3000+

Reference Manual

Larson Davis Inc.
1681 West 820 North
Provo, UT 84601-1341
Phone: (801) 375-0177
FAX: (801) 375-0182
www.larsondavis.com
I3000.01 Rev. D

Copyright

Disclaimer

The following paragraph does not apply in any state or country where such statements are not agreeable with local law:

Even though Larson Davis, Inc. has tested the software and reviewed its documentation, Larson Davis Incorporated makes no warranty or representation, either expressed or implied, with respect to this software and documentation, its quality, performance, merchantability, or fitness for a particular purpose. This documentation is subject to change without notice, and should not be construed as a commitment or representation by Larson•Davis Inc.

This publication may contain inaccuracies or typographical errors. Larson Davis Inc. will periodically update the material for inclusion in new editions. Changes and improvements to the information described in this manual may be made at any time.

Warranty

Larson Davis warrants this product to be free from defects in material and workmanship for two years from the date of the original purchase. During the two year warranty period, Larson Davis will repair or, at its option, replace any defective component(s) without charge for parts or labor if the unit is returned, freight prepaid, to an authorized service center. The product will be returned freight prepaid.

Product defects caused by misuse, accidents, or user modification are not covered by this warranty. The warranties described above are exclusive and in lieu of all other warranties whether statutory, express or implied including, but not limited to, any implied warranty of merchantability or fitness for a particular purpose and all waranties arising from the course of dealing of usage or trade. The buyer's sole and exclusive remedy is for repair or replacement of non-conforming product or part thereof, or refund of the purchase price, but in no event shall Larson Davis (its contractors or suppliers of any special indirect, incidental, or consequential damages) whether the claims are based in contract, in tort (including negligence), or otherwise with respect to or arising out of the product furnished hereunder. Representation and warranties made by any person, including its authorized dealers, representatives, and employees of Larson Davis which alter or are in addition to the terms of this warranty shall not be binding upon Larson Davis unless in writing and signed by one of its officers.

Equal Opportunity Statement

Larson Davis is an equal opportunity employer and does not discriminate on the basis of race, color, religion, gender, national origin, disability or veteran status.

TABLE OF CONTENTS

Chapter 5 Selection of Averaging Parameters 5-1

CHAPTER

1 Introduction

The Model 3000+ Real-time analyzer and SLM performs two measurement functions simultaneously; that of a Precision Sound Level Meter and that of a real-time frequency analyzer.

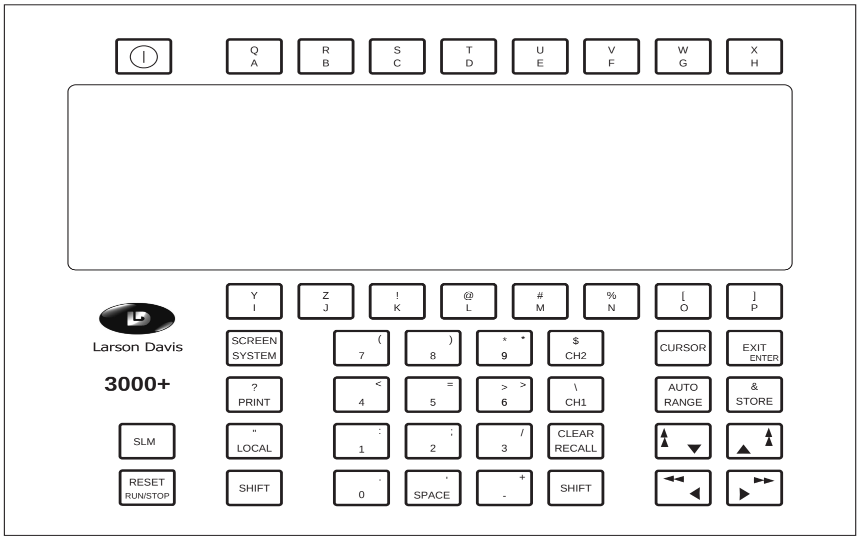
As a digital sound level meter, it simultaneously calculates the sound pressure level corresponding to the following detectors: RMS Slow, RMS Fast, Impulse and Peak. The Min and Max values of the Slow, Fast and Impulse detectors over the measurement period are maintained. At the same time, it calculates both LEQ and SEL integrated values. In certain versions of the Model 3000+, such as delivered to German users, the Min and Max values of the Impulse weighted sound level are replaced by the Taktmaximal (Fast weighted) 3 and 5 values.

As a real-time analyzer, it can perform single or dual channel frequency analysis using digital 1/1 and 1/3 octave bandwidths and FFT analysis using 100, 200, 400 or 800 line resolution. When equipped with the optional OPT 80 Acoustic Intensity Module, and using a Larson Davis intensity probe, it can perform acoustic intensity measurements in both digital (1/1 and 1/3 octave) and FFT filter formats. Using digital filters, it can generate statistics and Ln data using one or two channels.

In its autostore mode, it can store spectra as fast as 400 spectra/second to non-volatile memory, and subsequently display data in selected bandwidths as a function of time. Additional on-board software permits the calculation and display of reverberation time, sound transmission loss, NC, and STC. Along with these analysis capabilities, the 3000+ provides a high degree of versatility in data presentation on the screen. The user may control the vertical and horizontal screen formats (log/linear) as well as the display ranges. In conjunction with horizontal and vertical display expansion capabilities to provide the highest visual resolution of data, vertical window and horizontal pan functions may be used to move the expanded data blocks for easy examination.

Front Panel Controls

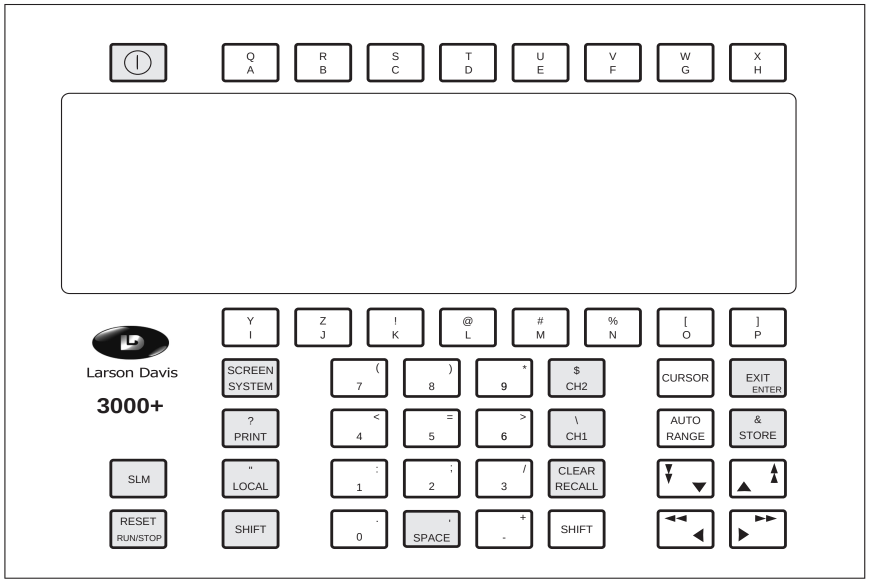
Figure 1- 1**Front Panel**



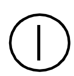
Some of these hardkeys have a label imprinted on the upper and lower levels of the key face. When simply pressing the key, the action associated with the lower level label is invoked. The action associated with the upper label is invoked by pressing the **SHIFT** key prior to pressing the key itself.

Dedicated Hardkeys

Figure 1-2 ***Dedicated Hardkeys***



The above illustration highlights a number of hardkeys on the front panel of the Model 3000+ whose functions are as follows:

<u>HardKeys</u>	<u>Hardkey Functions</u>
	Turn OFF analyzer Turn ON analyzer. When pressed simultaneously with the **SHIFT** key, produces a hard reset and re-boot
SCREEN	Adjust SCREEN angle and control backlight
SYSTEM	Display SYSTEM Menu
LOCAL	Put analyzer into LOCAL control mode
RESET	RESET data buffers.
R/S	Run/Stop of analyzer
PRINT	PRINT a hardcopy of data
SHIFT	SHIFT key to activate upper letter/symbol/word on hardkeys. Press **SHIFT**, release, then press hardkey

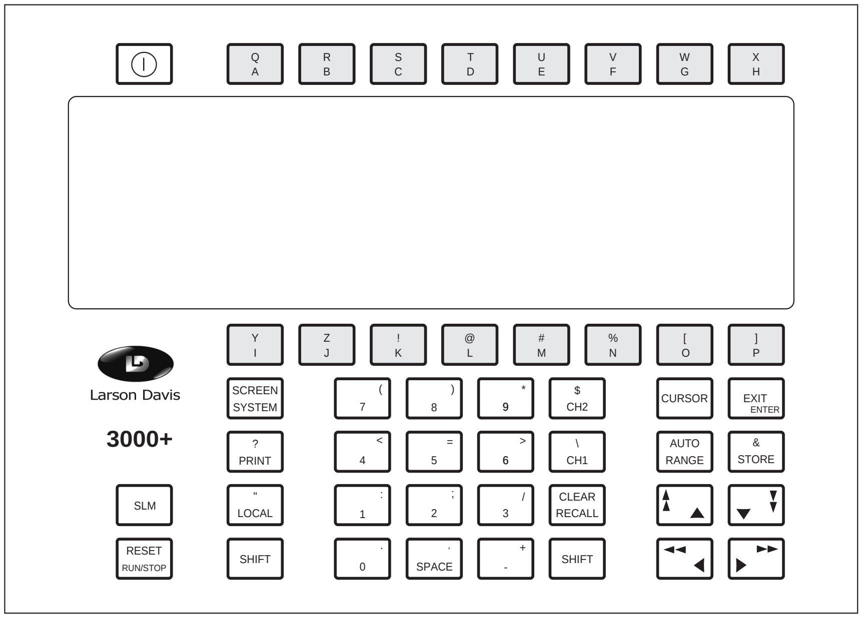
HardKeys	Hardkey Functions
SPACE	This key initiates a macro, previously created and stored by softkeys (I-P) by the user. When a data entry field is open, this key represents a space, or when used with the **SHIFT** key, a comma.
EXIT	EXIT from a softkey menu to a higher level menu. Also used to enter alphanumeric data after it has been input into the open data field on the upper right of the screen (e.g. when writing a note or entering a value of linear averaging time)
STORE	STORE displayed data block
RECALL	RECALL and display a stored data record
CLEAR	CLEAR the alphanumeric string in the open data field on the upper right of the screen (e.g. when editing a note)
AUTO	Activate input AUTOranging
CH 1	Select input connector 1 (nearest the right side of the top panel of the 3000+) for the input signal to the sound level meter and frequency analyzer functions indicated by the message "Input 1" on the right side of the screen, third line down.
CH 2	Select input connector 2 (nearer the center of top panel of the 3000+) for the input signal to the sound level meter and frequency analyzer functions indicated by the message "Input 2" on the right side of the screen, third line down.
SLM	Brings the Wide Dynamic Range Sound Level Meter [WDR SLM] Menu to the display of the 3000+ for setup and data display.

ASCII Hardkeys

With the exception of the hardkeys listed above, the remaining hardkeys on the front panel of the Model 3000+ are imprinted with two different ASCII characters (number, letter, character or space). One role of these keys is to input alphanumeric data when naming data files and when writing messages into the data block note fields previous to storage. Whenever such a data field is open at the upper right of the display, indicating that alphanumeric characters are to be entered, these keys will perform that function. When the upper character on the key is desired, the **SHIFT** hardkey must be pressed prior to pressing the key itself.

Softkeys

Figure 1-3 ***Softkeys***



The hardkeys aligned horizontally above and below the display, as illustrated above, play a major role in the operation of the 3000+. When the instrument is in operation, one of a variety of different softkey Menus will be displayed on the screen which will place a series of alphanumeric labels directly below (upper row) and above (lower row) these keys. There will not necessarily be a label for every key; some may be blank. We refer to these as programmable keys, or softkeys, because the role of each hardkey is to enable an action or activity associated with the particular label which is displayed above or below it. Thus, the role of each key will change as the Menu being displayed changes.

In some cases pressing a softkey will result in a specific action, such as opening a data entry field on the display so that a numerical or an alphanumeric value may be entered. We have adapted the convention that the softkey label be written in lower case letters when the action of the softkey is to call for an alphanumeric entry. In other cases, pressing a softkey will cause the Menu displayed on the screen to be changed to another Menu with a different set of softkey labels.

In this manual, when we refer to a particular softkey we shall use the format **XXXX [Y]** where **XXXX** is the softkey label and **Y** is the lower alphanumeric character imprinted on the associated hardkey. For example, **AUTOSTR [P]** would refer to the key on the far right of the row below the display, which has the character “P” imprinted on it, and the label “AUTOSTR” displayed on the screen directly above it.

The Arrow Keys and associated Hardkeys

The four keys on the lower right of the 3000+ front panel with arrow symbols indicating upward vertical, downward vertical, left horizontal and right horizontal, play a very important role in the operation of the Model 3000+. The lower pair of keys, denoted by the left and right horizontal arrow symbols, can perform a variety of functions which are user-assigned by pressing particular hardkeys or softkeys. At any time, the assigned role of these horizontal arrow keys is indicated on the lower right of the display by a message preceded by an asterisk *.

When the analyzer boots up, the message will read “*dotted crsr”. Use the horizontal arrow keys to move the cursor across the screen. Single presses advance the cursor a single step in the direction indicated by the symbol on the key. Holding the key down will produce a series of cursor movements as if the key were being pressed repeatedly. Pressing the **SHIFT** key along with an arrow key invokes the action associated with the double headed arrow symbol on the upper portion of the key label. In this case, the first cursor movement will be larger than for the single headed arrow, and each subsequent movement will be even larger. With 1/1 and 1/3 octave filters it may not be necessary to use the double headed arrow keys, but with the large number of filters associated with FFT analysis it is best to use the double headed arrow keys to move the cursor near to the desired position, then use the single headed arrows for exact placement of the cursor. The horizontal arrow keys are also used for paging through stored data records as part of a data recall.

Cursor Control

If the horizontal arrow keys are assigned to some function other than controlling the cursor, pressing the hardkey **CURSOR** will assign these keys to control whichever cursor was last under the control of these keys. Pressing the **CURSOR** key while these keys are already assigned to control the cursor will bring to the screen the Cursor Menu for the selection of cursor type.

Range Control

Pressing the hardkey **RANGE** will assign the horizontal arrow keys to control the input range by changing the input attenuators, indicated by the message “*range XXX” on the lower right of the screen where XXX is the full scale amplitude. Each press of the left arrow key will decrease the full scale level by 10 dB while presses of the right arrow key will increase the full scale level by 10 dB increments.

Although there is no message indication on the screen to indicate it, the vertical arrow keys can also control the range setting; each press producing a 10 dB change. It is necessary to observe the displayed amplitude indicators on the left of the display to determine the modification of the full scale level in response to presses of these two keys.

Instrument Boot-up Procedure

Upon pressing the hardkey **ON**, the Model 3000+ will go through a boot-up procedure during which the following message will appear briefly on the screen;

> Press: [-] boot, [*] reset RAM, [>] disk, [/] RS232
> BOOT ROM VERSION X.XX (C) 2000 Larson
> Davis, Inc.

The message is to inform the user which keys to press during the pause to reboot the RAM or to reload the internal software via the floppy disk or the RS-232 interface. If no action is taken, the message will disappear after about seven seconds and the bootup will continue. To terminate the pause and continue immediately with the bootup press the hardkey **[-]** as indicated in the message.

When the 3000+ is first booted up from the ROM, the software is transferred into the RAM. From then on, unless the ROM is reset, the 3000+ will boot-up using the software stored in the RAM.

Just before the bootup procedure is complete, on the upper right of the display will briefly appear a message of the form “Version 6.XX © 2001”, followed by another message of the form “Vers SLM+A 4.45 SLM 1.05”. For the purpose of certification, the code versions associated with the sound level meter functions have been separated from the general operating and analyzer code. Once a sound level meter code version has been released, no modifications are made without changing the code version. Most certifications are made for a specific code number. Thus, while the general operational and analyzer code may be modified frequently to make improvements and add new features, the sound level meter versions are rarely changed once they have been shown by a certifying agency to be acceptable. The first message (Version 6.XX) shown refers to the general operating and analyzer code. In the second message (SLM+A) refers to the code version for the (SLM with parallel frequency analysis) function and SLM refers to the wide dynamic range sound level meter (WDRSLM) function.

Resetting RAM

Should the software in the RAM somehow become corrupted, operational difficulties could be experienced. In that case, during the interval the above message is displayed, the user could press the ***** key (which requires a press of the **SHIFT** key first to invoke the upper character on the key) to reset the RAM as indicated by the message. This will generate a reset of the RAM, followed by a re-boot of the 3000+ from the ROM. Since all data stored in the 3000+ will be lost when the RAM is reset, it is recommended that the user contact the Larson Davis customer service department before initiating this procedure.

Upgrading Firmware

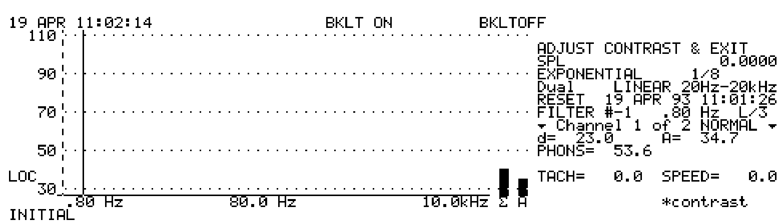
It is possible to upgrade the internal firmware of the 3000+ via the floppy disk drive. To upgrade via the floppy disk drive, place the disk containing the updated software file into the drive and, while the above message is being displayed, press the > key (which requires a press of the **SHIFT** key first to invoke the upper character of the key). This will load the ROM memory with the new software code, and initiate a re-boot using this software. A firmware upgrade is delivered on a single 3 1/2 inch disk. Following the upgrade, access the Reset Menu and manually reset all of the functions represented by softkeys in this menu before using the instrument.

Display Control

Setting Backlight and Viewing Angle

To adjust the screen display parameters, press **SCREEN** and note the message “ADJUST CONTRAST AND EXIT” on the upper right of the screen. Also the message “*contrast” on the lower right of the screen will indicate that the horizontal arrow keys are now controlling the view angle of the LCD screen. Press these keys until the optimum viewing angle for the present user position is obtained.

Figure 1-4 ***Screen Menu***



Pressing the softkey **BKLT ON [C]** will turn the display backlighting ON and pressing **BKLT OFF [E]** will turn it

OFF. The backlight does increase the current draw on the battery, so it is recommended that the backlight be used only when necessary when operating the 3000+ from internal batteries. In order to conserve power for battery powered units, the backlight automatically turns off when no keys have been pressed for a period of four minutes. It turns on again as soon as a key is pressed. Press **EXIT** to exit from the Screen Menu.

Beeper Control

The Model 3000+ can produce an audio output, or beep, corresponding to the following conditions:

Step 1 The press of a hardkey or softkey

Step 2 An overload condition at one of the inputs

Step 3 An error condition

Step 4 Any combination of the above three

The beeper function can be programmed by the user from the I/O Menu, as explained in Chapter 4. The default condition is that all of the above activities result in a beeper output.

Power Supply

Battery Power

The Model 3000+ can operate for up to four hours in the Run mode using the removable NiCd battery pack supplied with the instrument. When it is On, but not Running, the power consumption is reduced by 40%. The current drain of the instrument is too high for it to work effectively with alkaline batteries. The supply voltage is displayed on the upper right of the screen for approximately eight seconds whenever the hardkey **SYSTEM** is pressed. When operating on batteries, this will be the battery voltage. When connected to an external DC power supply, this will be the power supply voltage. The voltage will then be displayed on the upper right of the screen. In order to accurately read the battery voltage level just after unplugging the AC/DC converter (see below), let the instrument run for an instant and stop it prior to performing that measurement.

While operating, should the battery voltage drop to below 6.9 volts, the flashing message “Recharge BATTERY soon!” will be displayed on the upper right of the screen. Should the battery voltage level be further reduced to below 6.5 volts, the instrument will be shutdown automatically since the processor might not function properly at that reduced voltage level. Just prior to the shutdown, the message “DEAD BATTERY-Shut Down” will appear on the upper right of the screen, accompanied by an audible beep.

DC Power

The Model 3000+ can be powered from an external 11-16 Vdc power supply plugged into the miniature phono plug located on the extreme left of the top panel. An AC/DC converter is delivered with the 3000+ to permit operation from mains power. When a DC voltage source is supplied, the supply voltage can be read in the same manner as described above for reading the battery voltage.

Charging Batteries

When an external DC voltage is supplied, typically using the mains powered AC/DC converter supplied with the instrument, a charging voltage is applied to the rechargeable battery pack within the instrument. A totally discharged pack will require approximately 15 hours for a full recharge.

Caution: Because of the charging voltage applied to the battery pack within the instrument under DC operation, only rechargeable batteries should be used.

For rapid recharging of 3000+ battery packs external to the instrument, Larson Davis offers the optional PSA013. This “smart” charger unit provides a regulated charge rate to optimize battery life and avoid overcharging. A discharged battery pack can be fully recharged by the PSA013 in 3.5 hours.

Microphone Connection

Screw the microphone firmly onto the microphone preamplifier (PRM902) and use the short length of microphone cable supplied with the 3000+ (EXA001.5) to connect the microphone preamplifier to one of the microphone input connectors on the top panel of the 3000+. Then, while holding the preamplifier such that it is aligned with the microphone holder, with the connector end toward the cylindrical microphone holder, back it slowly into the holder until it is firmly in place with the cable passing through slot.

If the microphone boom is not used, the user should be aware of the potential for error associated with improper microphone placement and take appropriate measures when designing an alternative microphone placement system such as using an external tripod mount.

Alternative Inputs

Accelerometers with Internal Electronics

The Larson Davis Model PRA951, which plugs into a microphone input, has a BNC connector on one end to which a cable is connected. This device provides a 2 mA current to drive accelerometers containing internal preamplifiers such as the ICP™ designs from PCB Piezotronics.

Charge-coupled Accelerometers

Charge-coupled accelerometers should be connected to the analyzer inputs through the high impedance Model PRM902 microphone preamplifier by replacing the microphone cartridge with either the adapter ADP005 (BNC male cable connection) or the adapter AD007 (microdot male cable connection).

AC Outputs

There are two connectors on the top panel producing AC output signals; AC 1 and AC 2 as indicated on the rear panel label. The load impedance should be at least 2 kΩ. See the illustration on page 1-15.

Single Channel Standard Analysis Mode

When the 3000+ is configured to the single channel Standard Analysis Mode, the input signal may be applied to either Channel 1 or Channe l2. This is indicated by the message “Input 1” or “Input 2” on the right side of the screen, third line down from the top. The same AC output will be obtained from both AC 1 and AC 2, and this AC signal will be frequency weighted by the user-selectable analog input filter, as indicated by the message on the right side of the screen, third line down from the top, far right side. This could be A-weighting, C-weighting, or linear weighting with one of the various available combinations of highpass and lowpass filters.

Dual Channel Standard, Cross or Intensity Analysis Mode

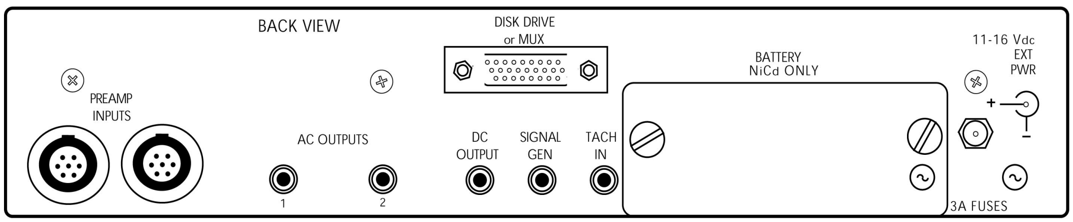
When the 3000+ is configured to the dual channel Standard, Cross or Intensity Analysis Mode, the signal produced from AC 1 will correspond to the Channel 1 input signal and the signal produced from AC 2 will correspond to the Channel 2 input signal. Both signals will be frequency weighted by the same choice of user-selectable analog input filter, as indicated by the message on the right side of the screen, third line down from the top, far right side. This could be A-weighting, C-weighting, or linear weighting with one of the various available combinations of highpass and lowpass filters.

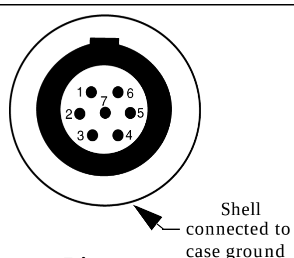
Back Panel and Side Panel Connections

This next section explains the connections that can be made to the 3000+.

Back Panel Connections

Below is an illustration of the back panel with an explanation of each of the connections.





Preamp Pin-out

1. No connection
2. Signal ground
3. Bias supply (0/28/200 Vdc)
4. Signal input
5. No connection
6. Power supply positive (+35 Vdc)
7. Power supply negative (+0.6 Vdc)

Preamp Inputs

The two preamp inputs are 7-pin LEMO connectors designed to work with the Larson Davis PRM902 microphone preamplifier and PRA951 ICP current source. A direct input connection (electrical) can be made by using an adapter cable (part number CBL094).

AC Outputs

Two AC outputs are supplied, AC1 and AC2, utilizing a mono 1/8” phone jack. The AC maximum voltage out is 5 Volts peak. The load impedance should be at least 2 kΩ.

DC Output

The DC output uses a 1/8” phone jack and produces a DC voltage proportional to a user-selected frequency or sound level meter parameter. The load impedance should be at least 2 kΩ. Full scale is represented by 4.5 volts, decreasing 1 volt/20 dB. See chapter 4 for a more detailed explanation.

Signal Generator

The signal generator output connector is a mono 1/8” phone jack. If OPT 10 is installed it provides pink or white noise. If OPT 11 is installed it provides swept frequency sine and a pulse generator in addition to pink and white noise. The load impedance should be at least 6 kΩ

Tach In

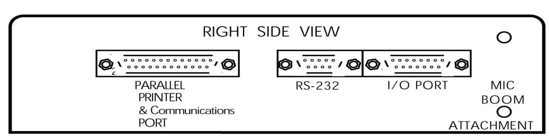
The Tach input uses a 1/8” mono phone jack. This input is designed to accept an analog pulse train (5 volt logic, positive edge trigger) whose frequency is proportional to the rotation rate of a rotating machine. See chapter 16 for a more detailed explanation.

Disk Drive or Mux

This connector is used to connect an external three and one half inch floppy drive (part number DVX003) or the model 2226 12 channel multiplexer. See page 1-38 for more information.

Side Panel Connections

The right side of the 3000+ contains three connectors: “Parallel printer & communications port”, “RS232 port”, and “I/O Port”. Below is an illustration of the side panel with an explanation of each of the connectors.



Parallel printer & Communications Port

25-pin connector for high speed parallel communications and for direct connection to a printer.

Printer Output: Centronics™ Parallel Port for use with a Hewlett-Packard™ compatible laser printer or an Epson™ compatible printer with graphics capability.

RS232 Port Pin-out

1. CD (input)
2. RD (input)
3. SD (output)
4. DTR (output)
5. Ground
6. No connection (DSR)
7. RTS (output)
8. CTS (input)

9 No connection (RI)

I/O Port Pin-out

1. Analog ground
2. A/D (input) #1

3.A/D (input) #2

4. A/D (input) #3
5. SPEED in or I/O #1
6. I/O #2
7. I/O #3
8. Digital ground
9. Opto out A

1. Opto out B

11. Common emitter
12. Vcc +5 volts
13. Common anode
14. Opto in A
15. Opto in B

RS232 Serial Port

9-pin serial communications port for computer interface with RS-232 compatibility.

I/O port

15-pin I/O port connector has 3 A/D inputs (0-5V) (8 bit), 2 optical input ports, and 2 optical output ports. See chapter 4 for a more detailed explanation.

SLM Mode

When the 3000+ is configured to one of the SLM Modes of operation, different weightings can be selected for the SLM and the Frequency Analysis functions. In this case, the output from AC 1 is weighted the same as the Frequency Analysis Function and the output from AC 2 is weighted the same as the SLM.

Front Panel Display Format

The setup of the 3000+ is completely described by the parameters shown on the screen. In addition, descriptive information concerning the data block is being displayed.

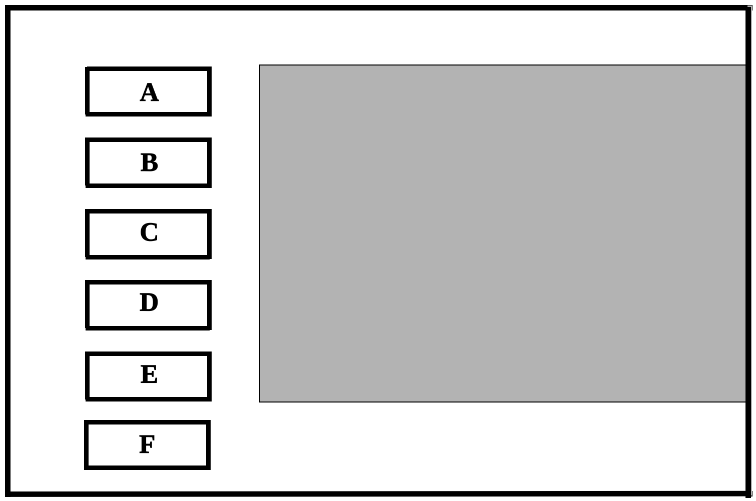
One way to assist you, the operator, to appreciate the control which you have over the manner in which the 3000+ measures and displays data is to list the many different messages which may appear on the screen, and to provide a brief explanation of each.

Use this section as a quick reference as well when the meaning of a particular message is not clear. We have used bold type to indicate messages which would appear literally as shown here, and regular type when the message will be an alphanumeric string which is not predefined by the system.

Messages Displayed on the Left of the Screen

The messages which may appear on the left of the screen will appear in six distinct positions, or locations, as shown in the Locations at the Left of the Screen on page 19. Listed below are the different messages which may appear within each of these locations.

