROR (CIRCULAR CONVOLUTION ERROR). When the convolution in the time domain (multiplication in the frequency domain) is performed on two transient signals, the result has a duration equal to the sum of the durations of the individual waveforms. When a discrete Fourier transform (DFT) is used to approximate the continuous transform of a transient, and the convolution is performed by multiplying DFTs, the time frame must be at least as long as the sum of the durations of the individual transients or an error known as "circular convolution" occurs. A DFT can be considered as one period of a periodic waveform, The response in excess of one period is "wrapped-around" into the beginning of the period (*See* TIME DOMAIN, DFT).

ZEROS. The complex frequencies for which the numerator of a system's transfer function becomes zero (*See* POLES, TRANSFER FUNCTION).

ZOOM ANALYSIS (COMPLEX DEMODULATION). A technique for examining the frequency content of a signal with a fine resolution over a relatively narrow band of frequencies. The technique basically takes a band of frequencies and translates them to a lower band of frequencies (demodulation) where the signals can be decimated to reduce the sample size. A standard DFT can then be used to analyze the data.

Note that the increased resolution of this technique requires a corresponding increase in the time record length. The sample rate is decreased by decimation, to reduce the number of samples in the time window, only after the demodulation.



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