Figure 1- 5 ***Locations at the Left of the Screen***



Location A, Displayed Data Type

The following messages may appear when Standard Analysis has been selected:

Leq	Equivalent Level Spectrum, an acoustic parameter
MAX	Maximum Spectrum
MIN	Minimum Spectrum
SEL	Single Event Level, an acoustic parameter
Max.S	Spectrum measured for highest broadband level
(blank)	Normal Spectrum

The following messages may appear when Cross Analysis has been selected:

MAG	Magnitude of a complex spectrum
PHASE	Phase of a complex spectrum
REAL	Real part of a complex spectrum
IMAG	Imaginary part of a complex spectrum
dBPP	Magnitude of the Cepstrum

Location B, vsREF Display Status and Statistics

vsREF	Indicates display is relative to a user-designated reference spectrum
STAT	Indicates the Statistics Mode (Ln) is active
(blank)	Indicates the display is not relative to a reference spectrum

Location C, Autostore Status

bTIME	Indicates autostore byTime is active
bTACH	Indicates autostore byTach is active
(blank)	Indicates autostore is not active

Location D, Frequency Trigger Status

TRIG	Indicates frequency trigger is enabled
(blank)	Indicates frequency trigger is inactive

Location E, Control Status

REM	Instrument is under remote control
LOC	Instrument is under local control

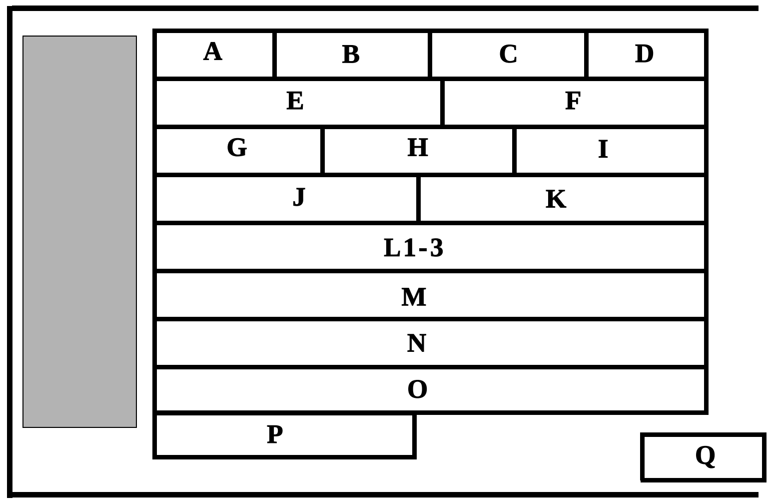
Location F, Active File

This location displays the name of the active memory file into which data will be stored and from which data will be recalled. Defined by the user from the Files Menu.

Messages Displayed on the Right of the Screen

There are 17 different locations on the right side of the screen, as shown in Locations at the Right of the screen on page 21, within which messages may be displayed.

Figure 1- 6***Locations at the Right of the screen***



Note Display Line

There may be another line displayed above these locations, which begins with the expression “Note:”. This is a user-defined note which can be stored with specific data blocks. However, because this line may not always appear unless some note operations have been performed, we have not included it as one of these locations. For the same reason, when in the following chapters we describe a particular parameter as being displayed on the right side of the screen, “Nth line down” we do not include the Note line in the count.

Location A, Units Name

The units name presently defined for the channel being displayed will appear in this location. This will be dBμV, SPL or a user-defined name created from the Units Menu.

Location B, Digital Differentiation or Integration and Bandwidth Compensation Status

$\mathbf{d}^2$ Double Differentiation (multiply by $-\omega^2$)

$\mathbf{d}^1$ Single Differentiation (multiply by $j\omega$)

$\int \mathbf{1}$ Single Integration (division by $j\omega$)

$\int 2$	Double Integration (division by $-\omega^2$)
$/\sqrt{\sim}$	Indicates that bandwidth compensation is active, producing amplitude as power spectral density. This symbol appears alongside those described above.
(blank)	Indicates that neither digital differentiation nor digital integration are active, and that the spectra are in RMS units (bandwidth compensation is inactive)

Location C, Digital Display Weighting and Status of Time Trigger

$\int$	Single Integration (division by $j\omega$)
$\iint$	Double Integration (division by $-\omega^2$)

A	A-Weighting Active
C	C-Weighting Active
USER	User Weighting Active
–A	Negative A-Weighting Active
–C	Negative C-Weighting Active
–USER	Negative User Weighting Active
(blank)	No Digital Weighting Active
ARM	Indicates that the Time Trigger is Active and Armed

Location D, Run Time

Elapsed runtime of analyzer, in seconds, since the last data buffer reset

Location E, Averaging Type

LINEAR SINGLE	Linear Single (seconds)
LINEAR REPEAT	Linear Repeat (seconds)
EXPONENTIAL	Exponential (seconds)
BT EXPONENTIAL	Constant Confidence with Exponential Averaging; Octave Bandwidths only

BT LINEAR — Constant Confidence with Linear Averaging; Octave Bandwidths only

EXPONENTIAL by N — Exponential Averaging based on number of spectra; FFT only

COUNT SINGLE — Linear Spectrum Averaging based on Number of Spectra; FFT only

COUNT REPEAT — Linear Repeat Spectrum Averaging based on Number of Spectra; FFT only

COUNT MANUAL ACCEPT — Linear Spectrum Averaging based on Number of Spectra, manual Accept; FFT only

Location F, Averaging Time

For Linear Single, Linear Repeat, and Exponential Averaging, a value in seconds will be displayed.

For BT Exponential and BT Linear Averaging, a value in units of Bandwidth-Time Product will be displayed

For Exponential by N, Count Single, Count Repeat and Count Manual Accept Averaging, a values representing Number of Spectra will be displayed.

Location G, Input Type

Input 1 — Indicates the 3000+ is in the SLM mode, or the single channel Standard Analysis mode, and that channel 1 is the active input.

Input 2 — Indicates the 3000+ is in the SLM mode, or the single channel Standard Analysis mode, and that channel 2 is the active input.

Dual — Indicates that the 3000+ is in one of the dual channel modes, such as dual channel Standard, Cross or Intensity, and that both channels 1 and 2 are active.

Location H, Analog Input Weighting

A-WEIGHT — Analog A-Weighting Active

C-WEIGHT — Analog C-Weighting Active

LINEAR — No Analog Weighting Active

Location I, Frequency Range between Highpass/Lowpass Filters with Linear Weighting Selected

1 Hz- 20 kHz

20 Hz-20 kHz

1 Hz-10 kHz

20 Hz-10 kHz

In Dual channel mode (STAND 2) these may be different for each channel. The frequency range indicated on the screen is for the channel being displayed, as indicated in Location M.

Location J, Operational Status

STOP	No Sampling in Progress
RESET	No Sampling, Data Buffer has been Reset
RUN	Sampling in Progress

Location K, Date and Time

These correspond to the beginning of a measurement. Add to this the run time to obtain the date and time of the completion of the measurement. The date format ddmmmyy. The time is in the 24 hour format hh:mm:ss.

Location L, Filter Status and Frequency at the Cursor Position

The message format for this location is a function of the active filter type.

L1	L2	L3

Octave Filters

L1	**FILTER #**	ANSI Filter Number
L2	Frequency Value	
L3	**L/1**	Long Filter, 1/1 Octave
	L/3	Long Filter, 1/3 Octave
	S/1	Short Filter, 1/1 Octave
	S/3	Short Filter, 1/3 Octave
	R/1	Reverse Filter, 1/1 Oct.
	R/3	Reverse Filter, 1/3 Oct.

FFT Filtering

L1 FREQ.

L2 Frequency Value

L3 Weighting of Time Buffer, AA indicates that anti-aliasing filters are active

R	Rectangular Weighting
H	Hanning Weighting
F	Flat Top Weighting
Z	Zero Pad Weighting
I	Impact Weighting on Channel 1; Rectangular on others
E	Impact Weighting on Channel 1; Exponential Weighting on others

Location M, Channel and Parameter Information

The format of the message in this location is a function of the Active Analysis Type.

Channel **X** of **Y**, Displayed Data Type where **X** is the Displayed Channel Number and **Y** is the Number of Active Channels

Possible displayed data types are as follows:

NORMAL, LEQ, MIN, MAX, SEL, Mx.Spec

Cross Analysis

Displayed Data Type, D- Channel Indication

When the data type is a single channel parameter such as autospectrum, the number displayed after the D- is the number of the displayed channel.

Possible single channel parameters are as follows:

Autospectrum, Auto Correlation, Impulse Response; Magnitude Cepstrum, Time, Weighted Time

When the data type is a cross channel parameter, the number displayed after the D- is the number of the channel which has been crossed with channel 1, the reference channel.

Possible cross channel parameters are as follows:

Cross Spectrum, Cross Correlation, Coherence, Transfer Function; H1, H2 or H3, Inverse Transfer; H1

Intensity Analysis

Displayed Data Type

Possible displayed data types are as follows:

INTENSITY, QUALITY, SPL, PARTICLE.V, POWER

Location N, Amplitude Data corresponding to Cursor Position

N1	N2

N1, Dotted Cursor Active

With the dotted cursor active, denoted by the message "*dotted cursor" on the lower right of the screen, the value displayed in location N1 will be the amplitude corresponding to the dotted cursor position, in the format "d = XX.X "to indicate that the level is for the dotted cursor.

N1, Solid Cursor Active

With the solid cursor active, denoted by the message "*solid cursor" on the lower right of the screen, the value displayed in location N1 will be the amplitude corresponding to the solid cursor position, in the format "s = XX.X" to indicate that the level is for the solid cursor.

N1, Both Cursors Active

With both cursors active, denoted by the message "*both crsrs" on the lower right of the screen, the value displayed in location N1 will be the level associated with the dotted cursor minus the level associated with the solid cursor. The format used is "Δ = XX.X" to indicated that the number represents a difference in levels.

N2, Dotted or Solid Cursors Active

With either the dotted or solid cursor active, the value in N2 corresponds to the total energy between the analog highpass and lowpass filters selected for the inputs modules, the frequency range of which is displayed in location I.

The value of the linear, or non-weighted, total energy is indicated in the format "Σ = XX.X"

The value of the A-Weighted total energy is indicated in the format "A = XX.X"

N2, Both Cursors Active

With both cursors active, the values displayed along with the Σ and A represent the total energy between the two cursors, rather than between the analog filters as is the case with either the dotted or solid cursor active.

Location O, Loudness Level

When the Model 3000+ is in the Standard Analysis Mode using 1/3 octave filters, this location will display the loudness level in units of phons and the Loudness in sones, as specified by ISO Recommendation R523, Method B.

The message "PH=?WGT?" indicates that an analog broadband weighting filter (A or C-weight) has been selected in the input path, making the measurement of Loudness impossible. With any other configuration of the 3000+, this location will be blank.

Location P, Data from Tacho

There are two inputs to accept pulse train signals from external transducers.The software-scaled values of the frequencies of these pulse trains are displayed in the format

TACH =XXX.X SPEED =XXX.X

Location Q, Status of the Horizontal Arrow Keys

This location indicates the assigned role of the horizontal arrow keys. Possible messages are as follows:

***dotted crsr**	Dotted Cursor Control
***solid crsr**	Solid Cursor Control
***both crsr**	Control Both Cursors Together
***OFF**	Cursor Control Off

***range**	Input Attenuator Control
***V.Offset**	Vertical Display Window Control
***H.Offset**	Horizontal Pan Control When Using an Expanded X-Axis.
***recall**	Data Block Recall Control
***contrast**	Screen Contrast Control
***new data**	Control of independent parameter during a paging process, such as selecting the frequency value for the display of vsTime records
***lifter**	Control of time domain editing while displaying liftered spectrum
***noise**	Control of Noise Generator Output Level
***MEMORY**	Files Menu, controlling Memory File Listing Highlight Position
***DISK**	Files Menu, controlling Disk File Listing Highlight Position
***RECORDS**	Files Menu, controlling Records Listing Highlight Position

Noise Floor

The noise floor of the Model 3000+ was measured by placing a Larson Davis Model ADP005 dummy microphone on the microphone preamplifier which provides a shunt capacitance equal to that of an actual 1⁄2 inch microphone (18 pF), and shorting the input.

Figure 1-7 ***Noise Floor in dB re. 1 microvolt 1/3 octave bandwidths***

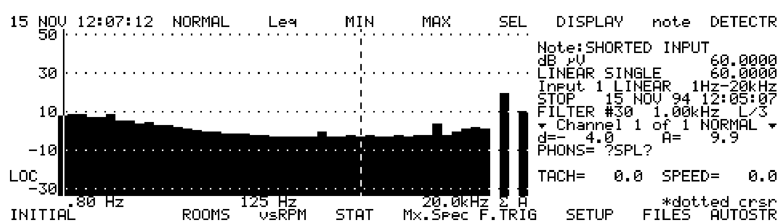


Figure 1- 8*Noise Floor in dB re. 1 microvolt*
200 line FFT

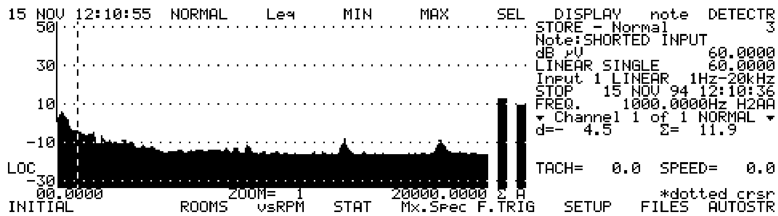
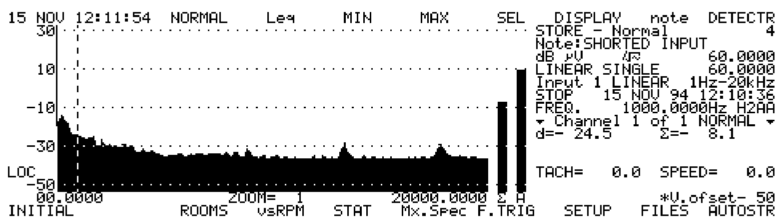


Figure 1- 9*Noise Floor in dB re. 1 microvolt*
200 line FFT; Energy Spectral Density



Model 3000+ Specifications

Input

Measuring Range: - 10 to 172 dB SPL with appropriate transducer

Input impedance: 10 GΩ || 0.5 pF with preamplifier

Direct input impedance: 100 KΩ || 60 pF

Polarization Voltage: 0, 28, 200 VDC

Gain: - 30 to 90 dB in 10 dB steps

Connector: 7-pin LEMO

Adapters: Available for use with ICP accelerometers and direct voltage inputs.

Direct input voltage range

DC	AC
-7 to +38 Volts	-7 to +7 Volts

Analog Input Filters

3-pole Chebyshev

Highpass: 1 Hz, 20 Hz

Lowpass: 10 kHz, 20 kHz

A-weight and C-weight Filters in accordance with the following:

ANSI S1.4-1983 Type 1

IEC 651 Type 1 and IEC 804 Type 1

Digital Characteristics

Digitization

16-bit A:D per channel

Anti-aliasing

Oversampling delta-sigma converter providing anti-aliasing stop band rejection >96 dB

Detector

Digital true RMS with 0.1 dB resolution

Dynamic Range

> 80 dB (> 100 dB using WDR SLM)

When used with LD 2540 or LD 2541 microphones and PRM902 preamplifier see the following table:

	2540	2541
Noise Floor A Wt (-20 to +60 dB range)	18 dB SPL (typical)	8 dB SPL (typical)
Noise Floor C Wt (-20 to +60 dB range)	22 dB SPL (typical)	12 dB SPL (typical)
Noise Floor Flat (-20 to +60 dB range)	25 dB SPL (typical)	15 dB SPL (typical)
Maximum measurement level	146 dB SPL	136 dB SPL
Maximum level with crest factor 3	149 dB SPL	139 dB SPL
Maximum level for Peak Max	149 dB SPL	139 dB SPL
Frequency range	3 Hz to 20 KHz	3 Hz to 20 KHz

Level Range:

	Primary Indicator Range	Dynamic Range (Noise floor to Overload
Standard and SLM+A	65 minimum (74 typical)	80 minimum (84 typical)
WDR-SLM	90 minimum (100 typical)	100 minimum (109 typical)

Reference range: 40 to +120 dB

Reference level: 114 dB SPL

Reference frequency: 1000 Hz

Reference direction: 0°

Amplitude Stability

± 0.1 dB

Amplitude Linearity

The greater of ± 0.05 dB or ± 0.005% of the maximum input signal.

Filters

Octave and Fractional Octave

1/1 and 1/3 octave real-time digital filters centered on base 2 frequencies

Satisfying or exceeding requirements for ANSI S1.11-1986 Type 0-AA and Type 1-D (user selectable) and IEC 61260.

IEC 61260 Class 0 (3 Hz to 12.5 kHz) and Class 1 (1 Hz to 20 kHz)

Lower Frequency Limit: 1 Hz

Upper Frequency Limit: 20 kHz (1-channel)
10 kHz (2-channels)

FFT

100, 200, 400, 800 line FFT analysis

Upper frequency limit: 20 kHz in 4 ranges (1 or 2 channels)

Maximum real-time frequency: 20 kHz (single or dual channel)

Zoom Capability

Real-time zoom: X512

Buffered* (non-real-time) Zoom: X32

* applies to dual channel FFT with full scale frequency of 20 kHz

Time Domain Windows (FFT analysis)

Rectangular, Hanning, Flat-Top, ZeroPad (w or w/o Bowtie correction), Impact, Exponential

Triggering

Continuous (free-run)

Digital remote (via interface)

Frequency domain: level in selected frequency band

Time Domain: Level in channel 1 (- 99% to +99% full scale) adjustable ch 1 delay (±)

adjustable ch 2 delay w/r to ch l (+ only)

Measured And Displayed Parameters

Sound Level Meter Mode

Simultaneous measurement of sound pressure level (A, C or Linear weighted) corresponding to the following detectors: RMS Slow, RMS Fast, Min and Max (RMS Slow and RMS Fast), Impulse, Leq, Peak, Taktmaximal 3 and Taktmaximal 5.

A time history trace of RMS Slow, RMS Fast, Leq, or Impulse is displayed in real-time, simultaneously with a frequency spectrum display.

Standard Analysis Mode, Octave and FFT

Normal, Leq, Max, Min and SEL Spectra; plus MaxSpec

Intensity Analysis Mode, Octave and FFT

Intensity, SPL, Particle Velocity, Quality (Int/SPL)

Cross Channel Analysis Mode, FFT

Autospectra, Cross Spectra, Transfer Functions (H1,H2,H3), Inverse Transfer Functions, Coherence, Coherent Output Power, Waveforms, Weighted Waveforms, Auto-correlation, Cross-correlation, Impulse Response, Cepstra, Liftered Spectra

Cross Channel Analysis Mode, Octave Bandwidths

Autospectra, Cross Spectra, Transfer Functions (H1,H2,H3), Inverse Transfer Functions, Coherence, Coherent Output Power

Digital Averaging

Octave Bandwidths

Linear Single: 0.0025 sec’s to 278 hours

Linear Repeat: 0.0025 sec’s to 278 hours

Exponential: 1/64 sec to 512 sec’s, binary sequence

BT/Exponential: 1 to 32,768 BT products,

binary sequency, Exponential averaging

BT/Lin: 1 to 32,768 BT products,

binary sequency, Linear averaging

FFT Bandwidths

Linear single, linear repeat, exponential,

Exponential by N (number of spectra),

Count single (number of spectra),

Count repeat (number of spectra),

Count manual (number of spectra, manual accept)

Digital Display Weighting

For Standard and Intensity Analysis Modes;

Octave and FFT Bandwidths:

No weight, A-weight, C-weight, user weight,

-(A-weight), -(C-weight), -(user weight)

Units

dB re 1 μV, dB SPL, dB re 1 pW/m2 (intensity),

dB re 1 pW (sound power), user definable (and named) units, log or linear scale, including:

single or double integration

single or double differentiation

scaling factor

User selectable bandwidth compensation

(e.g. power spectral density)

Memory

CMOS Non-volatile:

780 KB standard (typical capacity of 2976 1/3 octave or 426 800-line FFT spectra)

Floppy Disk

External 3 1/2” MS-DOS™ compatible floppy disk drive (part number DVX003), powered from the 3000+, is available as an option. Supports high density (1.44 MB).

Noise and Signal Generator

The optional noise generator (OPT 10) provides pink and white broadband random noise, with the On/Off synchronized with byTime autostore for automatic measurement of sound decay in rooms. It can also provide a sequence of digitally repeatable one millisecond duration noise bursts with the repetitions rate user adjustable.

The optional synthesized signal generator (OPT 11) provides swept sine (with tracking filter and feedback level control), dual frequency swept sine and a pulse generator in addition to the pink and white noise provided by the OPT 10 noise generator.

Digital Output and Control

Printer Output: Centronics™ Parallel Port for use with a Hewlett-Packard™ compatible laser printer or an Epson™ compatible printer with graphics capability.

Computer Interface: RS-232

I/O port: 3 A/D inputs (0-5V) (8 bit), 2 optical input ports, 2 optical output ports

Analog Outputs

AC output: 1 Volt rms (Units = dBμV)

Output impedance: 50Ω

Load impedance: ≥ 10 kΩ

DC Output: 0 to 4.4 Volts

Output impedance: 50Ω

Load impedance: ≥ 10 kΩ

Display Characteristics

Internal LCD

Type: Flat panel, supertwist with anti-reflective treatment

Backlighting: Electroluminescent

Contrast: Adjustable: dark to full sunlight

Size: Height 2.6 inch (6.60 cm)

Width 9.3 inch (23.62 cm)

Resolution: 128 X 489, with full graphics and alphanumerics

Environmental

Operating Temperature: 13 to 122° F (-10 to 50° C)

Storage Temperature: -13 to 158° F (-25 to 70° C)

Relative Humidity (non-condensing): 90% max at 104° F (40° C)

Physical

Size: 11” wide x 7.75” high x 2.4” thick
(28 cm x 19.7 cm x 6.1 cm)

Weight: 7.5 lb. (3.4 kg)

Power

Battery Power

Typical operating time in Run mode is 4 hours using removable NiCd pack supplied with the instrument, reduced by operation of the noise generator and the floppy disk drive. When On, but not Running, power consumption is reduced by approximately 40%. When using the AC/DC converter supplied with the instrument, the NiCd battery pack is charged while within the instrument. Typical charge time after total discharge is 15 hours.

DC Power

11-16 VDC. Typical current requirements:

.75 Amp @ 12Volt

Connector: 5.5 mm x 2.5 mm coaxial plug

AC/DC converter is supplied with the 3000+

EMC Compliance Testing

Compliance with the Electromagnetic Compatibility (EMC) directives

EC 60651 Amendment 2 (2000): Sound level meters.

IEC 60804 Edition 2.0 (2000) Integrating-averaging sound level meters.

EN 50081-1 (1993): Generic emission standard. Part 1: Residential, commercial, and light industrial.

EN 50081-2 (1993): Generic emission standard. Part 2: Industrial environment.

CISPR22 (1993): Limits and methods of radio disturbance characteristics of information technology equipment. Class B Limits.

FCC Part 15 Class B limits.

EN 50082-1 (1997): Generic immunity standard. Part 1: Residential, commercial, and light industrial.

EN 50082-2 (1995): Generic immunity standard. Part 2: Industrial environment.

Also complies with draft standard IEC 61260 Amendment 1: Octave-band and fractional-octave-band filters (draft date March, 2000), IEC 61672 Sound level meters Part-1 Specifications (draft date Jan, 2001) and Sound level meters Part-2 Pattern evaluations (draft date May, 2001).

The conditions called for in these EMC standards are listed below:

The 3000+ was tested with a 74 dB acoustic signal in SLM mode, A weight, Fast.

The reference orientation is with the 3000+ input connectors facing the emissions/immunity antenna. The microphone cable was not secured.

The following accessories are connected during testing of the 3000+: (2)PRM902 preamplifiers, (2)2541 microphones, (2) CBL092 mic cables, (5)CBL061 mini phone to BNC cables, (1) DVX003 floppy drive with cable, (1) PSA004 AC to DC power adapter.

The setting and configuration for greatest radio-frequency emissions is FFT 800 lines with all of the accessories listed above connected.

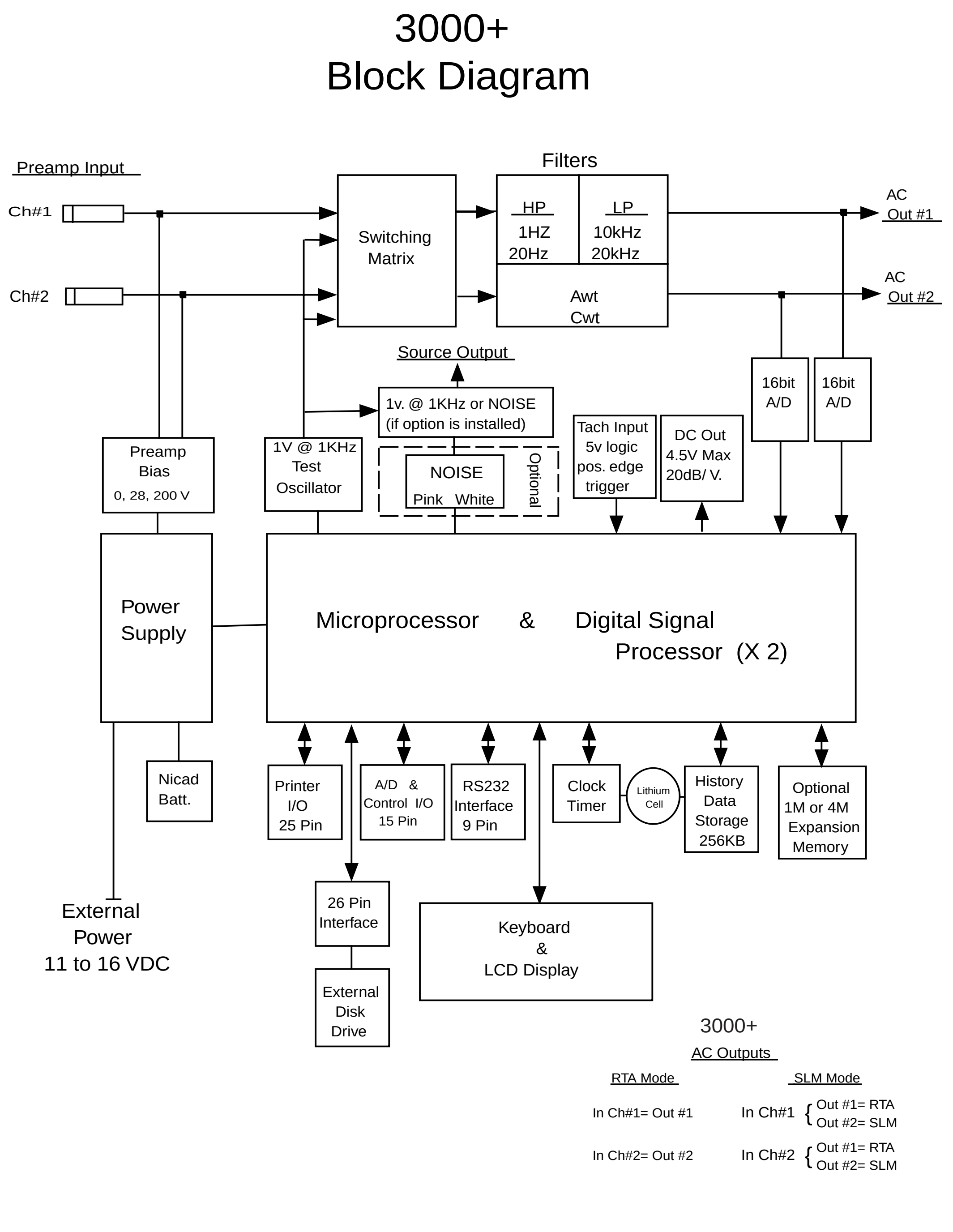
The mode of operation and connecting devices that produce minimum immunity to power and radio frequency fields are: SLM mode, A-weight, Fast with all of the accessories connected as listed above.

A short was applied to the input of the 3000+ for testing the bandpass filters (IEC 61260).

No degradation in performance or loss of functionality was found following the application of electrostatic discharges.

The method of mounting the instrument for acoustic testing is on a tripod with the microphone mounted in the microphone boom bracket.

3000+ Block Diagram



CHAPTER

2 Menu Structure For Instrument Operation

Softkey Menu Concept

The main user interface of the Model 3000+ consists of an interlinked network of displayed softkey Menus, each of which has associated with it a set of softkey labels. As explained in the Introduction, when the user presses a particular softkey, the result may be a direct action or it may result in the display of a different softkey Menu. We refer to a particular softkey using the format **XXXX [Y]** where **XXXX** is the softkey label and **Y** is the lower alphanumeric character imprinted on the associated hardkey. For example, **AUTOSTR [P]** would refer to the key on the far right of the row below the display, which has the character "P" imprinted on it, and the label "AUTOSTR" displayed on the screen directly above it.

References to front panel hardkeys are made using bold capital letters without any brackets, such as **SYSTEM**. To invoke the upper character of a hardkey, press **SHIFT** before pressing the key.

Analyzer Mode

When the Model 3000+ has completed its bootup sequence, it is configured as a single channel frequency analyzer. We say that it is "in the analyzer mode" and the set of softkeys which are displayed represent the Analyzer Menu. Because of the analysis flexibility inherent in the 3000+, offering the choice of octave and FFT bandwidths and such features as byTime autostorage, frequency domain triggering, noise generator control, room acoustics, etc., we draw a distinction between the setup and the operation of the analysis function by creating a hierarchy of Menus which differentiate between setup operations and general measurement and analysis operations. The following diagram indicates the structural interaction between the Analyzer Menu, the System Menu and the System submenus.

Analyzer Menu

↕

System Menu

(accessed by hardkey **SYSTEM**)

↕

Units ↕ **Filter** ↕ **Class** ↕ **I/O** ↕ **Sig.Gen or Noise** ↕ **Input** ↕ **Setup** ↕ **Files** ↕ **Resets** ↕ System Submenus

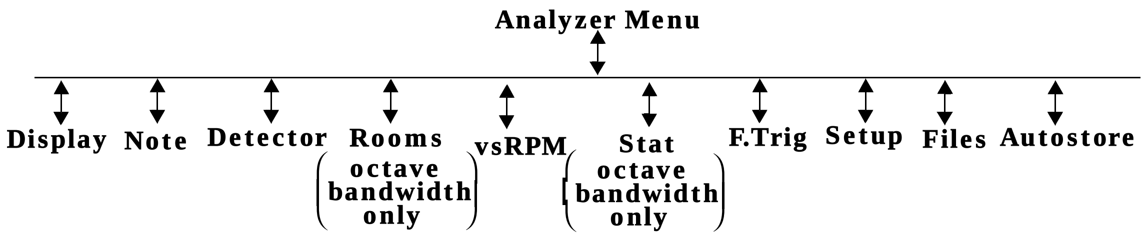
The functions which are performed within each of these Menus are as follows:

Menu Name	Menu Function
Analyzer	Operational menu for Analysis
System	Selection of number of input channels and path between the System submenus and the Main Menu.
Units	Select units, define and store user-defined units, perform calibration
Filter	Selection of Filter type and parameters
Class	Setup of Class Lines
I/O	Setup of I/O operations
Sig.Gen	Setup of digital signal generator
Noise	Setup of Noise generator
Input	Setup of Input modules
Setup	Storage and recall of user-created instrument Setups

Menu Name	Menu Function
Files	Creation, selection and directory of stored data Files
Resets	Menu for Resets

Submenus

The Main Menu has a number of submenus as shown in the following diagram.



The general functions performed within each of these Menus and Submenus are as follows:

Menu Name	Menu Function
Display	Selection of digital or user weighting for data Display
Notes	Annotate (add Notes to) data blocks before storage
Stat	Menu for measuring and displaying Statistics and Ln data
F.Trig	Setup of Frequency Trigger
Setup	Storage and recall of instrument Setups
Files	Creation, selection and directory of stored data Files
Autostore	Setup and selection of parameters for Autostore of data byTime
Rooms	Room acoustics post-processing routines
vsRPM	Access vsTach Autostore Display Menu

Sound Level Meter Modes

There are two sound level meter modes of operation available. The Wide Dynamic Range Sound Level Meter (WDRSLM) Mode serves solely as a precision sound level meter. This function is fully described in Chapter 3, Sound Level Meter Operating Modes.

Also available are the (SLM +A) Modes, single and dual channel, for which frequency analysis is performed in parallel with the sound level meter function. This function is also described in Chapter 3.

Because the Single and Dual Channel Sound Level Meter with Analyzer Modes provide frequency analysis in parallel with the sound level meter function, a brief description of a few of the most fundamental aspects of the frequency analysis function also appear. However, if the user is to fully appreciate all of the features of the frequency analysis function, he should become familiar with the remaining chapters of the manual. In some cases, the setup of the analysis function is best performed within the analyzer mode, after which the 3000+ can be returned to the SLM mode for measurement. When the 3000+ has been placed in the SLM mode, it can be returned to the single channel Standard Analysis mode to which it originally boots up by pressing the key combination SYSTEM, STANDRD [C].

Shift Menu

There is one more softkey Menu which is not accessed from one of the other Softkey Menus. We shall refer to this as the SHIFT Menu, and it is accessed by pressing the **SHIFT** hardkey. When this is done, the SHIFT Menu will appear on screen for about 4 seconds, during which time the user may press one of the softkeys. When exiting from this Menu, or if no softkeys have been pressed within 4 seconds, the instrument will return to the Menu which had been previously displayed.

A complete set of softkey Menus for the Model 3000+ is presented in Chapter 26.

CHAPTER

3 Sound Level Meter Operating Modes

A sound level meter is an instrument designed to measure and display the broadband sound pressure level of an acoustic signal. Very stringent performance specifications for sound level meters have been established internationally. The Model 3000+ is designed to satisfy or exceed the requirements of the following standards:

ANSI S1.4 1984 TYPE 1

ANSI S1.11 TYPE 0-AA and 1-D

IEC 60651 - 1979 and 60651 AM1 1993 TYPE 1

IEC 60804 TYPE 1

IEC 61260 - 1994

Although sound level meters can be equipped with filters to permit the user to measure the energy content of a signal as a function of frequency, this capability is outside the definition of a sound level meter and no reference to a frequency analysis measurement capability is contained in the international sound level meter standards.

In this chapter we discuss the setup and operation of the sound level meter functions available on the Model 3000+ in the following order:

- Single Channel Sound Level Meter with Frequency Analysis Mode 1/1 and 1/3 octave digital filters or FFT analysis to 20 kHz.

- Dual Channel Sound Level Meter with Frequency Analysis Mode 1/1 or 1/3 octave digital filters to 10 kHz.
- Wide Dynamic Range Sound Level Meter Mode; no frequency analysis provided.

Although the sections describing the Dual Channel Sound Level Meter with Frequency Analysis Mode and the Wide Dynamic Range Sound Level Meter Mode appear later in this chapter, it is strongly recommended that you read the preceding sections as well, since they contain many operational details, specifications and information such as microphone alignment, calibration, and noise floor measurement which are not repeated in the sections which follow.

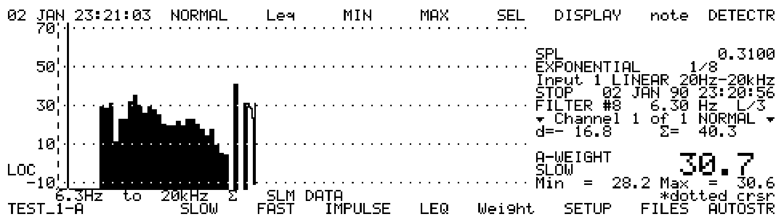
Sound Pressure Level Measurements: Single Channel Sound Level Meter with Frequency Analysis (SLM+A) Mode, Two Microphones

Setup

From the Main Menu, access the System Menu by pressing **SYSTEM**. Select the SLM+A mode by pressing **SLM+A [B]**. Since this section is concerned with the single channel version of the SLM+A Mode, press the key **#Chanls [A]** until the message "Channel 1 of 1 NORMAL" appears on the right of the screen, 6th line down. The dual channel mode corresponding to the message "Channel 1 of 2 NORMAL" is described later in this chapter.

Press **EXIT** to return to the SLM Menu shown in Figure 3-1.

Figure 3- 1 ***SLM Menu***



When the 3000+ is in the SLM+A mode, there are three lines of display on the lower right of the screen which indicate the setup of the sound level meter function as well as displaying the measured sound pressure level.

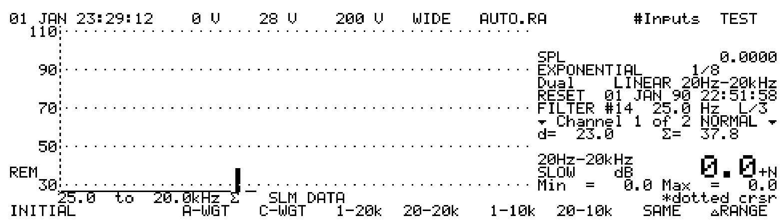
In the default setup as delivered from the factory, the Channel 1 microphone input will be active with a 200 volt DC bias voltage applied. This is the recommended bias voltage for use with Larson Davis air condenser microphones and the Model PRM902 1/2" microphone preamplifier. It is possible to change the bias voltage to 28 volts, or to turn off the polarization voltage for use with electret or prepolarized microphones as described below.

In the default setup an analog A-weighting filter is inserted into the input signal path and the 3000+ is thus set to display the A-weighted RMS Slow sound pressure level along with the Min and Max values of the RMS Slow level during the measurement interval. The A-weighted sound pressure level measured with the RMS Slow detector is the measurement most commonly called for in application standards.

Changing the Microphone Bias Voltage

Normally the Model 3000+ will be used with one of the Larson Davis air condenser microphones and the Model PRM902 1/2" preamplifier, which requires a highly stable DC polarization voltage. In the default setup, the 3000+ is set to use the Channel 1 microphone input with a 200 volt bias voltage. If an electret or prepolarized microphone is to be used, the polarization voltage should be switched off. Some users may wish to select a 28 volt bias voltage, either to reduce the microphone sensitivity, or to improve the microphone operation in extremely high humidity situations. To change the bias voltage, access the System Menu by pressing the hardkey **SYSTEM** and access the Input Menu, shown in Figure 3-2 by pressing **INPUT [K].**

Figure 3-2 ***Input Menu***



Note that the value of the microphone bias voltage presently active will be displayed on the upper right of the screen for approximately 4 seconds. To change the polarization voltage, press one of the following:

<u>Softkeys</u>	<u>Softkey Functions</u>
0 V [A]	Bias voltage OFF, for use with electret or prepolarized microphones
28 V [B]	28 volt bias voltage active
200 V [C]	200 volt bias voltage active

To return to the SLM Menu, press **EXIT** twice.

Changing the Microphone Input

To take the signal from either the Channel 1 or the Channel 2 microphone input connector, press either **CH1** or **CH2**. The active input will be indicated by the message on the right side of the screen, third line down from the top.

Changing the SLM Analog Filters

In the default setup the "Linear 20Hz-20kHz" is active. The SLM Input Menu shown in Figure 3-2, used to change the microphone bias voltage, is also used to select the analog filters for use in the signal path of the sound level meter. Press the key sequence **SYSTEM, INPUT [K]** to access this menu. To change the analog weighting, press one of the following:

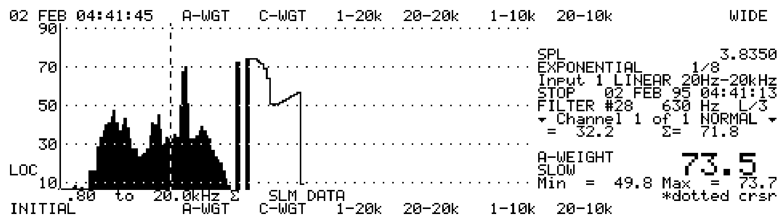
Softkeys	Softkey Functions
A-WGT [I]	Inserts an A-Weighting filter in the sound level meter signal path
C-WGT [J]	Inserts a C-Weighting filter in the sound level meter signal path
1 -20k [K]	Inserts a 1 Hz highpass filter and a 20 kHz lowpass filter in the sound level meter signal path
20 -20k [L]	Inserts a 20 Hz highpass filter and a 20 kHz lowpass filter in the sound level meter signal path
1 -10k [M]	Inserts a 1 Hz highpass filter and a 10 kHz lowpass filter in the sound level meter signal path
20 -10k [N]	Inserts a 20 Hz highpass filter and a 10 kHz lowpass filter in the sound level meter signal path

Note that the choice of analog filter selected for the sound level meter function is displayed on the lower right of the screen. To return to the SLM Menu, press **EXIT** twice.

Selecting SLM and Frequency Analysis Weighting

The weighting can also be modified from the Weight Menu, shown in Figure 3-3, which is accessed directly from the SLM Menu by pressing **Weight [M]**.

Figure 3-3 ***SLM Weight Menu***



This Menu permits the user to select the weightings for both the sound level meter function and the frequency analysis function from the same menu. These weightings are independent from one another. Unless one is changing the microphone bias voltage at the same time, most users will utilize the Weight Menu for changing the weighting function when in the sound level meter mode because it is more convenient to access (one keystroke from the SLM Menu) and because it permits modification of the frequency analysis weighting at the same time.

The weightings represented by the softkeys below the screen are for the sound level meter function, as seen by the fact that a change of this selection is reflected by a corresponding change in the SLM weighting displayed at the lower right of the screen.

The weightings represented by the softkeys above the screen are for the frequency analysis function, as seen by the fact that a change of this selection is reflected by a corresponding change in the frequency analysis weighting displayed on the right side of the screen, third line down.

For the linear weightings, the upper frequency of the spectrum display will always be the same as the upper frequency selected for the analysis. For A and C-weighting, the upper frequency will be 20 kHz. However, the value of the lower frequency used for the spectrum display can be selected between two different values, as follows:

Weighting	Lower Frequency Options, Hz
A-Weight	0.8, 25
C-Weight	0.8, 25
Linear; 1 - 10 kHz	0.8, 6.3
Linear; 1 - 20 kHz	0.8, 6.3
Linear; 20 - 10 kHz	0.8, 25
Linear; 20 - 20 kHz	0.8, 25

Upon selecting the frequency range, the higher of the two optional values of lower frequency will be used in the display. Repeated presses of the softkey **WIDE [H]** will toggle the lower frequency value between the two optional values.

Warm-up Time

A two minute warm-up time should be allowed before valid readings of Sound Level can be made.

Alignment of the Microphone Boom and Microphone/ Preamplifier

Microphone Boom Alignment

The microphone boom fixation is designed such that the main boom element extends outward from the upper right corner in a direction corresponding to a continuation of the diagonal between the lower left and the upper right corners of the front panel. When making a measurement of a specific noise source, whether the instrument is handheld or mounted on a tripod, the front panel should be approximately horizontal and the instrument aligned such that the main boom element is “aimed” at the noise source to be measured. Thus sound waves emanating from that source will impinge the instrument case along the front panel diagonal from the upper right corner to the lower left corner.

SLM Standards

The two major standards establishing performance specifications for sound level meters are ANSI S1.4-1984 and IEC 60651. In the United States, the ANSI standard is most generally utilized while the IEC standard is usually followed in other countries, particularly in Europe. While the Model 3000+ meets the specifications for Type 1 according to both standards, the selection of microphone type and orientation during a measurement may be different depending upon the standard being followed.

IEC 60651

The approach of the IEC standard is that the sound level to be measured is the result of a well-defined noise source whose position in space is known, and that the effects of reflections and other noise sources on the measured sound pressure level are secondary compared to the effect of the directly radiated sound energy. This is referred to as a free-field incidence measurement situation. In terms of this standard, the choice and orientation of the microphone should be such that the measurement will be most accurate in cases where the sound field is indeed radiated from that source. For this reason, most European users will select a free-field microphone for use with their Model 3000+.

ANSI S1.4-1984

The approach of the ANSI standard is that in many acoustic measurement situations the exact location of the noise source is not clear, such as observed in room acoustics situations where the sound field is often diffuse due to reflections of the sound waves from various solid surface and the existence of multiple sound sources. This is referred to as a random incidence measurement situation. Another instance is where the sound source is moving, such as in passby measurements of vehicle noise or aircraft operations. In terms of measuring a single noise source, the ANSI standard seeks to obtain the greatest accuracy for any possible position of the source with respect to the microphone. For this reason, most

American users will select a random incidence microphone for use with their Model 3000+.

Microphone/Preamplifier Alignment

When using the Model 3000+ for a sound level measurement, it is important to establish whether or not the measurement to be made is free-field or random incidence.

Free-Field Measurements

When the measurement is of the free-field type, the best results will be obtained using a free-field microphone (Larson Davis Models 2520, 2540, 2541 or 2570). In this case, the microphone should be aligned such that the sound waves radiated from the source impinge the microphone in a direction normal to the diaphragm. Thus, the axis of the microphone is “aimed” at the source. With the microphone boom aligned with the source as described in the section Microphone Boom Alignment, turn the microphone preamplifier holder such that the axis of the microphone and preamplifier are also “aimed” toward the source.

Figure 3-4 ***3000+ free-field response using Model 2541 free-field microphone at 0 degree incidence***

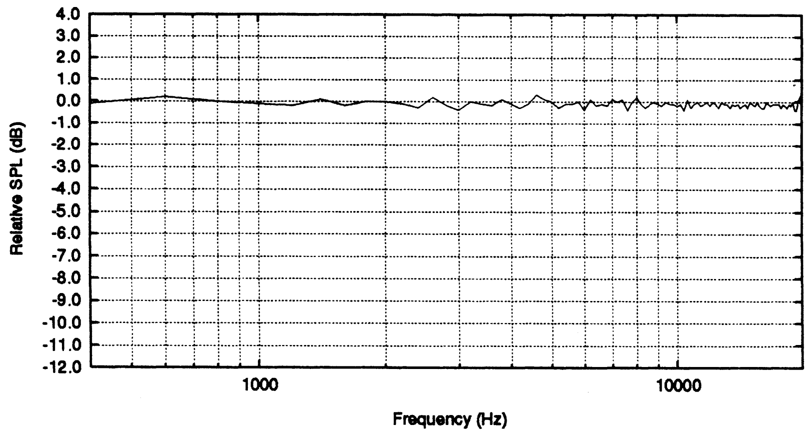


Figure 3-5 ***3000+ free-field response using Model 2541 free-field microphone at 30 degree incidence***

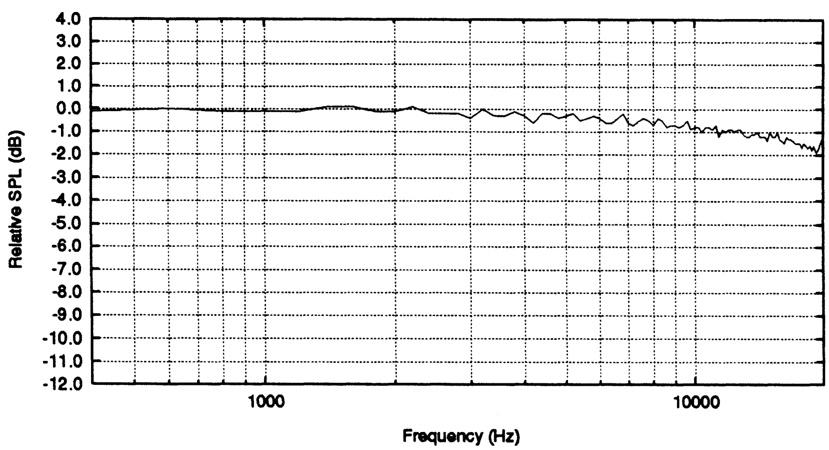


Figure 3- 6 ***3000+ free-field response using Model 2541 free-field microphone at 60 degree incidence***

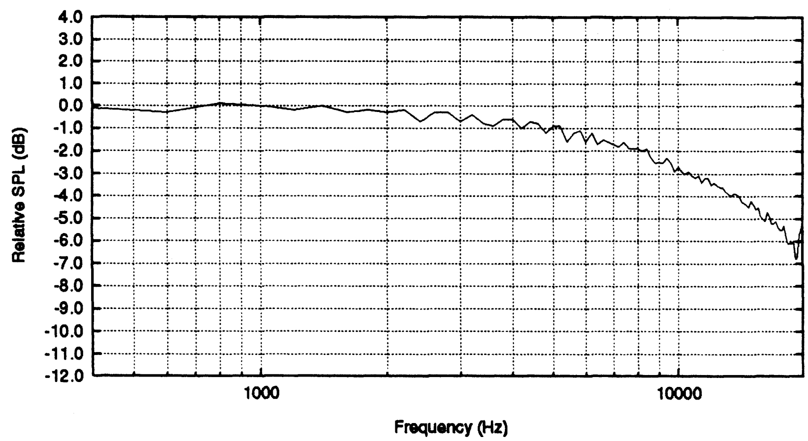


Figure 3- 7***3000+ free-field response using Model 2541 free-field microphone at 90 degree incidence***

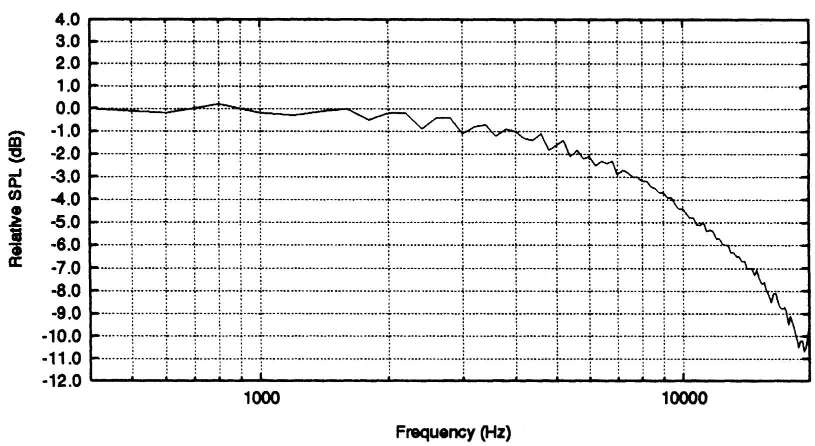


Figure 3-8 ***3000+ free-field response using Model 2541 free-field microphone at 120 degree incidence***

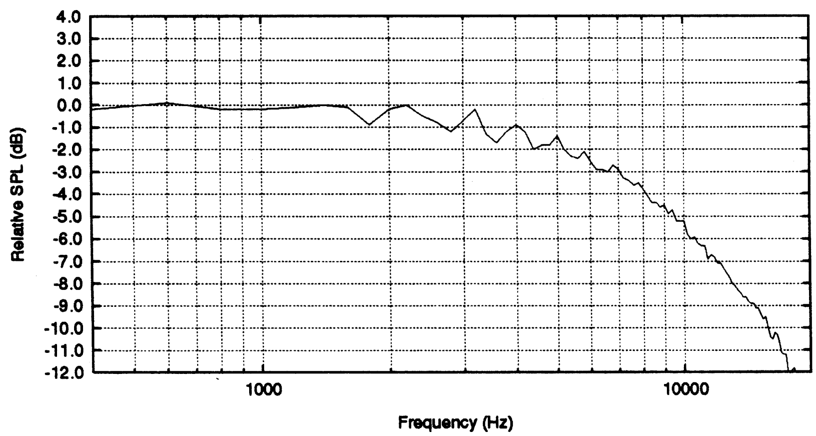


Figure 3-9 ***3000+ free-field response using Model 2541 free-field microphone at 150 degree incidence***

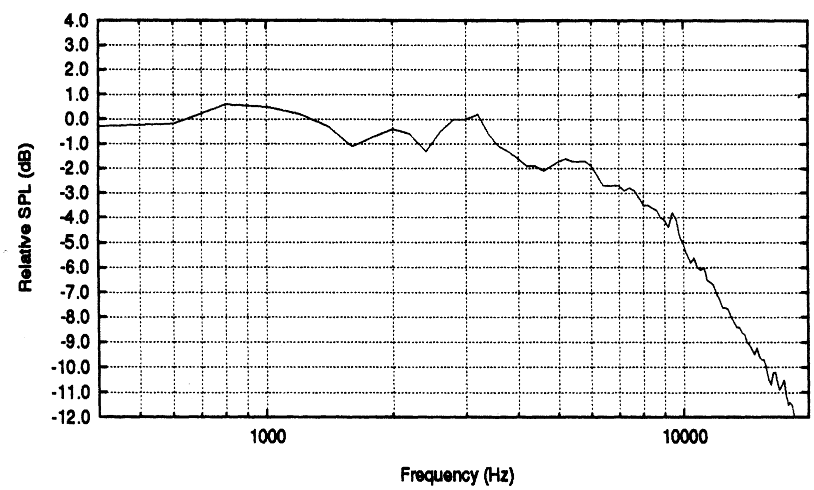


Figure 3-10 ***3000+ free-field response using Model 2541 free-field microphone at 180 degree incidence***

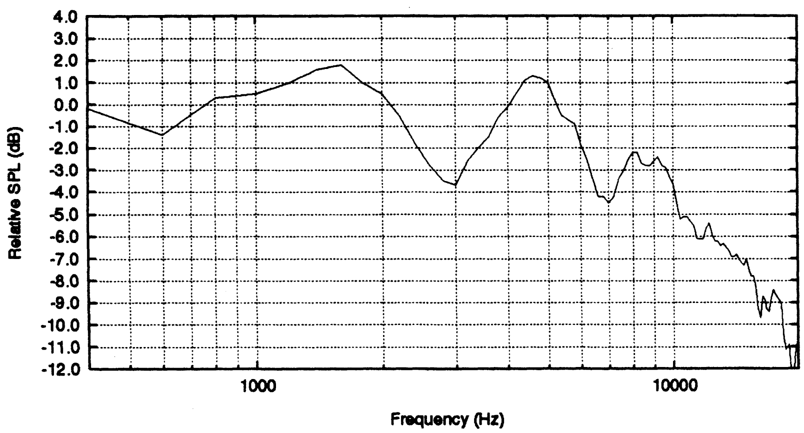
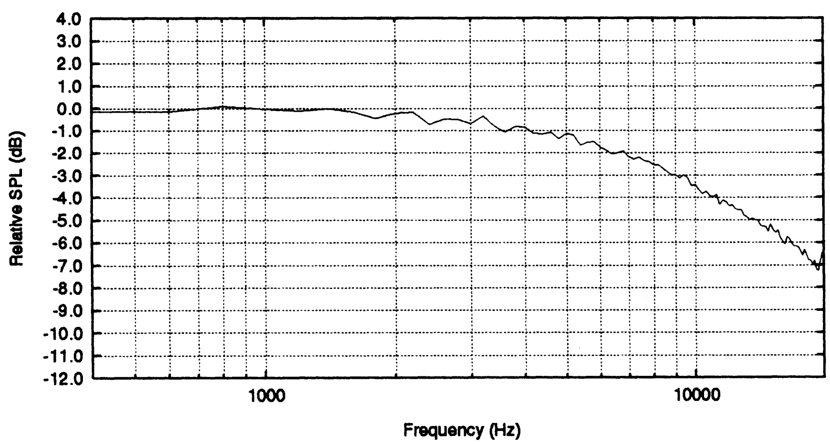


Figure 3-1 1 ***3000+ random response using Model 2541 free-field microphone (calculated from the free-field responses)***



Random Incidence Measurements

The best free-field response of a random incidence microphone is obtained when the sound waves radiated from the source pass over the microphone in a direction nearly parallel to the diaphragm, which is referred to as “grazing” incidence. Thus, in cases where it is necessary to make a free-field measurement with a random incidence microphone, the microphone should be aligned such that the angle between the microphone axis and the line between the microphone and the sound source is approximately 85°. As above, the microphone boom should be aligned with the source, but the microphone preamplifier holder turned to produce the desired angle. The angular response of the random incidence microphone is not so sensitive that the angle of incidence be exactly 85°, but this does produce the flattest frequency response. When the source is moving with respect to the microphone, such as during a vehicle passby measurement, this angle cannot be maintained for all positions, so a vertical microphone alignment is preferable.

When the measurement is of the random incidence type, the best results will be obtained using a random-incidence microphone (Larson Davis Models 2530, 2559 or 2560). In such a case alignment is of no concern, since the position of the radiating source cannot be clearly identified. In most cases, one would align the microphone preamplifier holder such that the axis is vertical, since this would minimize the effect of the body of the operator on the resulting sound field. Due to the relatively strong directionality characteristics of a free-field microphone, it is not recommended that they be used for precision measurements in a random-incidence measurement situation.

Figure 3-1 2 ***3000+random response using Model 2560 random incidence microphone***

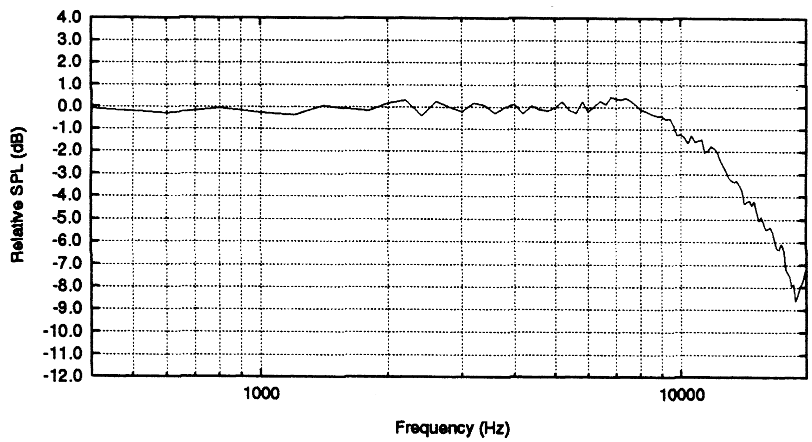


Figure 3-1 3 ***3000+ free-field response using Model 2560 random incidence microphone at 0 degree incidence***

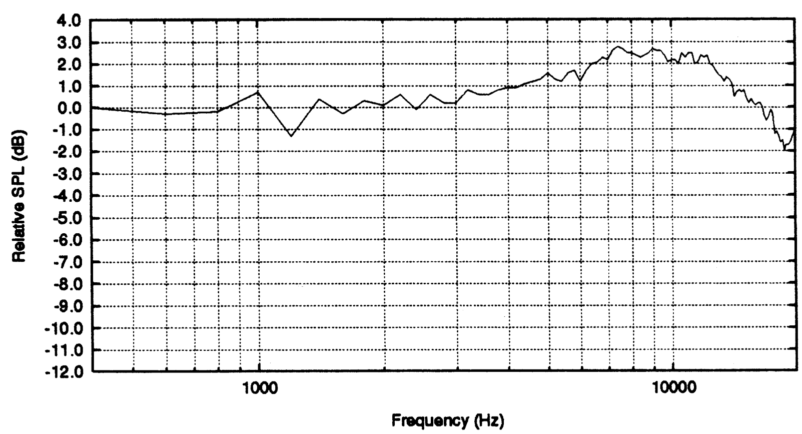


Figure 3-14 ***3000+ free-field response using Model 2560 random incidence microphone at 30 degree incidence***

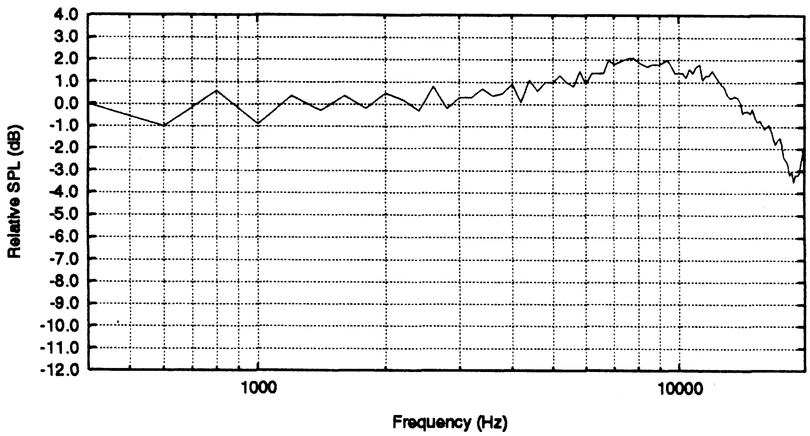


Figure 3-15 ***3000+ free-field response using Model 2560 random incidence microphone at 60 degree incidence***

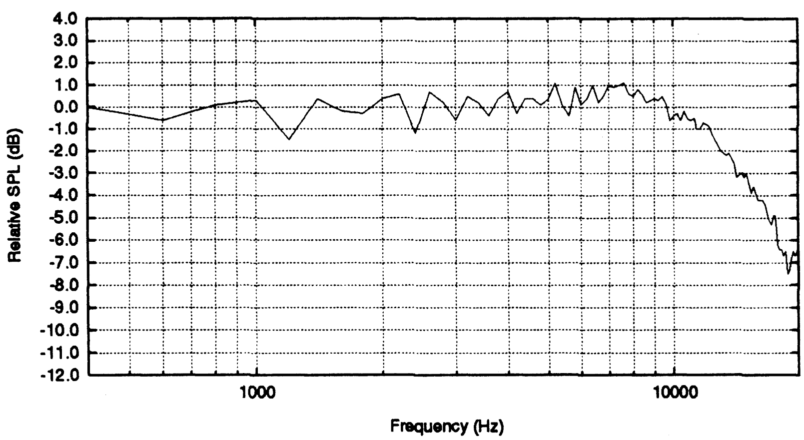


Figure 3-1 6 ***3000+ free-field response using Model 2560 random incidence microphone at 90 degree incidence***

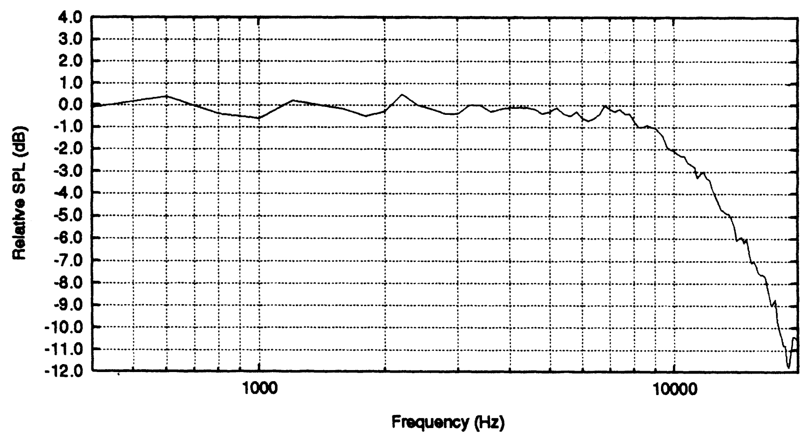


Figure 3-1 7 ***3000+ free-field response using Model 2560 random incidence microphone at 120 degree incidence***

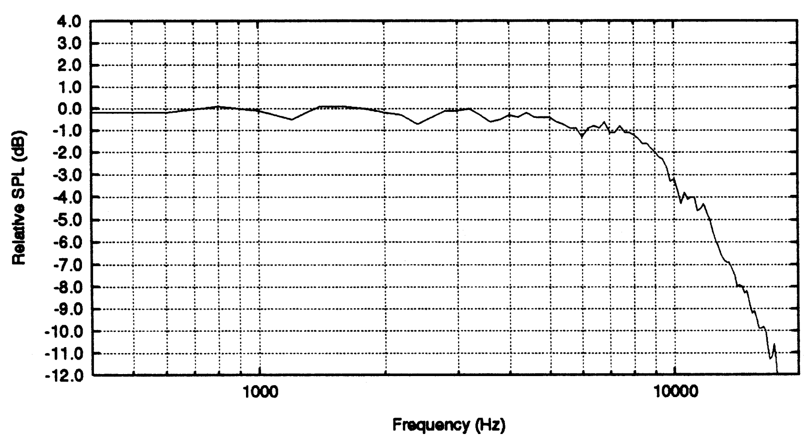


Figure 3-18 ***3000+ free-field response using Model 2560 random incidence microphone at 150 degree incidence***

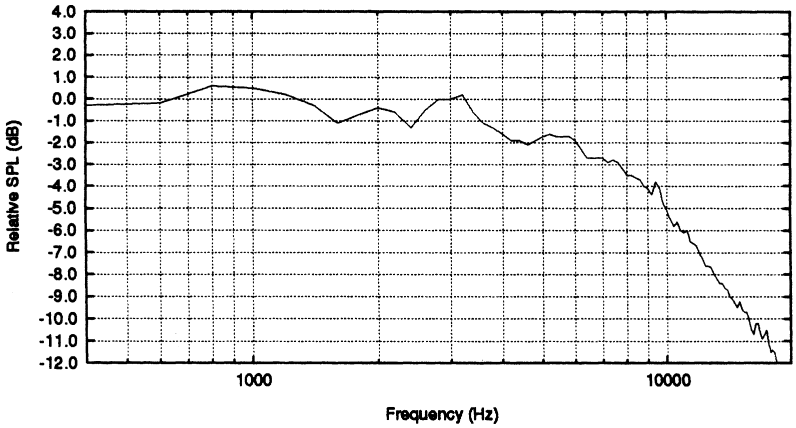
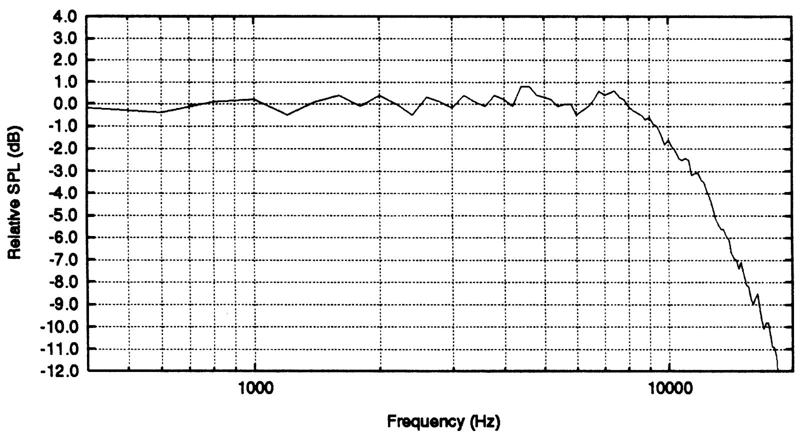
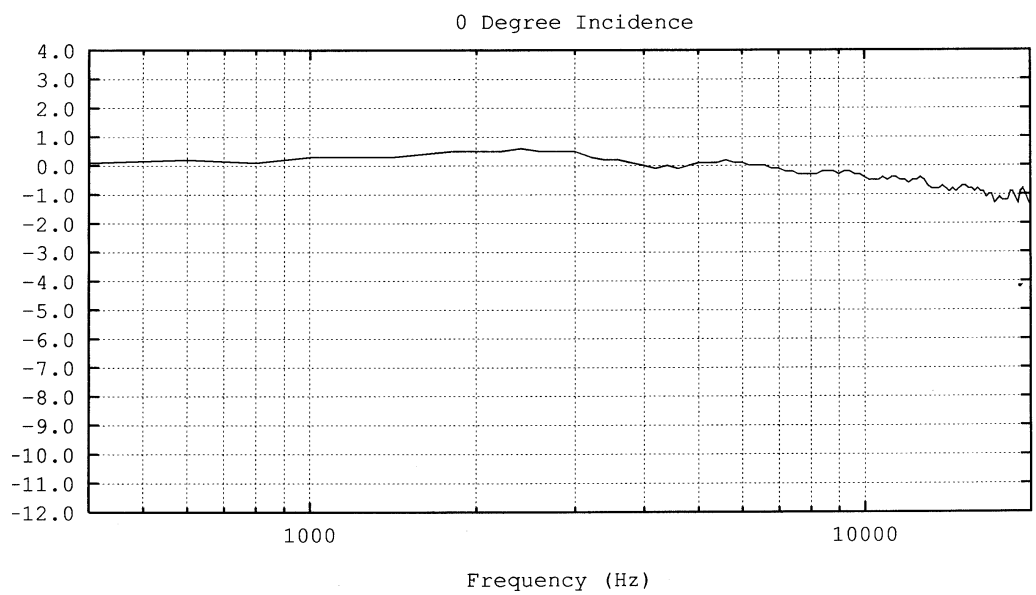


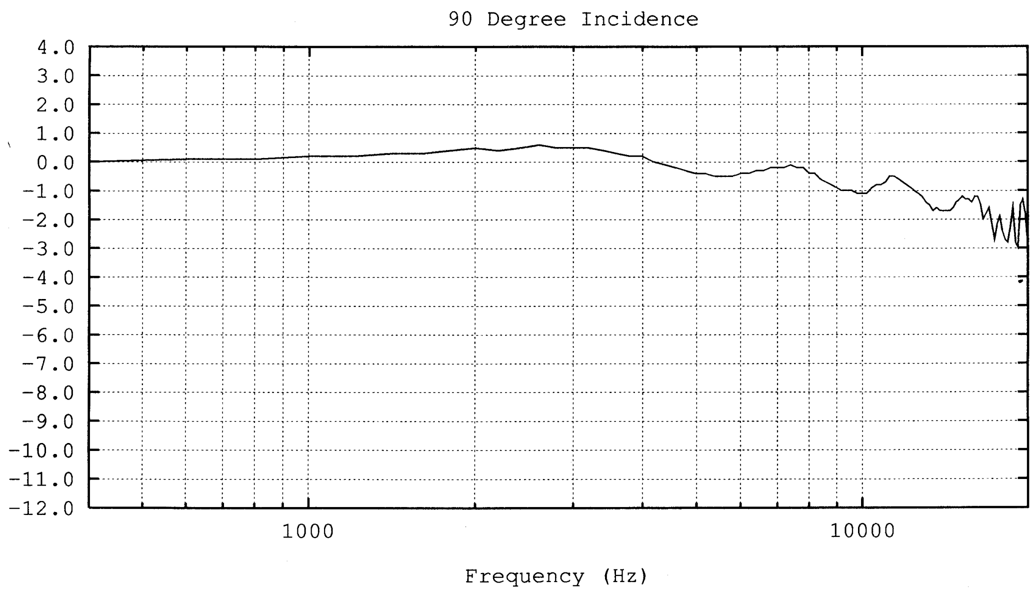
Figure 3-19 ***3000+ free-field response using Model 2560 random incidence microphone at 180 degree incidence***



Effect of Windscreen

The corrections which should be subtracted from the measured data when using the Larson Davis Model WS001 3 1/2" diameter windscreen with a 1/2" Larson Davis microphone are as indicated in the following graphs.





Position of Operator

When making a measurement, it is recommended that the observer be positioned as far behind and to the right of the instrument front panel as possible to minimize interference of the sound field at the microphone resulting from body reflections. Note that the viewing angle of the LCD screen may be adjusted to optimize viewing by an operator in this position. If possible, the instrument should be mounted on a tripod during a measurement. If the instrument is to be hand-held during the measurement, the user should hold the instrument as far away from his body as possible, and as far as possible to the left of his body centerline.

Making a Sound Level Measurement

Pressing the "RUN/STOP" key will start and stop a measurement. The sound pressure level corresponding to the SLM setup is displayed digitally on the lower right of the screen in large numbers. The height of the vertical bar nearest to the center of the screen is an analog indication of the same sound pressure level, and it will move up and down in response to variations in the sound level. As the measurement proceeds, the sound pressure level will be traced across the screen as a function of time in a manner analogous to a strip chart recorder.

The elapsed time of the measurement is indicated in seconds on the right side of the screen, first line down from the top. Repeatedly pressing the "RUN/STOP" key after beginning a measurement will pause, then restart the measurement without resetting the data buffer. Thus, the elapsed time will continue to increase and the integrated levels will represent data measured since the last reset of the instrument.

To reset the data buffers and set the elapsed time to zero, press **RESET**.

Adjusting the Input Gain

The hardkeys with the upward and downward vertical arrow symbols are used to control the input attenuators, and thus the vertical scaling of the display. Pressing the upward vertical arrow key will increase the full scale amplitude and pressing the downward vertical arrow key will decrease it. One can also use the horizontal arrow keys to adjust the input gain by first pressing **RANGE**. The gain can be changed with the 3000+ in either the RUN or the STOP mode. Whenever the gain is changed, the data buffer is reset and the elapsed time initialized to zero for another measurement. One would typically decrease the full scale value until the sound level trace is clearly visible, preferable in the upper half of the screen if possible, without overloading the input.

Overload Indication

An overload is indicated by an audible beep accompanied by the large inverse video message “OVER” on the screen. The inverse video overload message will disappear when the overload condition no longer exists. Since some of the measured parameters involve integration over time, the existence of an overload at any time during the measurement would be a source of error. To indicate that an overload had occurred during a measurement, the message “OVER CHANNELS:” will appear in the upper right of the display.

Autoranging

When measuring a stable sound level, the user may wish to utilize the Autoranging function. This is described in the section Autorange of Input Gain in Chapter 7. In the worst case, using flat weighting with the 1 Hz highpass filter selected, a time interval of 45 seconds should be allowed for filter settling after a range change. In practice, examination of the time history trace should indicate when the measured and displayed sound level has become stable.

Measurement Range

The range of sound pressure levels over which measurements in the SLM mode can be made to Type 1 accuracy are listed below for a range of Larson Davis microphones. The lower limit is established as being 10 dB above the measured noise floor in order to maintain the error to less than 0.4 dB. The upper limit is established as the level at which overload occurs when excited by a sinusoidal signal. For signals having a crest factor of 10, the overload will occur at a level 20 dB below the stated limit.

Figure 3-20 ***Using Larson Davis Model 2570 or 2575 1" microphones having a nominal sensitivity of 45 mV/Pa with a Model PRM902 preamplifier:***

Weighting	Measurement Range
A-weighting	13–135 dB
C–weighting	15–135 dB
20 Hz – 20 kHz	17–135 dB
1 Hz – 20 kHz	25–135 dB

Figure 3-2 1 ***Using Larson Davis Model 2541 or 2560 1/2" microphones having a nominal sensitivity of 44 mV/Pa with a Model PRM902 preamplifier:***

Weighting	Measurement Range
A-weighting	18–135 dB
C–weighting	23–135 dB
20 Hz – 20 kHz	25–135 dB
1 Hz – 20 kHz	35–135 dB

Figure 3-2 2 ***Using Larson Davis Model 2540 or 2559 1/2" microphones having a nominal sensitivity of 12.5 mV/Pa with a Model PRM902 preamplifier:***

Weighting	Measurement Range
A-weighting	36–148 dB
C–weighting	36–148 dB
20 Hz – 20 kHz	37–148 dB
1 Hz – 20 kHz	48–148 dB

Figure 3-2 3 ***Using Larson Davis Model 2520 1/4" microphone having a nominal sensitivity of 4 mV/Pa with a Model PRM902 preamplifier:***

Weighting	Measurement Range
A-weighting	47–157 dB
C–weighting	54–157 dB
20 Hz – 20 kHz	55–157 dB
1 Hz – 20 kHz	65–157 dB

Figure 3-24 ***Using Larson Davis Model 2530 1/4" microphone having a nominal sensitivity of 1.5 mV/Pa with a Model PRM902 preamplifier:***

Weighting	Measurement Range
A-weighting	54–166 dB
C–weighting	60–166 dB
20 Hz – 20 kHz	61–166 dB
1 Hz – 20 kHz	69–166 dB

Primary Indicator Range

The primary indicator range is defined by IEC 60651 and ANSI S1.4-1984 as a specified range of the indicator for which the sound level meter readings are within particularly close tolerances on level linearity. Linearity is measured using sinusoidal test signals.

The primary indicator range of the Model 3000+ SLM+A mode is 65 dB. This means that, although the dynamic range of the instrument as an analyzer is greater than 80 dB, measurements of sound level made using the instrument in the SLM mode will be within the Type 1 primary indicator range linearity specifications when the level is within 65 dB of the full scale value. During a measurement, whenever the measured sound pressure level drops to more than 65 dB below the full scale value, two question marks “??” will be displayed to the right of the sound level digital readout on the lower right side of the screen.

Selecting the Displayed Parameter

With the exception of the analog weighting and highpass/ lowpass filters, the Model 3000+ is totally digital. As such, simultaneous measurements are made using the following detectors: RMS Slow, RMS Fast, Impulse and Peak. The Min and Max values of the RMS Slow, Fast and Impulse detectors over the measurement period are maintained. At the same time, it calculates both LEQ and SEL integrated values. In certain versions of the Model 3000+, such as delivered to German users, the Min and Max values of the Impulse weighted sound level are replaced by the Taktmaximal (Fast weighted) 3 and 5 values.

Leq is a parameter used for the analysis of time-varying acoustic signals. It represents the steady level which, integrated over the measurement period, would produce the same energy as the actual signal. The time used for the calculation is the elapsed time since the last data reset.

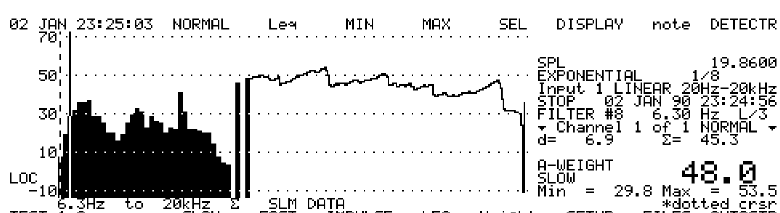
SEL (Single Event Level) is similar to Leq, except that it represents the steady signal which, integrated over a one second time period, would produce the same energy as the actual signal integrated over the elapsed time since the last data reset.

To select the desired display, press one of the following:

SLOW [I]

Produces a display of the RMS Slow level, along with the Min and Max values of the RMS Slow level since the last data reset, as shown in Figure 3-25. The averaging time of the Slow detector is 1 second.

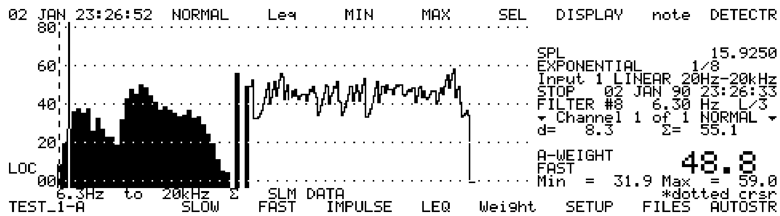
Figure 3-2 5***Slow Display***



FAST [J]

Produces a display of the RMS Fast level, along with the Min and Max values of the RMS Fast level since the last data reset as shown in Figure 3-26. The averaging time of the Fast detector is 1/8 second.

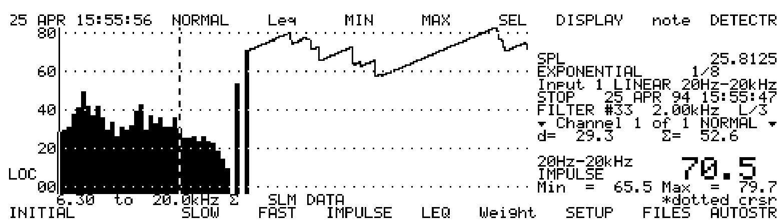
Figure 3-26 ***Fast Display***



IMPULSE [K]

Produces a display of the Impulse weighted level, along with the Min and Max values of the Impulse weighted level since the last data reset, as shown in Figure 3-27. The averaging time of the Impulse detector is 35 milliseconds, but it is also characterized by a very slow (3 dB/second) decay rate. For the versions producing the Taktmaximal 3 and 5 values, the weighting is Fast, even though the display of these is accessed by pressing the **IMPULSE [K]** softkey.

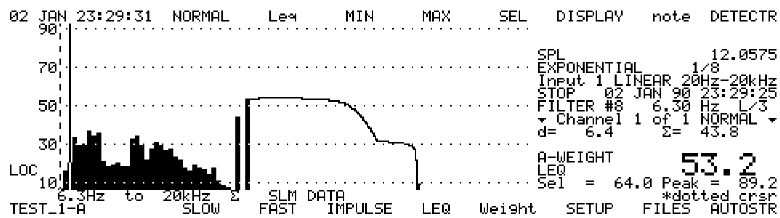
Figure 3-27 ***Impulse Display***



LEQ [L]

Produces a display of the Leq integrated level, along with the SEL integrated level and the maximum Peak level which has occurred since the last data reset as shown in Figure 3-28. The Peak detector has a rise time of 50 microseconds.

Figure 3-2 8*Leq Display*



Frequency Analysis Display

Since the Model 3000+ performs a single channel frequency analysis function simultaneous with sound level meter measurements, a frequency spectrum of the acoustic signal is presented on the left side of the screen at the same time the sound level is being displayed. Different values of analog highpass and lowpass filters may be inserted in the signal path of the frequency analysis function. The frequency spectrum display presented in the SLM Mode can represent the frequency range 25 Hz - 20 kHz or.8 Hz - 20 kHz by toggling the **WIDE [H]** softkey from the **Weight** menu.

As explained earlier in this chapter, the weighting functions for the sound level meter and the frequency analysis functions can be independently selected from the Weight Menu. In most cases, the user will select the frequency analysis weighting from among the four linear weightings with different combinations of highpass and lowpass filters; 1 Hz - 20 kHz, 1 Hz - 10 kHz, 20 Hz - 20 kHz and 2 0H z -10kHz.

There is tremendous power and flexibility built into the frequency analysis capability of the Model 3000+. The remaining chapters of this manual are largely devoted to describing its use as a frequency analyzer. It is recommended that the user read these chapters carefully to fully appreciate the features provided.

In the remainder of this chapter, we seek only to provide sufficient explanation of the frequency analysis function to permit the user to properly calibrate the instrument for sound level measurements.

In the default setup of the Model 3000+ as delivered from the factory, at the completion of the bootup sequence, the frequency analysis function is configured as follows:

Microphone input	Channel 1
Analog Filtering	20 Hz highpass and 20 kHz lowpass filters
Digital Filtering	1/3 Octave Bandwidths
Averaging Type	Exponential
Averaging Time	1/8 second (corresponds to SLM Fast)
Units	SPL (sound pressure level)

Calibration

The subject of calibration of the Model 3000+ for sound and vibration measurements is dealt with in detail in Chapter 9 of this manual. However, for the purpose of quickly calibrating the Model 3000+ for a sound level measurement, the following description is provided.

Sound Level Calibrator

It is the usual practice in the acoustics field to utilize a sound level calibrator to perform the calibration of a sound level meter. This device fits over the grid cap of the microphone and exposes the microphone to a known sound pressure level at a fixed frequency. Although the calibration may be performed using calibrators providing various combinations of signal level and frequency, the sound level meter standards require that the manufacturer recommend a specific reference level and reference frequency for calibration. Larson Davis recommends calibration be done using a reference level of 94.0 dB at a frequency of 1 kHz, which can be provided by the Larson Davis Model CAL200 or CAL250 Sound Level Calibrators.

The CAL200 begins producing the calibrated sound level upon pressing the button on the side. When equipped with fully charged batteries the sound will remain on for a period of at least one minute. As the batteries become weaker, the calibrator will shut off sooner than one minute, but the level and frequency will remain correct during the time it is operating.

Calibration Procedure

We will assume that the Model 3000+ has just been turned on and it has booted up to the default setup as delivered from the factory. The frequency analysis function will be configured for 1/3 octave analysis, with flat weighting over the frequency range 20 Hz - 20 kHz. This is indicated on the third line down on the right of the screen, at the right end of the line. If the weighting is different due to a boot-up modification, access the Weighting Menu and press **20-20k [D]** to select flat weighting with those values of analog lowpass and highpass filters at the input of the frequency analysis function. If the microphone bias voltage is to be different than the default 200 volts, change that as described at the beginning of this chapter.

Access the sound level meter function by pressing **SLM**, place the calibrator over the microphone, switch it ON, and press the **RUN/STOP** key to begin a measurement. Use the upward and downward vertical arrow keys to adjust the range until the sound level trace on the screen falls within 20 dB of the full scale. The spectrum displayed on the left of the screen should indicate a dominant peak at the frequency of the sound produced by the calibrator. Press **RUN/STOP** a second time to stop the measurement.

Using the left and right horizontal arrow keys, move the dotted cursor until it is located over the frequency band corresponding to the predominant frequency peak in the spectrum. On the right side of the screen, fifth line down from the top, is displayed the frequency of this cursor position. This value should agree with the frequency specification of the calibrator.

On the seventh line down from the top is displayed the level corresponding to the cursor position of the spectrum display, in the format “d=XXX.X”. The letter “d” indicates that this value is for the dotted cursor position. If the displayed level value is equal to the sound pressure level specification of the calibrator at that frequency, then the 3000+ is properly calibrated and no further adjustments are required.

In the default setup of the SLM function, the displayed sound pressure level and 1/3 octave spectrum is unweighted. The measured sound pressure level displayed on the lower right of the screen will most likely equal the level corresponding to the calibrator frequency of spectrum display. If A-weighting is selected there will be differences in the A-weighted level versus the spectrum level. The preferred method of calibration is performed using the spectrum level measured at the fundamental frequency of the calibrator.

If the displayed level is different than that specified for the calibrator, press the following key sequence:

SYSTEM, **UNITS[F]** and **level [H]**

which will produce the message “Enter Level + XXX.X” on the upper right of the screen. The XXX.X represents the level presently being displayed for the frequency band covered by the dotted cursor. The flashing cursor beneath the first digit prompts the user to enter the numerical value of the sound pressure level produced by the calibrator using the numeric keypad on the front of the 3000+ and press **EXIT**. Press **EXIT** twice more to return to the SLM Menu.

Check the calibration by switching the calibrator ON once more, setting the instrument to the recommended reference range setting of 120 dB full scale, making another measurement, and verifying that the sound pressure level read for the cursor position is the same as the level produced by the calibrator. When the original calibration was far from the proper setting, a second calibration may be required to get complete agreement within 0.1 dB.

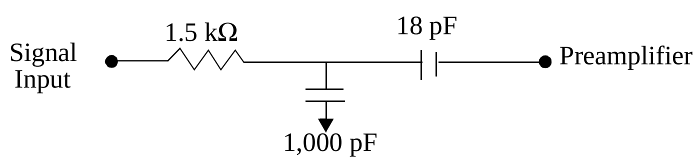
Effect of Microphone Extension Cable

Following calibration, insertion of a Larson Davis Model EXAXXX microphone extension cable having a length of less than 500 feet (150 m) between the instrument and the microphone preamplifier will not require a correction to the calibration.

Noise Floor Measurement and Proximity Message

When measuring low level sound pressure levels, we wish to be certain that the indicated value is not affected by the noise floor of the instrument. More specifically, we would like to know when the measured sound pressure level is within 5 dB of the noise floor of the instrument.

To do this, we first measure the noise floor of the instrument. This should be done by replacing the microphone to be used by a dummy microphone having the following electrical impedance:.



Input adapters are available from Larson Davis Laboratories. If an input adapter dummy microphone is not at hand, it is possible to simulate the same condition by turning off the microphone bias voltage. When using a Larson Davis input adapter (ADP005) a short circuit is made at its input before making the measurement. Set the 3000+ to the weighting which is to be used for the measurement; A, C or one of the linear weightings, select the Slow detector and make a measurement. Adjust the gain until the measured value is within 20 dB of the full scale. When the indicated value of SPL is stable, press **RUN/STOP** to stop the measurement. To modify the noise floor, press the following sequence of keys: **SYSTEM, UNITS [F], Noise.F [E].** This will produce the message “New NOISEFLOOR? was X.XX” where X.XX is the previously measured and stored value of noise floor for the selected weighting. Press **YES [A]** to replace the old value with the one just measured. Otherwise, press **NO [C]** to keep the previous value of noise floor.

Now, when performing a sound level measurement, whenever the measured sound pressure level falls to within 5 dB of the noise floor, the message "+N" will appear to the lower right of the digital display of the sound pressure level on the lower right of the screen.

Since there are six different weightings possible; A, C and four different combinations of highpass/lowpass filters, it is recommended that the user measure and store a noisefloor for each of these such that the proper noisefloor proximity indication will occur no matter which has been selected at the time of the measurement.

When the units are reset from the Resets Menu by pressing **R.UNITS [B]** as described in Chapter 4, the noise floor values for all the weightings will be reset to zero.

Environmental Effects on SLM Measurements

Magnetic Field

The maximum noise floor of the Model 3000+ equipped with a Model 2541 high sensitivity microphone when exposed to a 60 Hz magnetic field of strength 10 A/m^2 (1 Oersted) is as follows:

A-weighting	15 dB SPL
C-weighting	24 dB SPL
Flat weighting (20 Hz - 20 kHz)	25 dB SPL

Temperature

The maximum variation of sound pressure level due to temperature variation over the range -10 to 50° C, referred to the indication at 20° C, is less than ±0.5 dB. At low temperatures, approaching 0° C or below, the response of the LCD display may become very slow. However, the accuracy of the data measured and stored will remain within specifications.

Humidity

The range of humidity over which the complete instrument, including the microphone, is intended to operate continuously is 0 to 95% relative humidity, non-condensing.

The maximum variation of sound pressure level due to humidity variation over the range 30% to 95% relative humidity, non- condensing, referenced to the indication at 65%, is less than ± 0.5 dB.

Temperature and Humidity; Permanent Damage

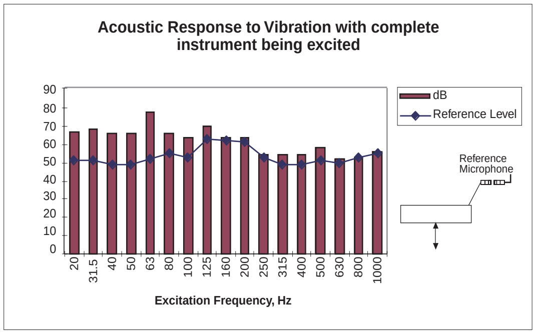
The range of temperature and humidity conditions beyond which permanent damage to the instrument may result:

Temperature:	- 20 to 60° C
Humidity:	0 to 99% relative humidity, non-condensing

Effect of Vibration

Figure 3-29 below presents the broadband sound pressure level measured over the frequency range 20 Hz–20 kHz when the instrument is excited by a sinusoidal vibration of amplitude 1 m/s^2, compared to that measured by a non-vibrating microphone placed near the vibrating microphone. The instrument is mounted on the exciter with the front panel horizontal and the vibration excitation applied in the vertical direction. Two cases are studied; with the microphone aligned horizontally such as would typically be used with a free-field microphone (left) and with the microphone aligned vertically such as would be used with a random incidence microphone (right).

Figure 3-29 ***Complete instrument being excited***



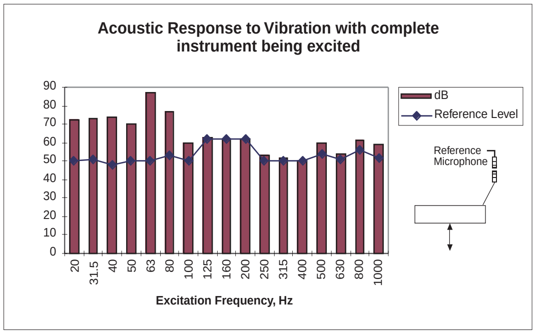
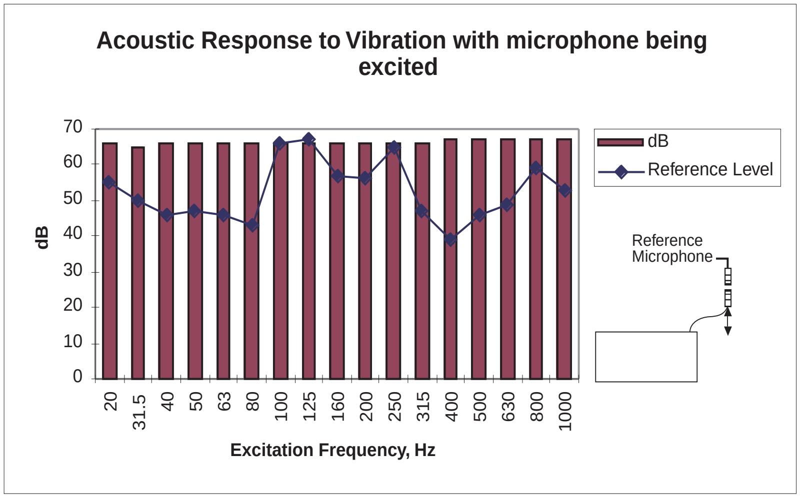


Figure 3-30 presents similar data measured when the microphone and preamplifier only are excited in the vertical direction, as indicated, with the instrument stationary.

Figure 3-30 ***Microphone only being excited***



Sound Pressure Level Measurements; Dual Channel Sound Level Meter with Frequency Analysis (SLM+A) Mode, Two Microphones

The Dual Channel Sound Level Meter with Frequency Analysis Mode provides much of the functionality of the Single Channel Sound Level Meter with Frequency Analysis Mode for both Channels 1 and 2 simultaneously. Due to the additional signal processing demands of the dual channel mode, only 1/1 and 1/3 octave spectra are measured and the highest frequency filter is limited to 10 kHz. This does not affect the ability of the SLM function to meet the Type 1 sound level meter standards.

Setup

From the Main Menu, access the System Menu by pressing **SYSTEM**. Select the SLM+A mode by pressing **SLM+A [B]**. Press the key **#Chanls [A]** until the message “Channel 1 of 2 NORMAL” appears on the right side of the screen, 6th line down. If the location of the right side of the screen, third line down, is not “Dual” (it could be “Input 1” or “Input 2”), press **INPUT [K]**, and **#Inputs [G]** to change it to “Dual”.

The operation in this mode is essentially the same as for the Single Channel Sound Level Meter with Analyzer Mode. However, the frequency weighting is selected individually for each channel; these need not be the same. Press the appropriate channel key, **CH1** or **CH2**, prior to selecting the frequency weighting for that channel, as indicated by the message “Channel 1 of 2 NORMAL” or “Channel 2 of 2 NORMAL on the right side of the screen.

Although the frequency weighting can be selected independently for each channel, the same weighting will apply to both the SLM and the frequency analysis function. The frequency weighting is selected using the lower set of softkeys in the Weighting Menu. The lower limit of the frequency display will be set to 6.3 Hz when the selection has a 1 Hz lower limiting frequency and 25 Hz when it has a 20 Hz lower limiting frequency or when the A or C weighting have been selected. The lower limit of the frequency display can be changed to 0.8 Hz by pressing **WIDE [H]**. Pressing this softkey again will return the lower limit of the frequency display to the previous setting.

Each channel should be individually calibrated, pressing **CH1** or **CH2** to access that channel prior to calibrating. Also, the noise floor measurement and proximity message should be set up individually for each channel.

Sound Pressure Level Measurement; Dual Channel Sound Level Meter with Frequency Analysis (SLM+A), Single Microphone

In this mode, the signal from a single microphone is branched to both SLM+A measurement channels, permitting measurements of the same signal to be performed using different frequency weightings. The two channels are setup as described in the section above. Then, before initiating the analysis, from the Input Menu press **# Inputs [G]** which will change the parameter displayed on the right of the screen, 3rd line down, to either “Input 1” or “Input 2”. Select the input channel to which the measurement microphone is connected by pressing either **CH1** or **CH2** and **EXIT**. Perform the measurement as usual, using the hardkeys **CH1** and **CH2** to select which of the two (SLM+A) analysis is being displayed. If different frequency weightings have been used, the corresponding weighting displayed on the lower right of the screen near the digital SPL readout will change as well.

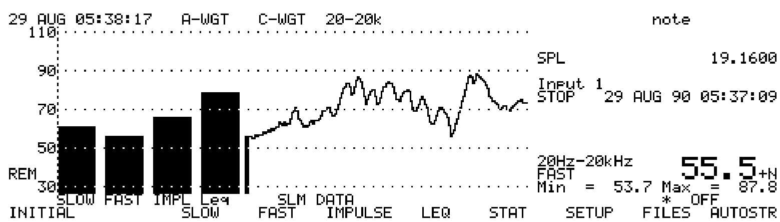
Sound Pressure Level Measurements using the Wide Dynamic Range Sound Level Meter (WDR SLM) function

In this mode, the two A/D convertors usually used for each of the two microphone inputs are used together, offset, to measure a single channel, providing a primary indicator range in excess of 80 dB. This means that the Type 1 specifications corresponding to ANSI S1.4 1984 and IEC 60651 and IEC 60804 are met over the entire 80 dB dynamic range of the instrument. Frequency analysis is not provided in this mode of operation.

Accessing the WRD SLM Menu

To access the WDR SLM menu, as shown in Figure 3-31, press the hardkey **SLM**.

Figure 3-3 1 ***WDR SLM Menu***



In this mode, the following parameters are measured simultaneously:

- Sound Pressure Level (L_p) using Slow Weighting
- Maximum and Minimum values of Slow L_p since last reset
- Sound Pressure Level (L_p) using Fast Weighting
- Maximum and Minimum values of Fast L_p since last reset
- Sound Pressure Level (L_p) using Impulse Weighting
- Maximum and Minimum values of Impulse L_p since last reset
- Equivalent-continuous Sound Pressure Level (L_{eq})
- Single Event Level (SEL)
- Peak Sound Pressure Level (L_{peak})

Slow, Fast, and Impulse sound pressure levels are indicated by the bar graphs on the left of the display. Digital values of the measured parameters are displayed on the lower right of the display. Select the parameters to be displayed using the following keys:

Softkeys	Softkey Functions
SLOW [I]	L_p Slow, with Max and Min values
FAST [J]	L_p FAST, with Max and Min values
IMPULSE [K]	Impulse, with Max and Min values
LEQ [L]	L_{eq}, SEL, L_{peak}

Note that the cursor is inactive in the WDR SLM function, as indicated by the message "*OFF" on the lower right side of the screen.

Selecting the Microphone Input and the Bias Voltage

The active microphone input is shown on the right of the display as "Input 1" or "Input 2". The default setting of the instrument when delivered from the factory is "Input 1" corresponding to the connector on the top panel nearest the edge. To select the microphone input press **CH1** or **CH2**.

To set the microphone bias voltage, press the key sequence **SYSTEM**, **INPUT [K]**, and one of the following: **0V [A]**, **28V [B]**, or **200V [C]**. The factory default bias voltage is 200 volts. Press **EXIT** twice to return to the WDR SLM menu.

Selecting the Frequency Weighting

The frequency weighting is selected as follows:

Softkeys	Softkey Functions
A-WGT [A]	A-Weighting
C-WGT [B]	C-Weighting
20 - 20k [C]	20 Hz highpass/20 kHz lowpass

The active frequency weighting is indicated on the lower right of the screen.

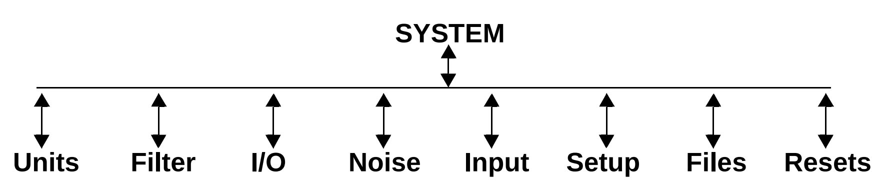
CHAPTER

4 3000+ Instrument Setup Via The System Menu

At any time, the setup and operational status of the analyzer are indicated on the display as described in Chapter 1, LCD Display Parameter Presentation Format. It is possible that the 3000+ will bootup to exactly the setup desired, but that is highly unlikely unless the bootup configuration has been modified by the user to match his requirements exactly. Usually the user will move immediately to the System Menu, from which he may recall and possibly modify a setup previously created and stored, or he may proceed to create a complete new setup. He will then exit to one of the Analysis Menus for instrument operation. If one or more particular instrument setups are used frequently, the user may save each of these setups or even replace the bootup setup with one of them.

System Menu

These Menus are accessed directly from the System Menu.

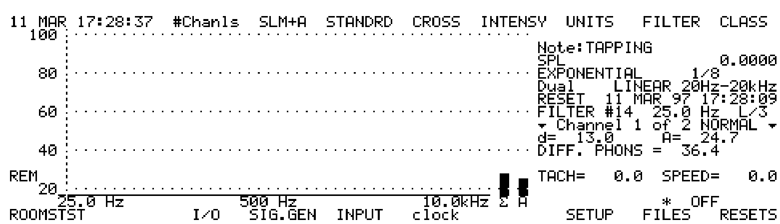


Menu Name	Menu Function
System	Selection of number of input channels and path between the System submenus and the Analysis Menus.
Units	Select units, define and store user-defined units, perform calibration
Filter	Selection of Filter type and parameters
I/O	Setup of computer I/O interface
Noise	Setup of Noise generator
Input	Setup of Input modules
Setup	Storage and recall of user-created instrument Setups
Files	Creation, selection and directory of stored data Files
Resets	Menu for Resets

Accessing the System Menu

The System Menu, figure 4-1, is accessed from any Softkey Menu by pressing **SYSTEM.**

Figure 4-1 ***System Menu***



The user might choose to examine the instrument setup in detail, as indicated on the right of the display, and change item-by-item those parameters which are not as desired for his measurement, but with some practice it is more efficient to simply create a totally new setup.

We will first concern ourselves with setting up those parameters related to the measurement itself, and address the I/O, Noise, Clock, Setup, Files and Reset Menus later in this Chapter.

Selection of Analysis Type

The Analysis mode is selected by pressing one of the following softkeys:

Softkeys	Softkey Functions
STAND [C]	for Standard Analysis
CROSS [D]	for Cross Analysis
INTENSY [E]	for Intensity Analysis

The softkey **SLM+A [B]** will place the 3000+ in the Sound Level Meter with Analysis Mode permitting it to perform sound pressure level and frequency spectral measurements simultaneously as described in Chapter 3.

Standard Mode

In the Standard Analysis Modes, six different forms of spectra (Normal, Leq, Min, Max, SEL and MaxSpec) are calculated for each channel, regardless of filter type. The distinction between these is explained in Chapter 6. The softkey **#Chanls [A]** toggles the Standard Mode between single and dual channel analysis, as indicated by the message "Channel 1 of 1" or "Channel 1 or 2" on the right of the display, 6th line down.

For single channel Standard Analysis, the user may select to use either of the two input connectors by using the hardkeys **CH1** and **CH2.** The connector closest to the right side of the instrument represents channel 1.

For dual channel Standard Analysis, the two input connectors are used simultaneously to represent channels 1 and 2, with the one closest to the right side of the instrument being channel 1. The results of this measurement are comparable to having a single channel analyzer connected to each of the inputs, since no cross channel parameters are measured.

Cross Mode

In the Cross Mode of operation with FFT filtering, the following data are measured and displayed: Autospectra, Cross Spectra, Auto Correlation, Cross Correlation, Transfer Functions (3 forms), Impulse Response, Coherence, Coherent

Output Power, Time Waveforms and Weighted Time Waveforms.

In the Cross Mode of operation with Octave filtering, the following data are measured and displayed: Autospectra, Cross Spectra, Transfer Functions (three forms), Coherence and Coherent Output Power.

Intensity Mode

Used in the Intensity Analysis mode with a Larson Davis Acoustic Intensity Probe, the Model 3000+ will measure and display Acoustic Intensity, Particle Velocity, Sound Pressure Level, and Quality (Intensity/ Pressure). The format of the data presentation in the frequency domain will be the same as the filter type selected (Octave or FFT).

When the Analysis Mode is selected by pressing one of these three softkeys, System Menu will remain active on the display. However, when exiting from the System Menu, the system will proceed to the analysis Menu corresponding to the selected analysis mode.

Frequency Range Considerations

Octave Frequency Analysis

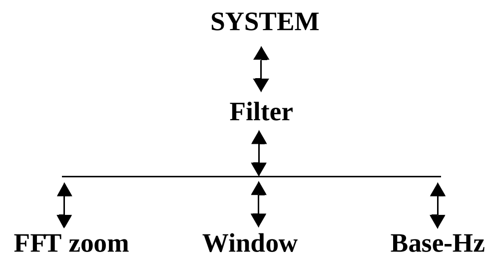
The Model 3000+ digital filters provide 1/1 and 1/3 octave real-time analysis over the frequency range 1 Hz-20kHz for both single and dual channel operation. When the highpass filters at the inputs of both channels have been selected to be 20 Hz, the minimum frequency displayed will be 25 Hz.

FFT Frequency Analysis

The Model 3000+ can perform real-time FFT analysis using a baseband frequency range to 2.5 kHz, 5 kHz, 10 kHz or 20 kHz in single or dual channel mode. In dual mode with a 20 kHz upper frequency and Hanning weighting, the processing is done with a 30% overlap. The lower frequency limit of the analysis will be 1 Hz or 20 Hz depending upon the choice of highpass filters at the inputs.

Selection of Filter Type

The following diagram shows the Menus accessed directly from the Filter Menu.



The functions performed within each of these Menus are as follows:

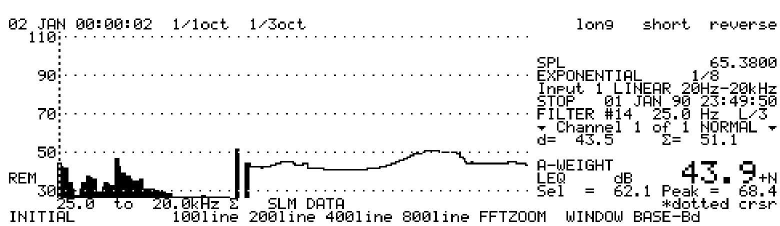
Menu Name	Menu Function
Filter	1. Select and configure octave-type Filters
	2. Select FFT Filtering and number of lines
FFTzoom	Select FFT zoom factor
Window	Select FFT time weighting Window
Base-Hz	Select FFT Baseband full scale frequency

Note that all the submenus refer to FFT analysis. All actions required to select and configure octave-type filters are done from the Filter Menu itself.

Accessing the Filter Menu

The Filter Menu, shown in figure 4-2, is accessed from the System Menu by pressing **FILTER [G]**.

Figure 4- 2 ***Filter Menu***



The softkeys along the top of the display apply to octave and fractional octave filters and those along the bottom apply to FFT.

Only one of these two filter types can be active at one time.

Selection of Octave and Fractional Octave Filters

Select the filter bandwidth by pressing one of the following:

Softkeys	Softkey Functions
1/1 oct [A]	for full octave bandwidth
1/3 oct [B]	for one-third octave bandwidth

The effective bandwidth for the 1/1 octave filters is $0.7071*f_c$, where f_c is the filter center frequency. The effective bandwidth for the 1/3 octave filters is $0.2316*f_c$. The maximum permissible input voltage is 10 Vrms, corresponding to the maximum permissible input voltage to the instrument; with the range selected appropriately to avoid overload. External ambient sound fields have no effect on the performance of the digital filters.

In North America, the applicable standard for this type of filter is ANSI S1.11-1986 Specification for Octave-Band and Fractional-Octave-Band Analog and Digital Filters. In this standard, filters are specified by both a Type number and a Sub-Type letter as shown in the following tables:

Table 4-1 **ANSI S1.11-1986 Tables**

White Noise Bandwidth Error, millibel	Type Number
≤ 10	0
≤ 25	1
≤ 41 * depends on passband ripple	2 or 3*

Criteria for selecting Type Number

Composite Bandwidth Error, millibel	Sub-Type letter
≤ 13	AA

Composite Bandwidth Error, millibel	Sub-Type letter
≤ 25	A
≤ 50	B
≤ 100	C
> 100	D

Criteria for selecting Sub-Type letter

In terms of the IEC (International Electrotechnical Commission), the most recent standard governing octave filters is IEC 61260-1994. The Technical Committee No. 29 is presently working on a revision to the document.

There are three different octave-type digital filter algorithms available in the Model 3000+, represented by the softkeys

LONG [F], SHORT [G], and **REVERSE [H]** on the right of the upper row.

With regard to the ANSI standard, the Long filter satisfies the ANSI S1.11-1986 requirements for Type 0-AA, the highest classification possible under that standard. The Short filter satisfies the requirements for Type 1-D. For measurements which require a particularly fast filter time response such as measuring gunshots, the Short filter may be preferable, although the slope of the filter skirts is less than that of the Long filter, so the filter resolution is not as fine. In the previous standard ANSI-S1.11-1966 (R-1976), which was superseded by the 1986 version, the highest classification of 1/3 octave filters was Class III. For those whose measurement requirements might still be governed by the old standard, note that all these 1/3 octave filters of the Model 3000+ exceed the requirements of Type III.

Before the advent of digital filters, many of the commercially available 1/3 octave analog filters were based on a 6-pole design. In instances where it is desired that the result of the measurement match as closely as possible the results which would have been obtained using one of these older analog filters, the Short filter is recommended.

The Long and Short digital filter algorithms are designed to provide a fast rise time and slow decay time. For the measurement of rapid time decays, a reversed filter algorithm has been implemented which provides a slower rise time but a faster decay time. The Reverse filter, selected by pressing **reverse [H]**, provides a decay time approximately ten times faster than that of the Short filter.

With regard to IEC 61260-1994, all these filters satisfy the requirements.

The filter algorithm is selected by pressing either **LONG [F], SHORT [G]** or **REVERSE [H]**. When octave and fractional octave filters are active this is indicated on the right side of the display, right end of the fifth line down, in the format of a letter (L for Long, S for Short or R for Reverse) followed by the symbol / and a number (1or 3) which represents the fraction of an octave used for the bandwidth). While in the Filter Menu, the bandwidth and the filter type may be changed independently by simply pressing the appropriate softkey.

Return to the System Menu by pressing **EXIT**.

Selection of FFT Filtering

There are three parameters which must be defined to perform a baseband FFT analysis; the number of lines, the time weighting window and the full scale frequency.

Selection of Number of Lines

Select the FFT analysis mode and the number of lines by pressing one of the following softkeys:

Lines	Softkeys
100 line	[I]
200 line	[J]
400 line	[K]
800 line	[L]

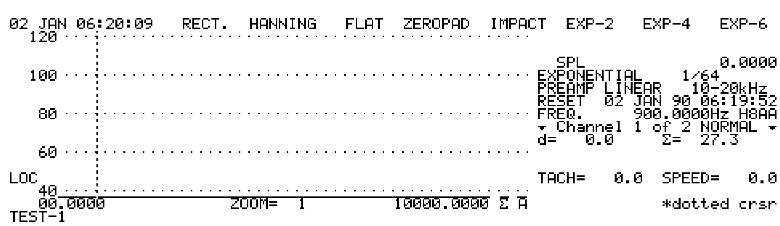
The last four characters in the fifth row down on the right side of the display will indicate the state of the FFT filter setup (H8AA, for example). The first character indicates the type of time weighting window which is active (R, H, F, Z, I or E) as explained in the following section. The second character indicates the multiple of 100 lines which has been selected (1,2,4 or 8). The characters “AA” appearing at the end of the field indicate that analog anti-aliasing filters are present before the A/D converter in the analog portion of the instrument.

*Because of the large number of parameters calculated in the Standard Analysis configurations (Normal, Max, Min, Leq, etc.), the 800 line resolution cannot be used in the dual channel Standard Analysis (STAND 2) mode. There is no restriction for the single Standard, Cross or Intensity modes.

Selection of Time Weighting Window

If the analyzer had been set to FFT analysis earlier in the same measurement session, the time weighting window will be the same as set at that time. Otherwise, it will be as set by the internal software when the unit is booted up. The default setup for the 3000+ as delivered from the factory selects the Hanning window. Because many users utilize the same time weighting window for most of their FFT analysis, particularly the Hanning window, it may not be necessary to modify the window selection when creating an FFT setup. The Time Weighting Window Menu, shown in figure 4-3, is accessed from the Filter Menu by pressing **WINDOW [N].**

Figure 4- 3***FFT Window Menu***



Select the desired time weighting window by pressing one of the following:

<u>Softkeys</u>	<u>Softkey Functions</u>
RECT. [A]	for Rectangular Weighting on all channels
HANNING [B]	for Hanning Weighting on all channels
FLAT [C]	for Flat Top Weighting on all channels
ZEROPAD [D]	for Zero Pad with/without Bow Tie Correction on all channels
IMPACT [E]	for Impact Weighting on channel 1 Rectangular Weighting on channel 2

Softkeys	Softkey Functions
EXP-2 [F]	Impact Weighting on channel 1 Exp-2 Weighting on channel 2
EXP-4 [G]	Impact Weighting on channel 1 Exp-4 Weighting on channel 2
EXP-6 [H]	Impact Weighting on channel 1 Exp-6 Weighting on channel 2

The active time weighting window is indicated on the right of the display (fifth line down, fourth character from the right); R, H, F, Z, I, or E corresponding to the first letter of one of the above choices.

For the analysis of steady signals, most users will select either the Hanning or the Flat Top window. The Hanning gives better frequency resolution while the Flat Top gives better amplitude accuracy in the passband. Rectangular weighting provides the finest frequency resolution, but it is often accompanied by excessive leakage of energy to the neighboring sidebands.

The Impact and Exponential Weighting windows are generally used for modal analysis applications. The Impact Weighting is used on the channel where an impact excitation is applied to a structure, typically using an instrumented hammer providing an analog signal of force versus time. The Impact Weighting window consists of a 1/2 cosine curve rising from zero to unity over eight samples, followed by a horizontal section of length thirty-two samples at unity height, followed by another 1/2 cosine dropping from unity to zero over eight samples. The user selects the trigger delay of the FFT analysis such that the actual force waveform occurs totally within the window, and signals outside the window are reduced to zero digital values.

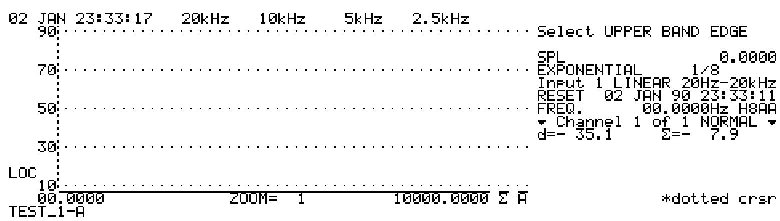
Exponential Weighting is used on the channel where the response of the structure to the impact excitation is being measured. The term Exp-N refers to an exponential weighting window where the attenuation at the end of the time window is N*10 dB with respect to the unity attenuation at the beginning of the time window. By forcing the response amplitude to near zero at the end of the response time window, the effect of leakage on the measurement of the frequency response function is minimized. However, this also has the effect of adding artificial damping to the measured results which must be taken into account when the analytical results of the modal analysis are used to estimate structural damping.

After selecting a time weighting window, the instrument will return to the Filter Menu.

Selection of Baseband Full Scale Frequency (Base-Bd)

The FFT analysis is begun with a baseband analysis, which means that the frequency range of the analysis will extend from DC (0 Hz) to a selected full scale frequency value. The Full Scale Frequency Menu, shown in figure 4-4, is accessed from the Filter Menu by pressing **BASE-Bd [O]**.

Figure 4- 4***Base Band Menu***



The message “Select UPPER BAND EDGE” will appear on the upper right of the display. Make a selection by pressing one of the following softkeys:

20kHz [A]
10kHz [B]
5kHz [C]
2.5kHz [D]

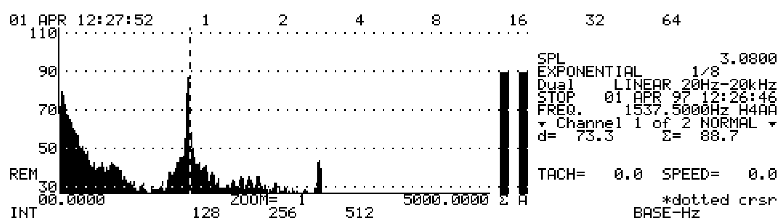
FFT Zoom Analysis to Increase Frequency Resolution

When the FFT analysis is first selected, the instrument is setup to perform a baseband analysis (zoom =1) which means that the analysis range will extend from DC (0 Hz) to the selected full-scale frequency. The frequency resolution of each filter will be the upper frequency divided by the number of lines used for the FFT analysis. For example, using 800 lines and a full scale of 20 kHz, the frequency resolution will be 25 Hz (20,000/800 lines). As the cursor is moved across the spectrum display, from line-to-line the frequency will change in steps of 25 Hz. If the full-scale frequency is reduced to 5 kHz, the frequency resolution will be increased; that is, the frequency increments from band-to-band will be decreased to 6.25 Hz (5000/800). Thus, reducing the full scale frequency will increase the resolution. However, unless the spectral components of interest are at low frequencies, the range of choices for the full scale frequency will be limited as a means to improve resolution.

The frequency resolution of the FFT analysis can be increased substantially by utilizing the zoom analysis mode. Essentially, this means that the number of lines selected for the analysis are used over a narrower frequency range than originally chosen for the baseband analysis.

Access the Zoom Menu, figure 4-5, from the Filter Menu by pressing **FFTZOOM [M]**.

Figure 4-5 ***Zoom Menu***



The zoom multiplier is selected from 2 to 512 (binary sequence) by pressing the key above or below the desired number. Selecting 1 produces a baseband analysis. The total frequency range covered by the zoom analysis will be equal to the full-scale frequency of the baseband analysis divided by the zoom multiplier. For example, when a zoom multiplier of 16 is used after a 0 - 20 kHz baseband analysis, the total width of the zoom analysis will be 1250 Hz (20 kHz/16). Using 800 lines, the frequency resolution would then be 1.56 Hz per band (1250/800).

The cursor position at the time the zoom analysis is initiated defines the center of the zoom frequency range (the use of the cursor is explained in Chapter 7). Continuing the above example, if the cursor had been placed at 1 kHz before the zoom, 1 kHz will be the center of the new 12 5 0Hz wide frequency range. Therefore, the frequency scale seen on the display will be from 375 Hz to 1,625 Hz (1250 Hz/2 =

625 Hz; 1 kHz-625 Hz = 375 Hz; 1 kHz + 625 Hz = 1,625 Hz). Had the cursor been located at 5 kHz, the zoom analysis would cover the range from 4,375 Hz to 5,625 Hz. During the zoom analysis, the cursor will move to the center of the screen, still representing the same frequency value, and the display will show the total frequency range of the zoom analysis.

If the cursor position before the zoom analysis is less than 1/2 of the zoom total frequency range (say 500 Hz, in our example), the above calculation indicates that a negative value for the low frequency limit would result (500 - 625 = 125 Hz). In such a case, the zoom center frequency is made equal to 1/2 of the zoom total frequency range (in the example to 625 Hz), producing a zoom frequency range beginning at 0 Hz. Similarly, should the cursor be located very near the upper frequency limit of the baseband range, the zoom center frequency would be adjusted to a lower frequency value if necessary so that the zoom frequency upper limit would never be greater than the baseband upper frequency limit.

The best procedure is to begin with a baseband full-scale frequency no higher than is sufficient to analyze the frequencies of interest. Then, to examine in more detail specific frequency components or sub-sections of the original frequency range, use the zoom analysis capability.

The analysis performed by the zoom FFT depends uniquely upon the upper frequency, time weighting and number of lines used for the baseband analysis and the selected zoom factor. If one performed a baseband analysis, followed by a zoom analysis using a multiplier of 4, then another zoom analysis using a zoom factor of 32, the result would be the same as if the zoom factor of 32 had been used initially after the baseband analysis (assuming the same center frequency were used for both analyses). The zoom analyses do not “build” upon one another.

The time required for the zoom analysis to be performed will always be longer than that required for the original baseband analysis. In fact, there is an inverse relationship between the time required and the bandwidth (zoom multiplier). A zoom analysis using a zoom multiplier of 64 will require a time interval 64 times longer than the baseband analysis upon which it is based. The user must bear this in mind when using very large zoom factors, since the time required until the first spectrum appears on the screen could be a number of minutes, even with the very high speed processor used by the Model 3000+!

Once in the Zoom Menu, move the cursor to the frequency about which the zoom analysis is to be performed, then press the softkey corresponding to the desired zoom factor. The message “ZOOM = XX” displayed below the frequency axis indicates that the active zoom factor is “XX”.

Once in the zoom mode, one can dynamically “pan” the frequency range of the analysis to lower or higher frequencies. This is done by pressing **BASE Hz [O]** and using the horizontal arrow keys to shift the location of the center frequency about which the analysis is being performed.

To exit from the Zoom Menu, press **EXIT**.

When the FFT parameters are set as desired, return to the System Menu by pressing **EXIT**. Once selected, FFT analysis can be operated from any of the Analysis Menus (Standard, Cross or Intensity), but to invoke or modify zoom analysis, it is necessary to return to the Zoom Menu.

Limitation on Zoom Multiplier

When the Model 3000+ is operating in a dual channel mode (Stand 2, Cross or Intensity) using FFT analysis and the full scale frequency has been selected to be 20 kHz, the analysis will not be performed in real-time. In this case, the zoom function is buffered rather than real-time, which will limit the maximum permissible value of zoom multiplier to 32.

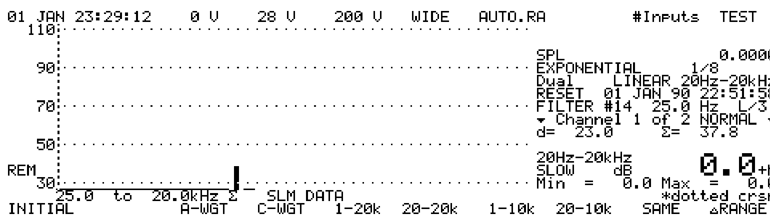
Printing FFT Data in Tabular Format

The use of the printing capabilities of the Model 3000+ is described in Chapter 24. When dealing with FFT spectra measured using many lines of resolution, it may happen that only data within small frequency regions of the total analysis frequency range are of interest, making it desirable that the Amplitude/Frequency data presented in the tabular printout cover only that range. The data presented in the tabular printout represent only those frequency bands shown in the display. Thus, by modifying the horizontal display range as described in Chapter 19, the frequency range of both the tabular printout and the screen display will be reduced accordingly.

Accessing Input Menu

To select the microphone bias voltage and define the input signal path to the analyzer, from the System Menu press **INPUT [K]** which will bring to the screen the Input Menu, shown in figure 4-6.

Figure 4- 6***Input Menu***



Setting the Microphone Bias Voltage

Upon accessing the Input Menu the value of the microphone bias voltage presently active will be displayed on the upper right of the screen for approximately 4 seconds. To change the polarization voltage, press one of the following:

Softkeys	Softkey Functions
O V [A]	Bias voltage OFF, for use with electret or pre-polarized microphones
28 V [B]	28 volt bias voltage active
200 V [C]	200 volt bias voltage active

Branching a Signal from One Input Connector to both Analysis Channels (Dual Channel Analysis Only, Standard or Sound Level Meter)

When performing a dual channel analysis, there may be applications where the user wishes to direct the signal from a single microphone, accelerometer or other input to both analysis channels, since these can be setup with different frequency weightings. Once the measurement parameters have been established for the two analyses, press **#Inputs [G]** and then either **CH1** or **CH2** to select the input connector to which the transducer whose signal is to be measured is connected. The selection will be confirmed by the message "Input 1" or "Input 2" on the right of the screen, 3rd line down.

Setting the Analog Filters for the Frequency Analysis Function

From the Input Menu the user can select to utilize either a broadband weighting filter (A-weight or C-weight) in the signal path, or a pair of highpass/lowpass filters, by pressing one of the following:

Softkeys	Softkey Functions
A-WGT [I]	Inserts an A-Weighting filter in the signal path
C-WGT [J]	Inserts a C-Weighting filter in the signal path
1 -20k [K]	Inserts a 1 Hz highpass filter and a 20 kHz lowpass filter in the signal path
20 -20k [L]	Inserts a 20 Hz highpass filter and a 20 kHz lowpass filter in the signal path
1 -10k [M]	Inserts a 1 Hz highpass filter and a 10 kHz lowpass filter in the signal path
20 -10k [N]	Inserts a 20 Hz highpass filter and a 10 kHz lowpass filter in the signal path

When the Model 3000+ is in a dual channel mode (STAND 2), the analog filters can be selected independently for each input channel. The filters for channel 1 are set by pressing **CH1** prior to making the selection (resulting in the message "Channel 1 of 2..." on the right of the screen) and the filters for channel 2 are set by pressing **CH2** prior to making the selection (resulting in the message "Channel 2 of 2..." on the right of the screen. To have the same choice of filters for both channels, press **SAME [0]**, which will set the filters for the channel not being displayed to be the same as those of the channel being displayed.

Internal Calibration Signal

From the Input Menu, the user can select to insert a 1 kHz square wave signal whose fundamental frequency amplitude is 1 volt through the input for purposes of verification and instrument calibration. Press **TEST [H]** to turn on this signal, which will also produce the message "Internal Calibration On” on the upper right of the screen. Press **TEST [H]** a second time to turn off the calibration signal.

Offsetting Gain Between Channels

The gain of channel 2 with respect to channel 1 is adjusted from the Input Menu by pressing **ΔRANGE [P]** which will assign the horizontal arrow keys the role of adjusting the gain offset as indicated by the message “*Δrange XX” on the lower right of the screen. The XX denotes the offset between channel 2 and channel 1, and this will change as the horizontal arrow keys are pressed. Both positive and negative values of offset are permitted. After the offset has been set, assign the horizontal keys to another function, such as controlling the cursor.

The normal range control will continue to adjust the gain of both channels together in 10 dB steps, but the offset will remain between them as seen by comparing the full scale values of the two channels.

To remove the offset, repeat the same procedure used to set the offset but adjust for a zero value of offset.

Setting the Autorange Aperture

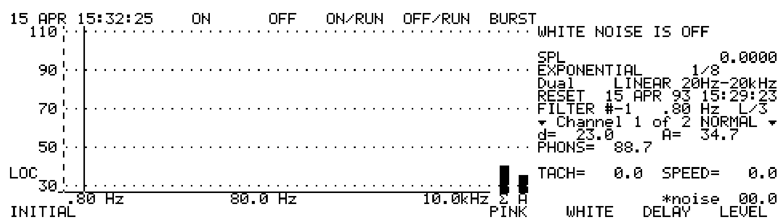
The 3000+ input range settings may be set automatically by the Autorange Function, which is described in detail in Chapter 6. Under autorange control the system seeks to set the input attenuators such that the maximum displayed signal amplitude falls within an amplitude window extending from full scale down to a level equal to the Autorange Aperture, without an overload. The default value of the Autorange Aperture is 20 dB. This may be changed by the user from the Input Menu by pressing **AUTO.RA [E]**, and in response to the prompt on the upper right of the display, typing in a value using the keypad and pressing **EXIT.**

This concludes the portion of this chapter which deals with setup parameters directly related to the measurement process. Pressing **EXIT** will return the system to the Analysis Menu corresponding to the Analysis Mode last selected while in the System Menu.

Operation of the Noise Generator (OPT 10 Required)

The Noise Menu, shown in figure 4-7, is accessed from the System Menu by pressing **NOISE [J].**

Figure 4-7 ***Noise Menu***



The status of the noise generator is indicated by the message on the upper right of the screen, indicating the spectral content of the noise (pink or white) and the operational status (ON, OFF, ON/RUN or OFF/RUN). Pink noise has equal energy content per percentage bandwidth and is usually used with octave bandwidth measurements. White noise has equal energy content per constant bandwidth and is most often used in conjunction with FFT analysis.

Connection

The connector for the noise source output is located on the top panel of the instrument as indicated by the rear panel label. The load impedance should be at leas t 6kΩ.

Selecting Spectral Content

The spectral content of the noise is selected by pressing either **PINK [M]** or **WHITE [N]**.

Selecting Operational Mode

Pressing **ON [A]** or **OFF [B]** will cause the generator to be continually ON or OFF. Pressing **ON/RUN [C]** will engage the noise generator when the analyzer is in the run mode. Conversely, pressing **OFF/RUN [D]** will disengage the noise generator when the analyzer is in the run mode. Changes in the operational status or the spectral content will bring the appropriate message to the upper right of the screen.

A typical application of the OFF/RUN mode is the measurement of reverberation time. A room is filled with acoustic energy and then the energy decay is measured after the sound source is turned off. With the 3000+ configured to perform 1/3 octave analysis using the autostore measurement mode, pressing the **R/S** key initiates RUN which turns off the noise generator and begins the autostorage of spectra during the decay process. During the initial interval of analysis, there will be a finite time before data is available from each of the filters. The lower the frequency of the filter, the longer the interval before the appearance of data from the filter. Therefore, it is convenient to delay the shutoff of the noise generator until data is being produced from all the filters of interest. The user sets this delay time by pressing **DELAY [O]**. Note the message "*delay X.X s" on the lower right of the screen indicating that by pressing the horizontal arrow keys, the user can set the delay time, in seconds, as desired. To release the horizontal arrow keys from controlling the delay time, press **CURSOR**. The use of the 3000+ for the measurement of reverberation time is described in detail in Chapter 21.

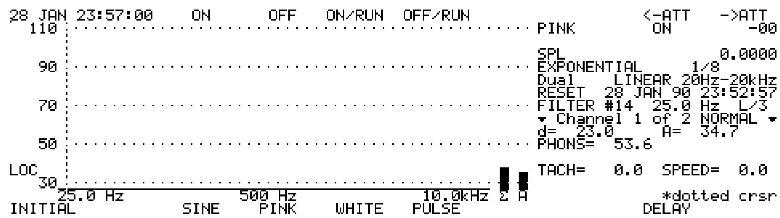
The 3000+ also provides a noise burst mode which generates repetitive noise bursts having a 1 millisecond duration. The initial burst is resequenced for the subsequent bursts, so that the spectral content of each burst is identical. The softkeys **PINK [M]** and **WHITE [N]** are used to select the general spectral shape of the noise burst. The bursts are initiated by pressing **BURST [E]**, which produces the message "PINK NOISE IS BURST" or "WHITE NOISE IS BURST" on the upper right of the screen. The repetition rate of the burst is set by pressing **DELAY [O]** and using the horizontal arrow keys to adjust the rate, in seconds, as indicated by the message "* delay XX.Xs" on the lower right of the screen.

Operation of the Signal Generator (OPT 11 Required)

The OPT 11 Signal Generator provides swept sine (with tracking filter and feedback level control), dual frequency swept sine and a pulse generator in addition to the pink and white noise provided by the OPT 10 Noise Generator. Also, there is an autolevel feature for use with the pink noise to assist in equalizing the noise level in a test room.

The Signal Generator Menu, shown in figure 4-8, is accessed from the System Menu by pressing **SIG.GEN [J]**.

Figure 4-8 ***Signal Generator***



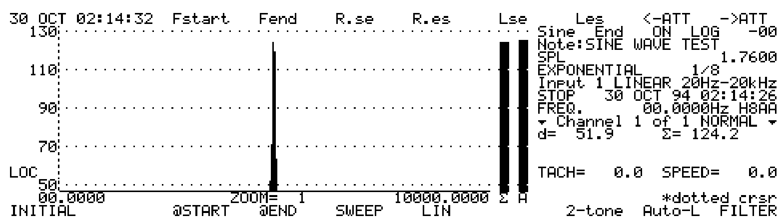
Operational Mode

The operational mode of the signal generator (On, Off, On/Run and Off/Run) is set from this Menu in exactly the same manner as described for the Noise Generator in the preceding section.

Sine Generator, Single Tone

The Sine Generator Menu, shown in figure 4-9, is accessed from the Signal Generator Menu, by pressing **SINE [I]**.

Figure 4-9 ***Sine Generator***



Upon accessing this menu, the first word in the message appearing briefly on the upper right of the screen should be "Sine". If instead it is "2tone" indicating that the dual tone mode is active, press **2-tone [N]** to put it back into the single tone mode. The sine generator can be used with either 1/1, 1/3 octave digital filters or FFT analysis. When learning to use the sine generator function, it is particularly useful to use the sine generator as an input to the analyzer with the FFT analysis mode selected.

The user can define two frequency limits, F_{start} and F_{end}, by pressing the softkeys **Fstart [A]** and **Fend [B]** respectively, using the numeric keypad to enter a value of frequency in Hz, and pressing **EXIT**. Upon pressing either key, the frequency value presently assigned is indicated on the upper right of the screen. The displayed value will change in response to modifications made using the numeric keys.

The output frequency from the signal generator can be manually set to the value of F_{start} by pressing **@START [I]**, and to F_{end} by pressing **@END [J]**. A frequency sweep is begun by pressing **SWEEP [K]**. The frequency of the output signal will then sweep from F_{start} to F_{end}, producing a constant user-defined voltage output (L_{se}) and then from F_{end} back to F_{start} producing another constant user-defined voltage output (L_{es}). The frequency will continue to sweep back and forth between F_{start} and F_{end} producing the programmed levels until either the sweep is paused by pressing **SWEEP [K]**, or the frequency is set manually to either F_{start} or F_{end} by pressing **@START [I]** or **@END [J]**. When the sweep has been paused, pressing **SWEEP [K]** again will result in the continuation of the sweep from the state it was in when the pause occurred.

The rate of the sweep can be either logarithmic or linear in the frequency domain. Whenever one of the parameters of the sine generator is changed, the frequency state of the output signal is indicated on the upper right of the screen for approximately ten seconds. This message will indicate whether the frequency is fixed at the “Start” frequency, fixed at the “End” frequency or in a “Sweep” mode. It will also indicate whether the generator output is “ON” or “OFF”, and whether the selected sweep rate is “LOG” or linear, in which case the sweep rate may be defined either by the “time” of the sweep or by the “count” of the number of cycles per sweep.

The sweep mode is toggled between logarithmic and linear modes by repeated presses of the softkey **[L]**. The label of that softkey at anytime will be either **LOG** or **LIN**, indicating that pressing the softkey will change it to the mode corresponding to the label. The active mode is briefly indicated on the upper right of the screen when most of the softkeys are pressed.

The F_{start}-to-F_{end} sweep rate is set using the softkey **[C]** and the F_{end}-to-F_{start} sweep rate is set using the softkey [**D]**. The labels on these keys will depend upon the selected sweep mode as described in the following four paragraphs.

When the sweep mode is “LOG”, these keys will be labeled **R.se [C]** and **R.es [D]** (R for “rate”). Pressing either of these keys will produce a message on the upper right of the screen indicating the present value and prompting the user to make a modification via the numeric keypad, if desired, and press **ENTER**. The units in the logarithmic sweep mode are decade/second.

When the sweep mode is linear and based on time, these keys will be labeled **T.se [C]** and **T.es [D]**, and the units are seconds per sweep. Press either key to display the present value and to modify the value, as described in the preceding paragraph.

When the sweep mode is linear and based on cycle count, these keys will be labeled **N.se [C]** and **N.es [D]**, and the units will be the number of cycles per sweep. Press either key to display the present value and to modify the value, as described previously.

When in the linear sweep mode, repeatedly pressing the softkey **[M]** will toggle between time and count, as can be seen by the changing of the label between **TIME** and **COUNT**.

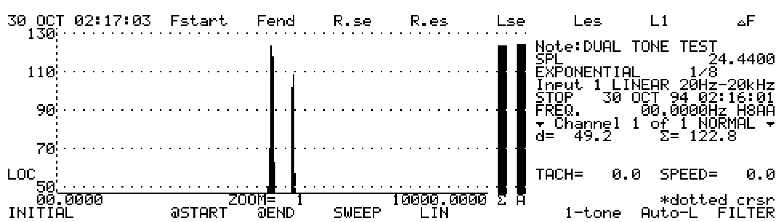
The output level corresponding to the F_{start}-to-F_{end} portion of the sweep is set using the softkey **Lse [E]**, and that for the F_{end}-to-F_{start} portion of the sweep using the softkey **Les [F]**. Upon first pressing either key, the presently assigned value is displayed. Either press **ENTER** to keep the same value or modify the value using the numeric keypad and then press **ENTER**. The value can range from a maximum of .9999 to a minimum of .0000. The maximum value will produce a voltage output of approximately 3 Vrms, and other values will produce a proportionally lower output voltage. The softkey **<-ATT [G]** will attenuate the output level by 20 dB each time it is pressed, up to a maximum attenuation of -60 dB. With attenuation in effect, the softkey **->ATT [H]** will reduce the attenuation by 20 dB each time it is pressed, until there is no attenuation in effect. A message on the upper right of the screen will indicate the status of the attenuation whenever either of these keys is pressed.

To obtain the optimum performance from the digital-to-analog converter (DAC), try to keep the level value as large as possible by using the 20 dB attenuator steps rather than continue reducing the level. For example, once the level is below .0999, the same output voltage can be obtained by increasing the attenuation by 20 dB and setting the level back to .9999.

Sine Generator, Dual Tone

With the sine generator set for a single tone, activate the dual tone mode by pressing **2-tone [N]**. The first word in the message appearing briefly on the upper right of the screen will be “2tone”. Repeated presses of this key toggles the status between single and dual tone. The Dual Tone Generator Menu is shown in figure 4-10.

Figure 4-1 0***Dual tone Generator***



The frequency of the second tone will be greater than the first tone by a fixed number of cycles. This is set by pressing **ΔF [H]**. The message on the upper right will indicate the presently assigned value of the frequency difference. Either press **ENTER** to keep the same value or modify the value using the numeric keypad and press **ENTER**.

The user selects the relative amplitude of the first tone (L1) by pressing **L1 [G]**, entering a value between 0 and 1 using the numeric keypad, and pressing **ENTER**. The sum of the relative amplitudes of the two tones are set equal to one, so the relative amplitude of the second tone will be given by the relationship L2 = (1 - L1). Setting L1 to 0.5 will result in both tones having equal amplitudes. Once setup, both tones can be swept in the same manner as a single tone.

Autolevel Control; Sine Generator

This feature is used with the dual channel Model 3000+ to perform calibration of accelerometers and microphones. For accelerometer testing, the output of the sine generator is used to drive a shaker, upon which are mounted both a reference accelerometer and a test accelerometer. The signal from the reference accelerometer is connected to the Channel 1 input and the signal from the accelerometer under test is connected to the Channel 2 input. The instrument is set to the STAND 2 Mode with FFT filtering active and the Max level selected for display.

Before beginning the test, the output frequency is set to F_{start} by pressing **@START [I]**, at which time the level will correspond to that set for Lse. The autolevel mode is actuated by pressing **Auto-L [O]** which will produce a message on the upper right of the screen whose first word is either "Auto" if in the single tone mode, or "Auto2" if in the dual tone mode. The test is begun by pressing **SWEEP [K]**. With the autolevel mode active, Channel 1 is used as a feedback channel, with the output of the sine generator being modified as necessary to maintain the level measured in Channel 1 constant. Thus, should the frequency of the signal approach a resonant frequency of the amplifier/shaker system, the detection of an increase in the measured level would result in a decrease of the output level in order to maintain a constant acceleration level of the shaker. A digital tracking filter in the feedback loop assures that only the feedback level at the signal generator frequency is being compared to the programmed output level for shaker control. Unless the system under test is extremely non-linear, the accelerometer signals measured in both channels will have the same frequency as the generator. By selecting the Max display mode, as the sweep progresses the displayed curves will represent the measured level as a function of frequency. If properly implemented, at the conclusion of the test the trace displayed for Channel 1 should be flat as a result of the autolevel control and the trace for Channel 2 will represent the frequency response of the test accelerometer.

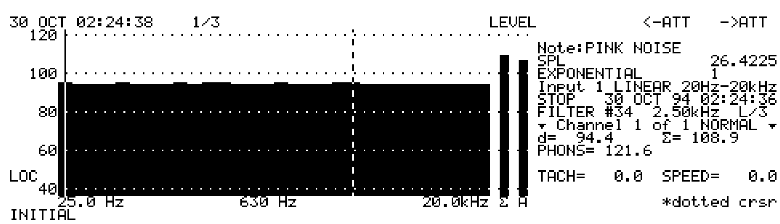
Like any circuit, the feedback loop has a finite response time, which means that if the level of the reference channel changes too rapidly, the modification of the output level may not be fast enough to maintain the reference level within desired limits. This is largely controlled by the sweep rate. For instance, when the frequency is sweeping through a sharp resonance, if the trace of the reference channel shows an increase during the test, the sweep rate should be reduced. Another source of instability in the feedback loop could be the existence of a time delay between the excitation and the response. For example, when generating a sine signal in a room there could be a significant delay between the generation of the signal and its detection by a measuring microphone due to the time required for the signal to propagate between the source and the microphone. In the signal generator there is a low pass filter after the detector whose response can be adjusted by the user using the softkey **FILTER [P]**. The selected value can be between 0 and 16, with 16 representing the slowest loop response. In general, the user would begin using a zero value and, should instability be observed which could be due to a time delay, this value could be increased to improve the stability.

For microphone testing in a relatively anechoic (acoustically non-reflective) environment, the reference and test accelerometer are replaced by reference and test microphones placed near one another (yet not so close as to interfere with the respective responses) in the radiated field of a loudspeaker. The output of the sine generator is used to drive the amplifier/speaker system and the reference microphone is used as the feedback signal for the generator.

Pink Noise Generator; Wideband or Bandlimited

The Wideband Pink Noise Menu, shown in figure 4-11, is accessed from the Signal Generator Menu by pressing **PINK [J]**.

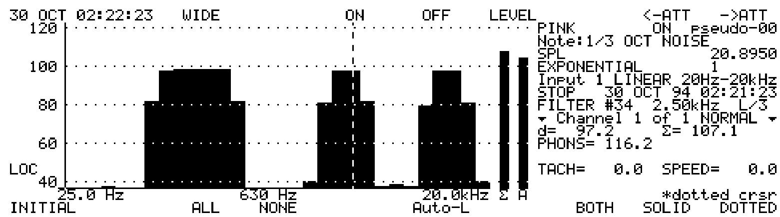
Figure 4-1 1 ***Wideband Pink Noise***



Pink noise has equal energy per percentage bandwidth (e.g., octave or 1/3 octave). As with the sine generator, the level is set using the softkey **LEVEL [E]** and entering a value between 0 and .9999 using the numeric keypad. The keys **<-ATTEN [G]** and **->ATTEN [H]** will produce attenuation or reduction in attenuation in 20 dB increments, also as described for the sine generator.

When 1/3 octave digital filters have been selected, the generator can also produce 1/3 octave bandlimited pink noise. From the Broadband Pink Noise Menu, the Bandlimited Pink Noise Menu, shown in figure 4-12, is accessed by pressing **1/3 [A]**.

Figure 4-12 ***Bandlimited Pink Noise***



As with the wideband pink noise, the key **LEVEL [E]** is used to set the relative output level of the signal, and the keys **<-ATTEN [G]** and **->ATTEN [H]** are used to add and remove attenuation at all frequencies in increments of 20 dB. The keys **ALL [I]** and **NONE [J]** are used to turn On or Off the noise in all 1/3 octave bands simultaneously. The On/Off status of the noise in individual bands can also be set on a band-by-band basis, using the keys **ON [C]** and **OFF [D]** to set the status of the band indicated by the active cursor. In this manner, noise can be generated for any combination of 1/3 octave bands, contiguous or not. When using the analyzer to measure the spectrum of the noise generated, the user should bear in mind the effect of filter selectivity (due to filter skirts not being perfectly vertical) on the measured spectrum. For example, with noise generated in a single band, the measurement will produce a spectrum indicating noise in the two adjacent sidebands at levels approximately 17 dB lower. This phenomenon is associated with the measurement process only, and does not represent the much more accurate bandlimited noise actually being produced.To return to the Wideband Pink Noise Menu, press **WIDE [A]**.

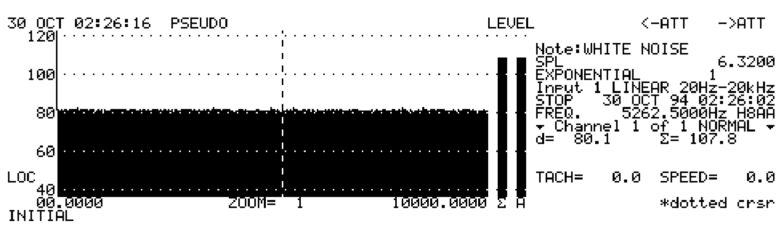
Autolevel Control; Bandlimited Pink Noise

In the Bandlimited Pink Noise mode, the autolevel feature is used to improve the noise spectrum in a test room for the purpose of making sound decay measurements. Because the sound absorption of room surfaces tends to increase with frequency, it usually happens that the spectrum of a room excited with pink noise will fall off greatly with increasing frequency. This makes it difficult to measure the decay of all bands in real-time because the levels in the higher frequency bands are not sufficiently greater than the background noise to make a good measurement. In well-equipped test laboratories, a 1/3 octave spectrum shaper is often used to shape the frequency spectrum of the electrical excitation to the amplifier driving the speaker such that a flatter sound spectrum is obtained in the room. As with the sine wave autoleveling function, the signal measured in channel 1 is used as the reference. Upon pressing **Auto-L [L]**, the level difference between each frequency band of the measured signal and that of the frequency band having the lowest level is noted. Then, the output level for each of these higher level frequencies is decreased by that amount in order to produce a spectrum which will more nearly approximate a flat 1/3 octave spectrum inside the room. Unlike the autolevel in the sine mode, this is not a feedback operation but simply a single correction to the output spectrum which takes place when the key is pressed.

White Noise Generator; Wideband or Pseudo

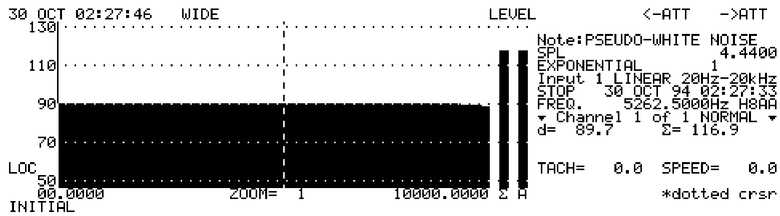
The Wideband White Noise Menu, shown in figure 4-13, is accessed from the Signal Generator Menu by pressing **WHITE [K]**. White noise has equal energy per constant bandwidth (e.g., narrowband FFT analysis). Wideband white noise can be used with either digital filters or FFT analysis. The **LEVEL [E]**, **<-ATTEN [G]** and **->ATTEN [H]** keys are used in the same manner as described above for the pink noise generator.

Figure 4-1 3 ***Wideband White Noise***



When FFT analysis has been selected, a pseudo-white noise output is possible. The Pseudo-White Noise Menu, shown in figure 4-14, is accessed from the White Noise Menu by pressing the key **PSEUDO [A]**.

Figure 4-14 ***Pseudo-White Noise***

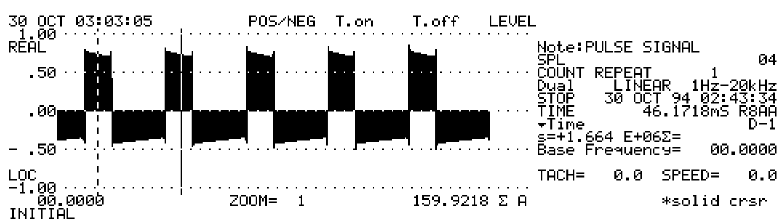


When FFT analysis is taking place, the analysis represents a finite number of frequency bands based on the number of lines selected for the measurement (100, 200, 400 or 800). It is not really necessary, therefore, for the generator to produce a truly wideband signal. In the pseudo-white noise mode, the signals are generated at the center frequency of each of the bands represented by the lines of the frequency analysis. This permits the generator output power to be concentrated on the same frequencies which are being measured by the FFT analysis, even when zoom analysis is being performed.

Pulse Generator

The Pulse Generator Menu, shown in figure 4-15, is accessed from the Signal Generator by pressing **PULSE [L]**.

Figure 4-15 ***Pulse Generator***



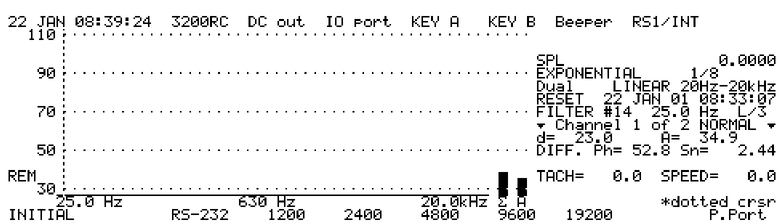
A series of either positive or negative pulses can be generated from this menu. The status of the generator is indicated briefly by a message on the upper right of the screen whenever one of the keys is pressed. The key **POS/NEG [B]** will toggle the polarity between positive and negative going pulses, as indicated on the upper right of the screen. The time increment for which the pulse is Positive or Negative, in milliseconds, is set by pressing **T.on [C]**, entering a value using the numeric keypad and pressing **ENTER**. The time increment for which the pulse is to have a zero value, in milliseconds, is set by pressing **T.off [C]**, entering a value using the numeric keypad and pressing **ENTER**. The **LEVEL [E]** key is used to set the output in the same manner as explained in the preceding sections.

When using the analyzer to measure and display the waveform generated using the pulse generator (Cross Mode, FFT Analysis, Count Averaging, Time display), the effect of even the lowest frequency highpass filter at the input (1 Hz) will be a DC offset as well as some distortion of the rectangular shape of the pulses. This is purely a measurement phenomenon, not a true representation of the actual signal being generated.

Interface Operations

The I/O Menu, shown in figure 4-16, is accessed from the System Menu by pressing **I/O [I]**.

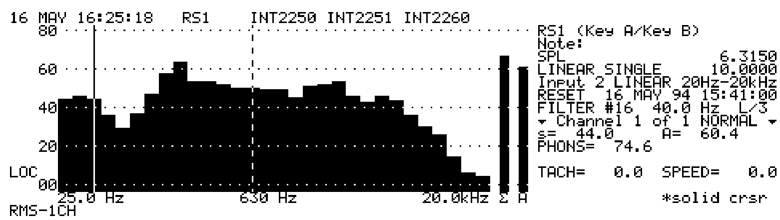
Figure 4-1 6 ***I/O Menu***



Selection of Intensity Probe or Remote Control

The opto-isolated inputs (see page 4-26) can be used, among other things, to communicate with a sound intensity probe or a remote control. To make a selection press **RS1/INT [H]** from the I/O Menu, which will bring up the RS1⁄ INT Menu, shown in figure 4-17.

Figure 4-17 ***RS1/INT Menu***



To select the use of a remote control, described in the following section, press **RS1 [A]**

To select the use of a sound intensity probe, press one of the following based on the Model number of the probe being used.

INT2250 [B], INT2251 [C] or INT2260 [D]

Remote Control using Model 3200RC Remote Control Module

The Model 3200RC Remote Control Module communicates with the Model 3000+ through a cable to the RS-232 interface. This module permits the user to perform the following activities:

Step 1 Run the analyzer

Step 2 Stop the analyzer

Step 3 Stop the analyzer and store the data block

Step 4 Examine the names (labels) of the seven user-defined analyzer setups

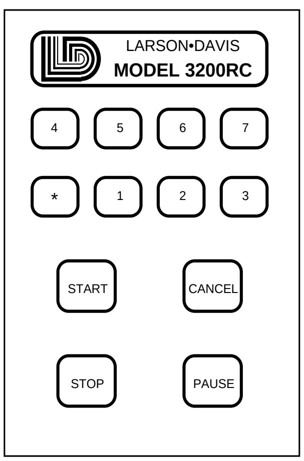
Step 5 Reboot the analyzer to one of the seven user-defined setups. The module is powered by an internal 9 volt battery. However, the RS-232 circuit board can be modified to power the remote control unit through the RS-232 cable. If the remote control unit is ordered at the same time as the analyzer, this modification will already have been made upon delivery.

Setup

Connect the 3200RC to the 3000+ serial port (RS-232 interface). From the I/O Menu, press **3200RC [A]** to activate the communication link. Press the 3200RC **START** key to turn on the module. The 3200RC will display the message “L-D RC Terminal System Ready”. If the cable is connected and working properly, the 3000+ will display the message “3200RC is on” on the upper right of the screen, and then the 3200RC will display the message “Communications with 3000+ OK”. If instead the message “L-D RC Terminal System Ready” remains on the module, there is either a problem with the cable or the user has not set the 3000+ for the 3200RC function.

Operation

The following keys on the 3200RC module are used for remote operation of the 3000+:



3200RC Key	Function
START	Places the 3000+ in the RUN mode and displays the message “START” on the 3200RC. If the last command from the 3200RC had been STOP or CANCEL, the data buffer will be reset before the analysis begins. If the last command from the 3200RC had been PAUSE the data buffer is not reset before the analysis begins.
STOP	Places the 3000+ in the STOP mode and stores the data block. The 3200RC will display the message “STOP, STORED n” where n is the record number into which the data block has been stored.The next press of START will reset the data buffer before starting the analysis.
CANCEL	Places the 3000+ in the STOP mode without storing a data block. The 3200RC will display the message “CANCEL”. The next press of START will reset the data buffer before starting the analysis.

3200RC Key	Function
PAUSE	Places the 3000+ in the STOP mode without storing a data block. The 3200RC will display the message “PAUSE”. The next press of START will continue the analysis without resetting the data buffers

Communication with User-defined Setups

The ability of the 3000+ to name (label), store and recall up to seven user-defined instrument setups is described in Chapter 12. The user should read this chapter before proceeding with this section. The following keys on the 3200RC are used to communicate with the remote setups of the Model 3000+.

3200RC Key	Function
Numeric keys, 1-7	The numbers 1-7 refer to the seven softkeys aligned below the 3000+ display which can be labeled by the user and to which user-defined setups can be stored. The numbers 1-7 are assigned from left to right across the row of softkeys.Upon pressing one of these numeric keys, the user-defined label for that softkey will be displayed on the 3200RC in the format “Setup #n is: sssssss”.
*	This key represents a shift function. Pressing it will produce an “s” on the lower right of the 3200RC display. The subsequent keypress will be treated as a different function than the keypress without the preceding shift operator. After the keypress following the shift has been made, the shift state is reset to normal. After the shift has been initiated by the * key, and before another key has been pressed, a second press of the * key will reset the shift state to normal.
• followed by a digit 1-7 (e.g. shifted digit)	The 3000+ will reboot the setup represented by the digit. The 3200RCwill display the message “Boot setup #n ssssss” where n is the setup number and sssssss is the setup name (label).

3200RC Key	Function
* START (e.g. shift, START)	The 3200RC will turn off. The 3000+ will display the message “3200RC is off”.

The 3200RC will turn itself off after a period of 15 minutes without a keypress in order to preserve the battery power.

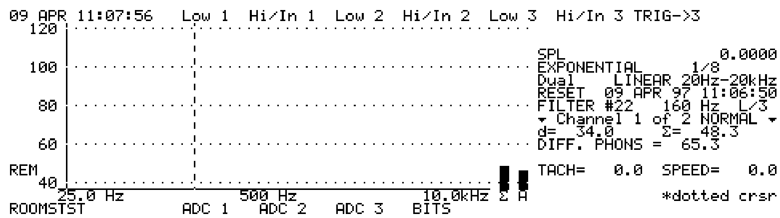
DC Output

There is a connector on the rear panel of the 3000+, as indicated on the label, which produces a DC voltage proportional to a user-selected frequency or sound level meter parameter. The load impedance should be at least 2 kΩ. Full scale is represented by 4.5 volts, decreasing 1 volt/20 dB. This is selected from the I/O Menu by pressing **DC out [B]**, which will produce the message “DC output @ XXXXXX”, where XXXXXX indicates which frequency band or sound level meter parameter is to be represented by the DC output voltage. If the 3000+ is set for 1/1 or 1/3 octave bandwidths, the message will indicate both the ANSI filter number and the center frequency. If it is set for FFT analysis, only the center frequency will appear. The horizontal arrow keys are now used to select a frequency or sound level meter parameter. The sound level meter parameters are higher in sequence than the frequency bands, meaning that when a frequency band is indicated, continual presses of the right arrow key will access higher and higher frequencies, and following that the sound level meter parameters SLOW, SLOW MIN, SLOW MAX, FAST, FAST MIN, FAST MAX, IMPULSE, IMPL.MIN, IMPL.MAX, LEQ, SEL, PEAK, and finally Spectrum ∑. Spectrum ∑ is not a sound level meter parameter, but the summation of the energy contained in all the bands of the frequency spectrum. Presses of the left arrow will move downward in sequence through these sound level meter parameters and then the frequency bands.

I/O Port Control

As indicated by the label on the rear panel of the 3000+, there is a separate I/O Port connector for communication with external devices. Operations with this port are performed from the I/O Port Menu, shown in figure 4-18, which is accessed from the I/O Menu by pressing **IO Port [C]**.

Figure 4-18 ***I/O Port Menu***



A/D Inputs #1, #2 and #3

The connector pins 2, 3 and 4 are connected to three separate 8-bit A/D converters for the purpose of reading the DC voltage applied by external devices. These could, for example, be used to read the output of pressure and temperature transducers. The input voltage can cover the range 0 to 5 volts. To read the voltage applied to these inputs, from the I/ O Menu press one of the following: **ADC 1 [I]**, **ADC 2 [J]**, or **ADC 3 [K]**. This will produce the message “ADC Volts = X.XX” on the upper right of the screen, where X.XX is the voltage level corresponding to the ADC which was selected by the keystroke.

I/O Channels #1, #2 and #3

These are associated with the connector pins 5, 6 and 7. Each one has two TTL states; either set and held Low or set High with the possibility of being set Low by an external device. The CPU monitors the states of all three of these pins. The state of each is set Low by pressing the following:

Softkeys	Softkey Functions
Low 1 [A]	Sets I/O Channel 1 Low
Low 2 [C]	Sets I/O Channel 2 Low
Low 3 [E]	Sets I/O Channel 3 Low

The state of each is set High, with the potential of having the state changed by an input signal, by pressing the following:

Softkeys	Softkey Functions
Hi/In 1 [B]	Sets I/O Channel 1 High
Hi/In 2 [D]	Sets I/O Channel 2 High
Hi/In 3 [F]	Sets I/O Channel 3 High

To display the status of all three of these pins, press **BITS [L]**. The I/O Channel 1 has an alternative function as a counter such that is can measure the frequency of an input pulse train. A typical application is with a wind speed monitor which produces pulses as it rotates.

TTL Indication ofTriggering

When the analyzer is set to be triggered by a frequency domain trigger, this function will cause the state of I/O Channel 3 to switch from High to Low when the trigger occurs. With the frequency domain trigger setup, before running the analyzer, access the I/O Port Menu and press **TRIG->3 [G]** which will activate the function. The bit status message will appear on the upper right to show that the state of the I/O Channel 3 is HT; High waiting for a Trigger. It will be necessary to exit from this menu before putting the analyzer in the run state. Press **R/S** to begin frequency analysis in the armed state. When the trigger criteria are satisfied, the spectrum averaging will begin and the state of the I/O Channel 3 will switch to low.

Frequency Domain Interface Trigger of I/O Channel 3

It is possible to trigger the analyzer into the Run state based on the level of the input signal in a specified frequency band (or the sum band, or one of the sound level meter parameters), as described in Chapter 11. This is referred to as the Frequency Domain Trigger Function.

This same criteria can be used to change the state of the I/O Channel 3 from High to Low, independent of whether the Frequency Domain Trigger Function controlling the Run state is active or not. We shall refer to this as the Frequency Domain Interface Trigger. By monitoring the state of I/O Channel 3, the user can detect the satisfaction of the trigger criteria by the change of state from High to Low. It is the responsibility of the user, however, to provide the hardware necessary to invoke the desired action based upon the detection of this change of state.

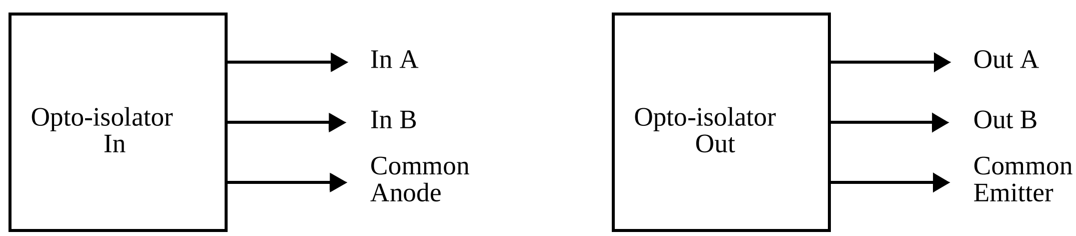
The desired frequency domain trigger criteria is established from the Frequency Trigger Menu exactly as described in Chapter 11. After the trigger criteria has been set, if this is not to be used to trigger the Run state (i.e., the frequency domain trigger function is to be inactive), be sure to press **OFF [O]** before exiting from the Frequency Trigger Menu. In preparation for the use of the frequency domain interface trigger, access the I/O Port Menu and press **Hi/ln 3 [F]** to set the Channel 3 state to High. To assign the frequency trigger criteria already established to change the state to Low upon satisfaction of the trigger criteria, press **TRIG [G]**. Subsequent presses of this key will toggle the Frequency Domain Interface Trigger between On and Off. When this is On, or active, a “T” will appear as part of the status of I/O Channel 3.

For example, suppose the model 3000+ is monitoring the sound pressure level produced by an external electronic noise source and the noise source is to be shut down if the peak sound pressure level exceeds a user-specified level. The user could develop a hardware module capable of switching off the signal to the speaker upon detection of the High-to-Low state change of I/O Channel 3. Since the Peak SPL is the parameter of interest, the SLM Mode should be selected. From the SLM Mode, it is necessary to access the Autostore Menu in order to access the Frequency Trigger Menu. From there, select to monitor the Peak Level, and select the “≥” criteria and use the **level [P]** to set the maximum permitted Peak level. Since the Frequency Domain Trigger function is not to be used, press **OFF [O]** before exiting the menu by pressing **EXIT**. Following this, return to the I/O Port Menu and press the key sequence **Hi/ln 3 [F]**, **TRIG [G]** and exit to the Main Menu to run.

Key A and Key B Control

The pins 9, 10, 11, 13, 14 and 15 are used to implement a pair of opto-isolated inputs and a pair of opto-isolated outputs, as shown in figure 4-19.

Figure 4-1 9 ***Opto-isolated Connections***



The most common use of this is in conjunction with the Larson Davis Model 2260 Acoustic Intensity Probe, which has two buttons (thumb and forefinger actuated) on the handle and a plug on the analyzer end of the cable for connection to the above mentioned receptacle. A separate control box for use with this receptacle is also available from Larson Davis Laboratories. These two keys are then programmed such that a press of each of these keys simulates a press of one of the softkeys or hardkeys of the 3000+.

Programming of the acoustic intensity probe keys is done from the I/O Menu. To program the forefinger actuated key, press **KEY A [D**], which will produce the message "PRESS EXIT, THEN THE KEY" on the upper right of the screen. Proceed by pressing **EXIT**, followed by whatever softkey or hardkey is to be simulated by a press of Key A. Program the thumb actuated key by similarly pressing **KEY B [E]**, **EXIT**, then the softkey or hardkey which KEY B is to simulate.

In many cases, these keys are programmed for rapid acquisition and storage of data when using the acoustic intensity probe by programming KEY A to simulate the **RUN/STOP** key and KEY B to simulate the hardkey **STORE**. When exponential averaging has been selected, the user will press KEY A once to initiate averaging, another time to stop averaging, then KEY B to store the measured data.

In many instances, however, linear averaging will be selected and the user will scan the probe across a particular surface during the linear averaging process. In this case, one press of Key A will initiate averaging, and the averaging will automatically stop at the end of the averaging time, which will be indicated on the LED on the upper end of the probe handle. Then, simply press KEY B to store the data.

Beeper Control

The function of the audio beeper signal is controlled from the Beeper Menu, accessed from the I/O Menu by pressing **Beeper [F]**. The beeper can be programmed to output an audio signal corresponding to the following:

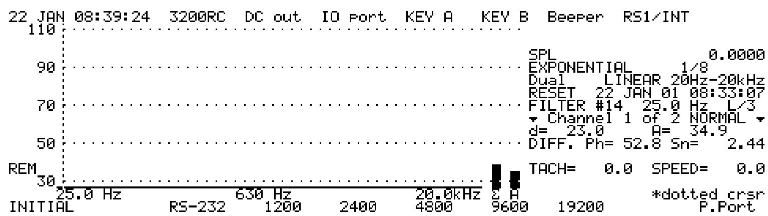
- The press of a hardkey or softkey
- An overload condition at one of the inputs
- An error condition
- Any combination of the above three

The following softkeys program the beeper function as follows:

Softkeys	Softkey Functions
NONE [A[	No beeper signal at all
ALL [B]	Beeper signal for 1, 2 and 3
KEYS [C]	Toggles On/Off the beeper signal for a key press
OVER [D]	Toggles On/Off the beeper signal on overload condition
ERROR [E]	Toggles On/Off the beeper signal on error condition

Selecting the RS-232 or Parallel Interface

The RS-232 interface is made active by pressing **RS-232 [I]**.



The Baud rate for the data transfer across the interface (kilo-byte/second) is selected by pressing one of the following softkeys:

Baud	Softkeys
1200	**[J]**
2400	**[K]**
4800	**[L]**
9600	**[M]**
19200	**[N]**

The Parallel port is made active by pressing **P.Port [P]**. Use the optional CBL126 cable to connect between the 3000+ parallel port and the computer.

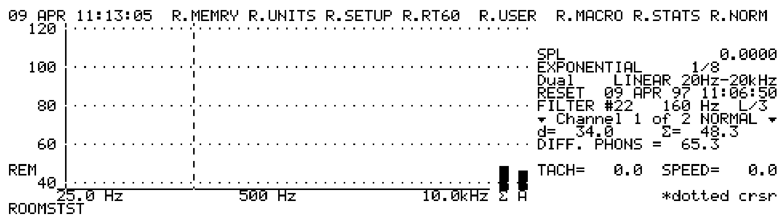
Setting the Clock

The internal clock of the 3000+ is set from the System Menu by pressing **clock [M]**. A message on the upper right of the screen will prompt the user to enter the data and time, in the format "DD/MM/YY HH:MM:SS" using the keypad, and press **EXIT**.

The Resets Menu

Although there is often a reset softkey in individual softkey menus to reset certain user-defined items, or even the entire RAM memory, the Model 3000+ has combined these into a single Reset Menu as well for efficient access. The Reset Menu, shown in Figure 4-20, is accessed from the System Menu by pressing **RESETS [P]**.

Figure 4-20 ***Resets Menu***



Pressing each of the following keys will initiate the particular reset indicated.

Softkeys	Softkey Functions
R.MEMRY [A]	Reset of RAM Memory, loss of stored data
R.UNITS [B]	Reset of User-Defined Units
R.SETUP [C]	Reset of User-Created Setups
R.RT60 [D]	Reset of RT60 Register
R.USER [E]	Reset of User-Weighting
R.MACRO [F]	Reset of Macros
R.STATS [G]	Reset of Ln Statistics Table
R.NORM [H]	Reset of Normalization Function

In each case, upon pressing the softkey a message on the upper right of the screen will request user verification of the reset operation. To continue the reset operation press **YES [A]**. To abort the reset operation, press **NO [C]**. To exit from the Reset Menu, press **EXIT**.

Remaining System Softkeys

The softkeys appearing in the System Menu which have not been described in this Chapter are explained in detail in later chapters as follow:

Softkeys	Softkey Functions
SETUP [N]	This softkey is used to access the Setup Menu for creation, storage and recall of user-created setups of the Model 3000+. See Chapter 12.
FILES [O]	This softkey is used to access the Files Menu for creation and manipulation of data files and records associated with the internal memory and the floppy disk of the Model 3000+. See Chapter 13.

CHAPTER

5 Selection of Averaging Parameters

After the analyzer has been setup from the System Menu, the user will exit to one of the three Analysis Menus to perform a measurement. Before actually beginning the measurement, the averaging parameters should be defined. Since the same theoretical considerations apply to the averaging setup for each of these analysis types, this chapter is devoted to that subject and placed in sequence before the chapters describing the detailed operation of the analyzer.

Selecting Averaging Type

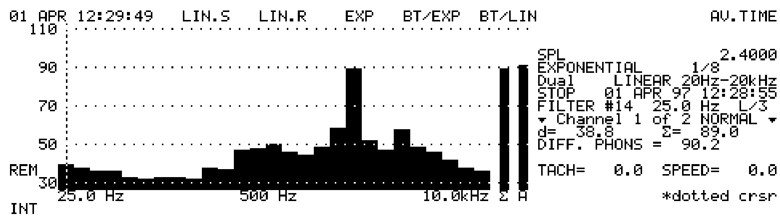
Accessing Averaging Menu

From any of the three Analysis Menus, access the Averaging Menu by pressing **DETECTR [H]**. The range of choices of averaging type will be represented by the softkeys along the top of the display. The choices will be different depending upon whether Octave or FFT filtering has previously been selected. Further information about these averaging methods is presented later in this chapter under Signal Averaging Considerations.

Averaging Type: Octave Filters

When Octave filtering has been selected, pressing **DETECTR [H]** will cause the Menu illustrated in Figure 5-1 to be displayed.

Figure 5-1 ***Octave Averaging Type Menu***



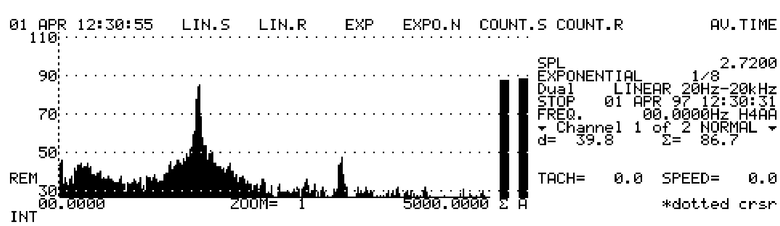
Select the desired averaging method by pressing one of the following:

<u>Softkeys</u>	<u>Softkey Functions</u>
LIN.S [A]	Linear Single (seconds)
LIN.R [B]	Linear Repeat (seconds)
EXP [C]	Exponential (seconds)
BT/EXP [D]	Constant Confidence with Exponential Averaging
BT/LIN [E]	Constant Confidence with Linear Averaging

Averaging Type: FFT Filters

When FFT filtering has been selected, pressing **DETECTR [H]** will cause the Menu illustrated in Figure 5-2 to be displayed.

Figure 5-2 ***FFT Averaging Type Menu***



Select the desired averaging method by pressing one of the following:

Softkeys	Softkey Functions
LIN.S [A]	Linear Single (seconds)
LIN.R [B]	Linear Repeat (seconds)
EXP [C]	Exponential (seconds)
EXPO.N [D]	Exponential Averaging based on number of spectra (# spectra)
COUNT.S [E]	Spectral Single (# spectra)
COUNT.R [F]	Spectral Repeat (# spectra)
COUNT.M [G]	Spectral, Manual Accept (only available in Cross analysis mode)

With Linear Single, averaging continues until the specified averaging time is reached, then averaging is stopped.

With Linear Repeat, after the averaging time is reached the detector is reset and the averaging process is begun again; this repeats until manually stopped.

Exponential produces a running time-averaging process similar to an RLC analog circuit. This averaging must be stopped manually.

Constant Confidence averaging produces a running time-averaging with a different effective averaging time for each filter such that the same statistical accuracy for a noise signal is obtained for each frequency band. The Exp or Lin refer to the algorithm used to calculate the averaged value. This averaging method must also be stopped manually.

In Spectral Single averaging, the individual spectra are averaged together until a specified number of spectra is reached, then averaging is stopped. A check for overloads is performed as each spectrum is produced; overloaded spectra are rejected from the averaging process.

With Spectral Repeat, the detector is reset and the spectrum averaging begun again, until a manual stop.

Spectrum averaging with Manual Accept is mainly used when measuring structural frequency response functions with an instrumented hammer. The data generated from each individual hammer blow is examined, and if satisfactory, accepted manually for inclusion in the spectrum averaging process. This continues until the desired number of spectra have been averaged together, or until manually stopped.

Note that the averaging type message displayed on the right of the 3000+ will change as the selected type is changed.

Averaging Time

After an averaging type has been selected, press **AV.TIME [H]** to select an averaging time.

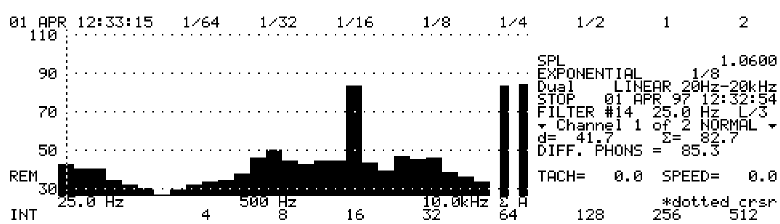
Averaging Time with Linear Types

When the selected averaging type is Linear Single or Linear Repeat, a message on the upper right of the display will prompt the user to enter a value, in seconds, using the keypad, then press **EXIT**.

Averaging Time with Exponential Types

When the selected averaging type is Exponential, the Menu shown in Figure 5-3 presents the user with 16 different values of averaging time, from 1/64s to 512s, in a binary sequence. Press the key above or below the desired value, then press **EXIT**.

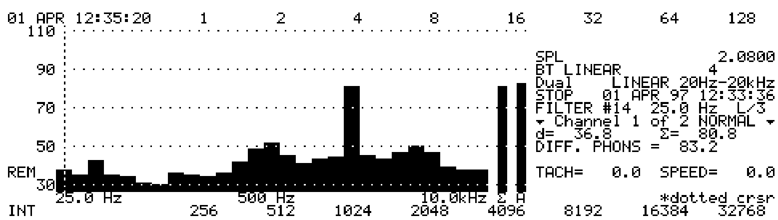
Figure 5-3 ***Exponential Averaging Time Menu***



Averaging Time with Constant Confidence Type (Octave Bandwidths Only)

When the selected averaging type is Constant Confidence, the Menu shown in Figure 5-4 will present the user with 16 different values of BT product, from 1 to 32,786, in a binary sequence. Press the key above or below the desired value, then press **EXIT**.

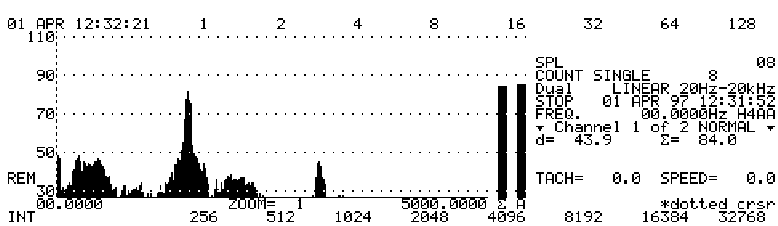
Figure 5- 4 ***Constant Confidence Averaging Time Menu***



Averaging Time with Spectral Type Averaging (FFT Bandwidths Only)

When the selected averaging type is EXP.N or Count Single, Count Repeat or Count Manual, the Menu shown in Figure 5-5 will present the user with 16 different values, from 1 spectrum to 32,786 spectra, in a binary sequence. Press the key above or below the desired value, then press **EXIT**.

Figure 5- 5 ***Count Averaging Menu***



The averaging times for Linear, Exponential, Constant Confidence and Spectral averaging are stored independently. Once a value has been selected for each of these, that value of averaging time will reappear with the selection of that averaging type.

A common procedure is to begin an analysis using a fast exponential averaging while the range setting is adjusted, then change to a long linear average to obtain a very accurate result. Prepare by selecting exponential averaging, select an averaging time of 1/64s, then select linear single, and an averaging time of 16s. Now, whenever exponential is selected the 1/64s time will appear, but when the averaging method is changed to linear single, the 16s time will be recalled.

Signal Averaging Considerations

The spectra measured with most acoustic and vibration transducers contain a certain amount of random variability, partly because the exciting force and mechanical response characteristics are frequently random in nature (such as aerodynamically-generated sound and structural vibration) and partly because there is often random noise associated with the measurement system itself.

Stationary Signals

A signal which is steady, except for the effect of random variations which have well-defined statistical characteristics, is referred to as stationary.

Time Averaging

The statistical accuracy of measuring stationary signals can be increased by Time Averaging. In principle, the accuracy of the measurement increases as the averaging time is increased: the trade-off is between accuracy and measurement time.

For Gaussian white noise passing through an ideal bandpass filter, the standard deviation can be approximated as follows:

$$\varepsilon = \frac{4.34}{\sqrt{BT}}$$

ε: Standard deviation of the rms level (dB)

B: Bandwidth of Bandpass (Hz)

T: Averaging Time (sec)

From this, the statistical accuracy of the measurement can be estimated as follows:

There is a 68.3% probability that the measured value is within $\pm 1\varepsilon$ of the statistically correct value, a 95.5% probability that it is within $\pm 2\varepsilon$ of the correct value, and a 99.7% probability that it is within $\pm 3\varepsilon$ of the correct value.

Example: We are performing an 800 line FFT measurement on a random signal using a frequency range of 0 to 10 kHz, and wish to have 68.3% confidence that the result is within 0.5 dB of the statistically correct value. What value of averaging time should we select?

B = 10,000 Hz/800 = 12.5 Hz

$$T = \frac{1}{B}\left(\frac{4.34}{\varepsilon}\right)^2 = \frac{1}{12.5}\left(\frac{4.34}{0.5}\right)^2 = 6s$$

If the frequency range were 0-500 Hz, an equally accurate measurement would require an averaging time of 120s.

Linear Time Averaging

When using the FFT measurement mode of the instrument, the bandwidths are all equal, and therefore the averaging time required for equal accuracy in each bandwidth will be the same. Therefore, a linear average is generally selected when measuring stationary signals with constant bandwidth filters. The linear average for each filter is calculated by dividing the sum of the individually sampled values (in linear units) by the number of samples.

Constant Confidence Time Averaging

When using constant percentage bandwidth filters such as 1/1 and 1/3, octave bands, the bandwidths become narrower as the center frequencies become lower. The statistical accuracy will be progressively less at lower frequencies when using a linear average with these filters. In order to have equal statistical accuracy for all bandwidths, each filter must have a different averaging time. For this reason, the constant confidence averaging (BT = constant) is generally used.

The following table shows the standard deviation associated with some of the BT product values provided with the digital filter version of the 3000+.

BT	ε (dB)
1	4.3
4	2.2
16	1.0
32	0.8
64	0.5
256	0.3
512	0.2
2048	0.1
8192	0.05

When using the Model 3000+ in the constant confidence mode, either linear or exponential averaging may be selected. Exponential averaging is discussed below under Transient Signals.

Spectrum Averaging

When the amplitude of the signal is too high during Time Averaging, the instrument will indicate with the message OVER that an overload has occurred. The only way to correct this situation is to keep increasing the range and taking measurements until no overloads occur.

In the FFT mode, spectrum averaging is permitted: The spectrum of each measurement is included in a single averaged spectrum. The number of spectra to be averaged is specified, but before a spectrum is included in the average, it is examined for overloads. Overloaded spectra are rejected. Measurement and rejection continue until the number of specified spectra to be averaged is reached.

Periodic Signals

When the signal is periodic (such as a sinusoid, square, or triangular wave), the amplitude as a function of frequency is well-defined. The detector will provide an accurate measurement using an averaging time on the order of one period (1/frequency), so lengthy averaging times are not required. Near periodic signals are observed in rotating machinery at the first 3 to 5 harmonics of the shaft rotation speed and in gearboxes at the tooth mesh frequencies and their harmonics.

Transient Signals

When the spectral characteristics of a signal are changing with time, we face a more complex situation. Typically, one would wish to measure and observe a series of spectra taken at regular time intervals chosen sufficiently small that the time-varying behavior of the spectra is clear. This implies that the averaging time must be no larger than the time interval which represents a significant change in the spectral characteristics. Conflicting with this is the requirement that the averaging time be sufficiently long so that a statistically accurate result is obtained. Because it is not always possible to satisfy both requirements, a certain amount of experimentation with averaging times may be required to define the signal properly.

Linear Repeat Time Averaging

To utilize linear averaging for the analysis of a transient signal, select Linear Repeat. Beginning with the trigger or the pressing of the **R/S** key, the analyzer will measure and display a series of linearly averaged spectra. Since the detector is reset at the end of each averaging period, each spectrum will represent the frequency analysis of time data occurring only during the time period when that average was being calculated. To store these spectra, put the unit in the Autostore mode, and use a data storage interval equal to the value of the linear averaging time.

Exponential Time Averaging

Exponential averaging is based on the averaging characteristics of analog RC detector circuits. If a step input is applied to such a detector, the output value rises in an exponential manner until the output level finally reaches that of the input level. The time constant of such a detector is defined as the time required for the output level to reach 99% of the input level.

With a digital analyzer, a time constant is selected, and a linearly averaged value is calculated for each time interval equal to the time constant. The exponentially averaged value produced at any instant is calculated as a weighted sum of the previously measured linear values, with the most recent values contributing the most weight to the sum. As the analysis proceeds, the result is a running average which is dominated by the most recent value but which is also smoothed out by the contributions of the preceding values.

An example is the analog meter. The position of the needle follows the output of the RC-averaging circuit. When the averaging time is small, the needle may oscillate very rapidly in response to a varying signal. The observer can see the short-term variation of the signal, but the averaging time may be too short to provide a readable value. With a longer averaging time, a smaller, less rapid needle variation occurs, providing a more readable number.

Exponential averaging is frequently used when one wishes to visually observe on the display screen the time-dependent behavior of the spectrum of a signal. The averaging time is adjusted until the short-term variations are minimized, yet the response is still sufficiently rapid to follow the long-term time variation of the spectrum. Even with steady signals, the user will frequently use exponential averaging to initially observe the signal and set the input attenuators for a near full-scale signal level without overloads. Then he will switch to linear averaging to perform the analysis more accurately.

CHAPTER

6

Analysis Menus; Selection Of Measurement And Display Parameters

The parameters which can be measured and displayed by the Model 3000+ depend upon which Analysis mode has been selected by the user from the system menu.

Standard Analysis: Spectral data are measured for each of the input channels, but no cross channel parameters are measured.

Cross Analysis: Frequency and Time Domain (FFT only) data are measured for each input channel and also cross channel parameters are measured between channels 1 and 2.

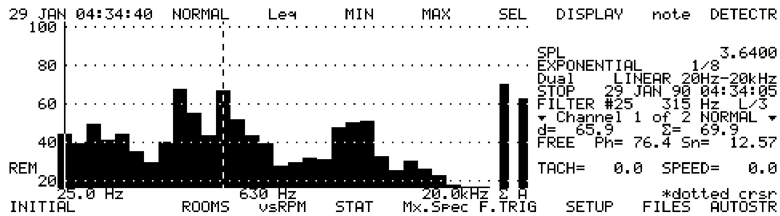
Intensity Analysis: Spectral data corresponding to acoustic intensity are measured. This implies a cross channel measurement between channel 1 and 2.

Since these are very different, we will discuss each within a separate section of this chapter. Control of the display formats, such as linear amplitude scales, logarithmic frequency scales for FFT, combining percentage bandwidths to obtain larger bandwidths, and the readout of the broadband (A-Weight and Linear) levels will be discussed later.

Standard Analysis

After having setup the analyzer for Standard Analysis, Single or Dual Channel, in the System Menu, upon exiting from that Menu the Standard Analysis Menu will be displayed as shown in Figure 6-1. This Menu will be the same for either type of frequency analysis, Octave or FFT.

Figure 6-1 ***Standard Analysis Menu***



When single channel Standard Analysis has been selected, the active input channel is indicated on the right side of the screen by the message “Input 1” or “Input 2”. Use the hard-keys **CH1** and **CH2** to select which is to be used.

When dual channel Standard Analysis has been selected, this message will read “Dual” to indicate that both channels are being measured simultaneously.

Selection of Display Format for Dual Channel Mode

When dual channel Standard Analysis has been selected the user can choose to view the spectrum for channel 1, the spectrum for channel 2, or the spectra for both channels in a side-by-side format. For a single channel spectral display press **CH1** or **CH2** depending upon which channel is to be displayed. This selection is also indicated on the right of the display, 6th line down, by the message “Channel 1 of 2” or “Channel 2 of 2”.

The dual channel display mode is activated from the Display Menu, accessed from the Main Menu by pressing **DISPLAY [F]**. From the Display Menu, repeated presses of the softkey **Multi [H]** will toggle the dual display mode between ON and OFF. When the dual channel display mode is active the message “Multichannel Display” will appear below the horizontal axis toward the left side. When the dual channel display mode is active the **CH1** and **CH2** keys are used to position the cursor to readout the levels corresponding to either the channel 1 spectrum on the left half of the display, or the channel 2 spectrum on the right half of the display. This will also be indicated by the message “Channel 1 of 2” or “Channel 2 of 2” on the right of the display, 6th line down.

Selection of Display Parameter

Select the particular spectrum type to be displayed by pressing one of the following:

Softkeys	Softkey Functions
NORMAL [A]	Normal displays the averaged spectrum as selected from the Averaging Menu.
Leq [B]	Leq is a parameter generally used for the analysis of time-varying acoustic signals. It represents the steady level which, integrated over a time period, would produce the same energy as the actual signal. The time period used for the calculation is the elapsed time since the last data reset.
MIN [C]	A Minimum spectrum displays the minimum signal level measured in each filter band since the last data reset.
MAX [D]	A Maximum spectrum displays the maximum signal level measured in each filter band since the last data reset.
SEL [E]	SEL (Single Event Level) is similar to Leq, except that it represents the steady signal which, integrated over a one second time period, would produce the same energy as the actual signal over the time since the last data reset.

While making a measurement, one generally selects Normal in order to view the spectrum of one, or both, of the input channels live on the display. However, for any measurement made in the Standard Analysis Mode, the four other spectra (Leq, Min, Max and SEL) are also calculated for channel 1 (STAND 1) or both channels (STAND 2). These can be viewed during or after the measurement by simply selecting the desired spectrum type as explained above. To change the displayed channel for dual channel analysis, simply press **CH1** or **CH2**.

The displayed spectrum type is indicated on the right of the screen (sixth line down). If the measurement is stopped and started repeatedly without a reset, the calculation of the Leq, Min, Max and SEL spectra continue without reset and will include the contributions of all signals since the last reset.

Max Spectrum Display

There are some applications such as vehicle passby measurements where the user wishes to display the frequency spectrum associated with the highest broadband level which has occurred during a testing interval. In addition to calculating the Leq, Minimum, Maximum and SEL spectra during a measurement sequence (since last data reset) as described above, the 3000+ also saves a spectrum (Max Spectrum) corresponding to the highest broadband level which has occurred. The specific broadband level which is used to determine the Max Spectrum depends upon the operational mode of the 3000+. When the 3000+ is in the SLM mode, it is the sound pressure level measured by the sound level meter function which is used. This permits different weightings to be applied to the SLM and analysis functions. For example, if the user wishes to see the unweighted spectrum associated with the maximum weighted sound pressure level, he should measure the Max Spectrum with the 3000+ in the SLM mode, with SLM function weighted appropriately (A or C) and the analysis function set to one of the four linear weightings. When the sound level meter response is set to Slow, Fast or Impulse, the sound pressure level measured for that response is used in determining the Max Spectrum. If the response is set to Leq, the Slow response will be used for this.

When the 3000+ is in the Standard Analysis mode, it is the broadband sum level calculated by summing the energy in all the frequency bands indicated by the height of the vertical bar on the right of the frequency spectrum display which has the summation symbol beneath it. In this case, both the broadband level and all measured spectra will have the same analog weighting as selected by the user. However, if the measurement had been made using A or C-weighting, the user could still use the (-A) or (-C) digital display weighting functions to examine the measured spectra in an unweighted form.

To display the Max Spectrum when in the frequency analysis mode, from the Analysis Menu press **Mx.Spec [L]**. When in the SLM mode, this is done by pressing the key sequence **DISPLAY [F], Mx.Spec [C]**. The message “Mx.Spec” on the right of the screen alongside the displayed channel number indicates that the spectrum being displayed represents the spectrum associated with the highest broadband level since the last data reset. Note that this spectrum is not automatically stored to memory. The user must press **STORE** to store the Mx.Spectrum or Spectra. The storage and recall of data records is discussed in Chapter 13.

For each vehicle passby measurement, the user would reset the data buffer by pressing **SHIFT, RESET**, begin the measurement by pressing **RUN/STOP** as the vehicle approaches, end the measurement by pressing **RUN/STOP** again after the vehicle has passed by, press **Mx.Spec [L]** to recall the Max Spectrum and then press **STORE** to store the spectrum. The user can perform a data reset operation while the measurement is in progress, which will initialize the Leq, Minimum, Maximum, SEL and Max Spectrum values to zero and begin the calculations again.

Dual Channel Display Mode

When the Model 3000+ is configured for dual channel measurements in the Standard Analysis Mode, it is possible to display the spectra for both channels simultaneously in a side-by-side configuration. From the Main Menu, presses of the softkey sequence **DISPLAY [F]**, **Multi [H]** will toggle between the single and the dual channel display formats. This function is described in more detail in Chapter 19, Control of Display Formats.

Loudness Measurement

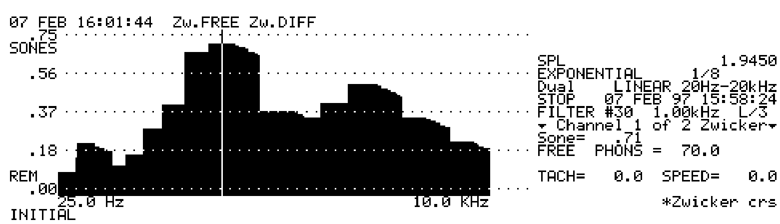
When in the Standard Analysis Mode with 1/3 octave filtering, the Loudness in sones and the Loudness Level in phons corresponding to ISO Recommendation R523 Method B (sometimes referred to as Zwicker loudness after the researcher who developed the method) are calculated and displayed on the lower right of the screen. There are two forms which correspond to measurements made in diffuse acoustic fields where energy is from all directions such as inside a reflective space, and measurements made in the free-field where the sound is radiated without reflection from a single acoustic source. The display form “DIFF. Ph=XX Sn=YY” indicates that the diffuse field method is active while the format “FREE Ph=XX Sn=YY” indicates that the free-field method is active. The units of loudness level is phones.

The selection of which form of the Loudness function is to be calculated is done from the Digital Display Menu, accessed by pressing the softkey sequence **DISPLAY [F], Dig.WGT [I]**. To select the desired form of loudness press one of the following:

Softkeys	Softkey Functions
Zw.FREE [A]	Free-field Loudness.
Zw.DIFF [B]	Diffuse-field Loudness

After one of these keys have been pressed, the loudness in sones, from which the loudness level is being calculated, is displayed as a function of critical bands as shown in Figure 6-2.

Figure 6-2 ***Loudness versus Critical Bands***



Although the cursor will move through each critical band and display the amplitude, the frequency values are only given to the nearest 1/3 octave band center frequency. When the analyzer is running a real-time display of loudness is provided. When stopped, the display is for the last spectrum in the data buffer.

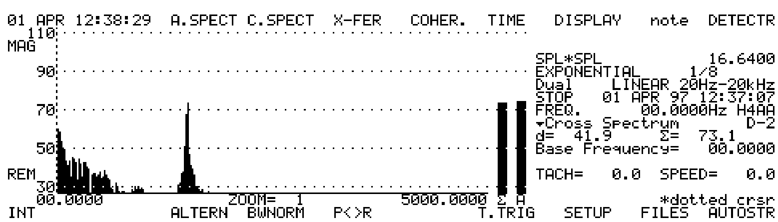
Cross Analysis

The Cross Menu which appears after setting up the analyzer and exiting from the System Menu will depend on the filter type (octave or FFT) which was selected, because FFT filters measure both time domain and frequency domain functions while octave filters measure only frequency domain functions.

Cross Analysis of FFT Filters

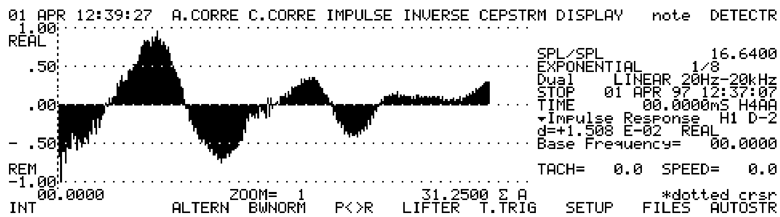
The Menu which is first displayed when Cross Analysis and FFT filtering have been selected is shown in Figure 6-3.

Figure 6- 3 ***FFT Cross Analysis Menu 1***



There are more parameters which can be measured and displayed when doing Cross Analysis with FFT filters than can be presented in one screen Menu, so two forms of this Menu are available. Pressing the softkey **DISPLAY [F]** will toggle the Cross Menu from the form shown above to the one shown below, which offers a different set of parameter choices.

Figure 6- 4 ***FFT Cross Analysis Menu 2***



Select the parameter to be displayed by pressing one of the softkeys listed below. The complex data types are indicated by an asterisk *. It is explained below how to change the display format of complex variables.

From first Menu form:

Softkeys	Softkey Functions
A.SPECT [A]	Autospectrum (each channel)
C.SPECT [B]	Cross Spectrum* (ch.2 vs. ch.1)
X-FER [C]	Transfer Function* (ch.2 vs. ch.1). Repeated presses display H1, H2 or H3 as defined below
COHER [D]	Coherence (ch.2 vs. ch.1). When the Coherence is being displayed, pressing **ALTERN [I]** will produce a display of Coherent Output Power. Repeated presses will toggle the display between Coherent Output Power and Coherence.
TIME [E] (count ave. only)	Weighted and Unweighted Time Waveforms (each channel). Sequence of Displayed Data: repeated presses of **TIME [E]** 1. Weighted Time Record 2. Time Record

From second Menu form:

Softkeys	Softkey Functions
A.CORRE [A]	Auto Correlation (each channel)
C.CORRE [B]	Cross Correlation* (ch.2 vs. ch.1)
IMPULSE [C]	Impulse Response* (ch.2 vs. ch.1)
INVERSE [D]	Inverse Transfer Function* (H1) (ch2 vs. ch.1)
CEPSTRM [E]	Cepstrum (each channel)

Softkeys	Softkey Functions
LIFTER [L]	Liftered Spectrum (each channel)

In the Model 3000+, cross channel measurements are between channels 1 and 2. The basic measurements performed by the Model 3000+ in the FFT analysis mode are an autospectrum for each input channel (G_{11} and G_{22}) and a cross spectrum between Channel 1 and channel 2 (G_{12}). The remaining parameters are calculated from these as indicated below.

- Coherence

$$\gamma_{12}{}^2 = \frac{|G_{12}|^2}{G_{11}G_{22}}$$

- Transfer Function Estimates

$$H_1 = \frac{G_{12}}{G_{11}}$$

$$H_2 = \frac{G_{22}}{G_{21}}$$

$$H_3 = \left|\frac{G_{22}}{G_{11}}\right|$$

Displayed phase is from H_1

In the following, F^{-1} is inverse Fourier Transform.

- Autocorrelation

$$R_{ii(\tau)} = F^{-1}[G_{ii}];i = 1, 2$$

- Cross Correlation

$$R_{12}(\tau) = F^{-1}[G_{12}]$$

- Impulse Response

$$h(\tau) = F^{-1}[H_1]$$

- Cepstrum

$$C_{ii}(\tau) = F^{-1}[\log G_{ii}];i = 1, 2$$

- Liftered Spectrum

$$L_{ii}(f) = F[L(\tau_L) \bullet C_{ii}(\tau)];i = 1, 2$$

Where L (τ_L) is the lifter defined below:

L_s (t_L)	Short-pass lifter equals 0 for frequency greater than t_L, unity otherwise.
LL (t_L)	Long-pass lifter equals 0 for quefrency less than t_L, unity otherwise.

The value of τ_L is selected with the horizontal cursor keys as indicated by the "*lifter" message on the lower right of the display.

Selection and Indication of Displayed Channel

For all of the above frequency and time-domain parameters, the displayed channel is changed by pressing **CH1** or **CH2**.

In the case of autospectrum and autocorrelation, the displayed channel number refers to the input channel. The cross channel functions are always calculated between channel 1 (the reference channel) and channel 2.

Display of Complex Data Records:

A complex variable requires two functions to express it. One can represent such a function by a pair consisting of a magnitude function and a phase function (polar coordinates), or by a pair consisting of a real function and an imaginary function (rectangular coordinates). When the user presses the key to display one of these functions, he will obtain either a real or a magnitude function, as indicated at the upper left of the screen. To observe the other function within the coordinate system (real/imaginary or magnitude/phase) simply press **ALTERN [I]**.

To change from one coordinate system to the other (polar-rectangular), press **P<>R [K]**.

NOTE: The cursor does not change position when these keys are used. This permits the user to move the cursor to a frequency of interest, and read the values of magnitude, phase, real and imaginary for that frequency by simply changing the display format, *even though he cannot display more that a single function at one time.*

Display of Time Records

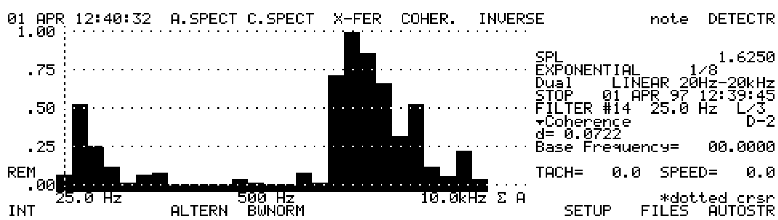
It is necessary to select Count Averaging (L, R or M) in order to be able to view one of the time records for each channel. This is so that an integral number of time records are dealt with in the averaging process. Sequential presses of **TIME [E]** will produce displays of the Weighted Time Record and the Time Record (unweighted).

The Time Record is the time waveform for each input channel as sampled and stored into the time buffer. The Weighted Time Record is the same data after having been multiplied by the time weighting function (Hanning, Flat Top, etc.) previous to the FFT calculation.

Cross Analysis with Octave Filters

Compared to the many parameters produced by Cross Analysis using FFT filtering, with Octave filtering only the frequency domain parameters are calculated and displayed. After setting up the analyzer and exiting from the System Menu, the Cross Menu with Octave Filters, shown in Figure 6-5, is displayed.

Figure 6- 5 ***Octave Cross Analysis Menu***



Select the parameter to be displayed by pressing one of the softkeys listed below.

Softkeys	Softkey Functions
A.SPECT [A]	Auto Spectrum (each channel)
C. SPECT [B]	Cross Spectrum* (ch.2 vs. ch.1)
X-FER [C]	Transfer Function* (ch.2 vs. ch.1). Repeated presses display H1, H2 or H3.

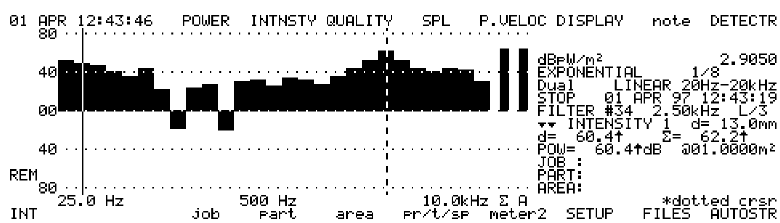
Softkeys	Softkey Functions
COHER. [D]	Coherence (ch.2 vs. ch.1). A second press will display Coherent Output Power
INVERSE [E]	Inverse Transfer Function* (ch.2 vs. ch.1)

An asterisk “*” indicates a complex parameter. In the Cross Analysis Mode using Octave filters, these are only available in the magnitude/phase format. Thus, the softkey **P<>R** will not appear in the Menu. Otherwise, the selection of channel number, form of transfer function and magnitude or phase display is as explained in the proceeding section.

Intensity Analysis

The Intensity Analysis Menu which appears after setting up the analyzer and exiting from the system Menu is shown in Figure 6-6.

Figure 6-6 ***Intensity Analysis Menu***



To select the desired display parameter, press one of the following:

Softkeys	Softkey Functions
POWER [A]	Sound Power
INTNSTY [B]	Intensity Spectrum
QUALITY [C]	Quality Spectrum (Intensity/SPL of Ch. 1)
SPL [D]	SPL Spectrum of Channel 1
P.VELOC [E]	Particle Velocity

3000+ Intensity parameters are measured between channel 1 and channel 2. See Chapter 20 for a more detailed discussion of intensity measurements and analysis.

Display of Broadband Data

Regardless of the Analysis Type, there are two vertical bars on the right of the spectrum display whose height represents broadband data calculated from the spectral data. The one with the summation symbol beneath it is the total energy calculated from the spectral data being displayed. This could be referred to as the linear or overall level. The one with the "A" beneath it represents the same spectral data with the A-weighting correction included, so it can be referred to as the A-weighted level.

If it is desired to accurately measure the energy between two frequencies other than the available combinations of lower and upper frequencies, use the Both Cursor readout format and place the two cursors at the desired frequency limits as described in Chapter 8.

When analog A or C-weighting has been selected at the input of the frequency analysis function, the "A" bar will show no data. However, since an analog weighting function is being used and the broadband data is calculated from the spectrum itself, the summation band now includes the effect of the analog weighting, and thus represents the A or C-weighted level.

A digital readout of the broadband level is presented on the right of the screen, seventh line down. Either the summation symbol or the letter "A" will appear to indicate which of the two broadband levels is being displayed. To switch between these two presentations, access the Display Menu and press **SUM [E]**, which toggles this digital display between them.

CHAPTER

7 Performing a Measurement

In this section we discuss the use of the **RUN/STOP** key to manually initiate and stop a measurement, how to reset the data buffer, manual control of input gain and use of the autoranging function for automatic input gain adjustment. A measurement can also be initiated by the input signal amplitude itself and from an external pulse, as explained in Chapter 11 Trigger Functions.

Manual Control of Run/Stop

Manual control of the 3000+ to initiate and stop a measurement is via the **RUN/STOP** hardkey at the lower left of the front panel. The initial press will begin the averaging process. On the right of the display (fourth line down), the operational status will be indicated as “RUN”. The run time display on the right end of the first line will indicate the number of seconds which have elapsed since the averaging was initiated.

Continuously Running Time Averaging

If the averaging type has been selected to be Linear Repeat, Exponential, BT/Exp, BT/Lin or Count Repeat, the averaging will continue until the **RUN/STOP** is pressed again. This will pause the averaging, the run time will cease to increase, and the operational status will change to “STOP”. Should **RUN/STOP** be pressed again, averaging will recommence, the operational status will return to “RUN” and the run time will pick up the count from the time displayed when the averaging had been previously paused. If in Standard Mode, the Max, Min, Leq, and SEL data will include contributions from all signals applied to the inputs during all time intervals when the operational status was RUN, beginning from the original initiation of the averaging process.

To reset the data buffer in order to begin a new averaging period containing no prior signal data, press **RESET** (**SHIFT**, then **RESET**). This will clear the data buffers, set the run time to zero, and change the operational status to “RESET”.

Caution: Do not press **SHIFT** and **OFF** simultaneously, as this will produce a hard reset and reboot of the instrument.

Finite Length Time Averaging

If the averaging type has been selected to be Linear Single, Count Single or Count Manual, and if the **RUN/STOP** key is not pressed again, the averaging will continue until such time as the selected averaging time (number of spectra) has been reached. The operational status will then change to “STOP” and the run time will display the same value as the averaging time shown on the line directly below. After completing an averaging cycle, a subsequent press of the **RUN/STOP** key will automatically reset the data buffer and begin another average. It is not necessary to manually reset the data buffer before beginning another averaging cycle. If the **RUN/STOP** key is pressed before the averaging cycle is completed, the averaging process will be paused. Pressing **RUN/STOP** again will recommence the same averaging cycle without loss of data measured during the previous time interval, and the run time will continue from the value displayed at the time of the pause.

Input Gain Control

When making a measurement, the input gain should be set such that none of the input channels are overloaded, yet the signals produce significant amplitude levels on the frequency display. When there is an overload in any of the input channels while the analyzer is running, the message

“OVER” will appear in an inverse video form near the center of the screen. When making a measurement with fairly steady signals, it is common practice to set the Averaging Type to one of the continuously running forms of averaging such as Exponential, BT/Exp or BT/Lin, using a small value of averaging time such that time variations of the signal amplitude will be clearly visible on the screen. The input gain control is then adjusted to obtain an optimum setting for the given input signals. Once a satisfactory setting has been made, the averaging time is then increased for superior accuracy.

Manual Control of Input Gain

Press the hardkey **RANGE** to put the input range under the control of the horizontal arrow keys, which will be indicated by the message “*range” on the lower right of the display. This message also indicates the full scale value of the screen.

Pressing the right horizontal arrow key increases the full scale value in 10 dB steps (decrease gain) and pressing the left horizontal arrow key decreases the full scale value (increase gain) in 10 dB steps.

The upward and downward vertical arrow keys can also be used to change the gain, although there is no indication on the lower right of the screen to indicate this function. Pressing the upward vertical arrow key increases the full scale value and pressing the downward vertical arrow key decreases the full scale value.

Offsetting Gain Between Channels

If there is a great difference between the signal levels at the different inputs, it may happen that one channel will be near overload while another will have such a low signal level as to represent only a fraction of the available dynamic range of the input. In such a case, it may be necessary to set an offset between the channels such that there is a difference between the full scale values.

In the Model 3000+, the channel 2 gain can be offset with respect to the channel 1 gain from the Input Menu by pressing **ΔRANGE [P]** which will produce the message "* Δrange XX" on the lower right side of the screen indicating that the horizontal arrow keys are now programmed to adjust the offset of channel 2 with respect to channel 1. The numerical field XX in the message indicates the amount of offset, in units of dB, presently active. Each press of the left arrow key will decrease the offset in 10 dB steps while pressing the right arrow key will increase the offset in 10 dB steps, as indicated by the changing value of XX in the message on the lower right of the screen. The offset is limited to ±30 dB. When the desired amount of offset has been selected, press **CURSOR** to remove the horizontal arrow keys from continuing to adjust the offset.

Autorange of Input Gain

A more convenient way to set the input gain is to use the autorange feature. With the analyzer running, press **AUTO**, which will produce the message "Auto Ranging is ENABLED" on the upper right of the display. This will automatically adjust the input gain until the amplitude of the peak detector for any input channel falls within the aperture of full scale without an overload. While the autoranging process is in progress, the screen message will switch between "Auto Ranging is ENABLED" and "Ranging". This switching will cease when the proper range has been achieved. Pressing **RANGE** will return the range control to the horizontal arrow keys. The 20 dB window below full scale into which the autoranging seeks to place the highest peak component is the default value of the Ranging Aperture. This value may be changed from the Input Menu by pressing **AUTO.RA [E]** and entering a new value. When the Ranging Aperture is small with respect to the variability of the input signals, the autoranging may be unable to find a stable setting. If this occurs, increase the Aperture.

The speed with which the autorange responds is related to the response of the input modules, which in turn depends upon the values of highpass filters which are active. For the fastest autoranging operation, select a frequency range having a 20 Hz lower limit.

After the autoranging process has stabilized to the proper gain setting, press **RANGE** to turn off the autoranging function and put the range under manual control

Response Time of Digital Filters

In the case of frequency analysis using digital filters, when the analysis is initiated following a **STOP**, **RESET** sequence, there is a time delay associated with the output of each filter. The lower frequency filters, which have the narrowest bandwidths, have the longest response time. Because the filter levels are not displayed until valid data are available, the upper frequency filter levels will appear before those of the lower frequency filters. Once the filters are running, however, and the measurement is stopped by pressing **RUN/STOP** without a reset, the frequency analysis function continues to run in the background. When the **RUN/STOP** key is pressed again, there will be no additional response time and the data display and calculation of parameters such as Leq will be resumed immediately.

Possible Overload Indication upon Resuming Analysis

Suppose that during the period when the measurement is paused between presses of the **RUN/STOP** key, as explained above, an input signal capable of overloading the analyzer occurs. Since the display is not running, the overload indication cannot appear at that instant. But because the overload indication is a latching one, the overload indication will appear when the **RUN/STOP** key is pressed to resume the analysis. In such an instance, it could be puzzling to the user to see an overload indication upon resuming the measurement when it is known that the input signal at that instant was not sufficient to produce an overload. it is, rather, an indication that an overload did occur sometime during the time interval when the measurement was paused, but the analysis function was still running.

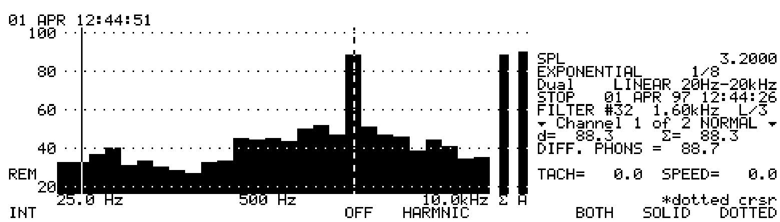
CHAPTER

8 Cursor Control

The cursors are controlled by the horizontal arrow softkeys at the lower right of the front panel. The hardkey **CURSOR** is used to place the cursors under control of these keys. The manner of which this is done depends on the state of the horizontal arrow keys at the time. If these keys have been assigned to perform some function other than cursor movement, such as controlling the gain, then pressing **CURSOR** will assign the horizontal arrow keys to control the movement of which ever cursor (dotted or solid) was last active. A second press of **CURSOR** will produce the display of the Cursor Menu, shown in Figure 8-1.

If the cursor had already been under control of the horizontal arrow keys, then a single press of **CURSOR** will produce the Cursor Menu.

Figure 8- 1 ***Cursor Menu***



Solid and Dotted Cursors Moving Independently

There are two cursors available, a solid cursor and a dotted cursor. To have the horizontal arrow keys control the position of either of these, press one of the following keys:

Softkeys	Softkey Functions
SOLID [O]	Solid Cursor Control by horizontal arrow keys
DOTTED [P]	Dotted Cursor Control by horizontal arrow keys

When one of these is selected, the message “*solid crsr” or “*dotted crsr” will appear on the lower right of the display to indicate which is active. The frequency corresponding to the active cursor position will be displayed on the right of the screen fifth line down. The amplitude value corresponding to the frequency (time) position of the active cursor will be displayed on the lower right of the display (seventh line down) just below the channel number indication. The letter “s” indicates that the solid cursor is being read and the letter “d” indicates that the dotted cursor is being read.

The two vertical bars at the right of the display represent the Summation and A-Weighted broadband levels, designated by the symbol for summation “S” and the letter “A” below, respectively.

Solid and Dotted Cursors Moving Together

If the key **BOTH [N]** is pressed, the two cursors will move together under control of the horizontal arrow keys, maintaining a constant spacing between them as they move across the display. The message will be “both crsrs”. The frequency readout (fifth line down) will be indicated by the symbol “Δ” and the values of the frequencies corresponding to both solid and dotted cursors will be indicated in the form of "dotted freq. - solid freq.".

The amplitude readout, indicated by the symbol “Δ” on the seventh line down, will display the difference between the amplitude corresponding to the solid cursor and the amplitude corresponding to the dotted cursor.

The displayed broadband levels representing the Summation and A-Weighted Levels will now represent the energy between the two cursor frequencies rather than between the analog input filters. With a displayed spectrum which is not uniform, note that moving the two cursors together across the screen will result in amplitude variations of these two broadband levels as more or less energy falls between them.

Harmonic Cursors

When FFT filtering is being used and the horizontal frequency axis is linear, it is possible to use the display to investigate possible harmonic relationships between peaks in frequency domain functions. Place the active cursor (solid or dotted) at a frequency which might be the fundamental frequency of a series of harmonically related spectral peaks, and from the Cursor Menu press **HARMNIC [L]**. Superimposed upon the spectral data will be a series of very finely dotted vertical lines, each located at one of the frequencies representing a harmonic (integer multiple of the fundamental frequency). Note that the horizontal arrow keys continue to control the active cursor. With the Harmonic Cursors active, shift the position of the active cursor and note that the harmonic cursors follow in order to maintain their relative positions at harmonic frequencies. To turn off the harmonic cursors, simply press **HARMNIC [L]** a second time. Repeated pressing **HARMNIC [L]** toggles the harmonic cursors on and off.

When examining a spectrum which does indeed contain a number of peaks which are harmonically related, moving the cursor until there is good alignment between the harmonic cursors and these spectral peaks is a good way to accurately determine the fundamental frequency of the harmonic series. The higher harmonics are very sensitive to slight changes in the value of the fundamental frequency, so small cursor movements which seem to have a negligible effect upon the position of the cursor with respect to the fundamental frequency will produce large displacements of these higher harmonic cursors.

Fixing Cursor Positions

Pressing **OFF [K]** will fix the cursor positions on the screen, essentially disengaging the horizontal arrow keys from controlling the cursors without assigning it to another role. This will be accompanied by the message “*OFF” on the lower right of the screen. Pressing **BOTH [N], SOLID [O]** or **DOTTED [[P]** will reassign the cursor to horizontal arrow key control.

Pressing **EXIT** exits from the Cursor Menu to whichever menu had been previously displayed.

CHAPTER

9 Selection of Units and Calibration

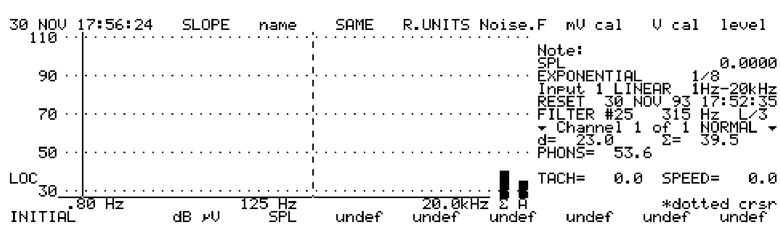
The input to each channel of the 3000+ will in all cases be a voltage signal from a transducer. The role of calibration is to relate this signal to the sensitivity of the transducer, and if required, to perform integration or differentiation of the signal and modify the units of the data output to those desired by the user.

Units

Accessing Units Menu

From the System Menu, access the Units Menu by pressing **UNITS [F]**, which will produce the display shown in Figure 9-1.

Figure 9- 1 ***Units Menu***



Creation of Unit Names

The first step in calibration is to define the name of the measurement unit which is to be used for each channel. The row of softkey labels at the bottom of the display, keys **[I] - [P]**, presents the choice of unit names. When delivered, the only keys representing actual unit names are **dB μV [I]** and **SPL [J]**; the remaining keys names are labeled “undef” for UNDEFINED. The user can create names to associate with the remainder of these keys, such as g, m/sec, ft/sec2, psi, mil, etc. as required. Once created, unless the user changes them or clears them, these names will remain among the choices available.

To name (attach a label to) a softkey, press **name [B]** and respond to the prompt “Push units to name” by pressing one of the softkeys **[K] - [P]**. The message “Enter setup name” followed by a flashing cursor prompts the user to type a setup name of up to 7 characters and press **EXIT.** The newly created label will now appear above the designated softkey. Unless this label is changed or the set of user-defined labels is reset, it will remain active in the Units Menu. The softkeys **dB μV [I]** and **SPL [J]** cannot be changed. If the key **R.UNITS [D]** is pressed, the labels for all the user definable softkeys will be reset to UNDEFINED.

Assignment of Unit Names

The unit names to be assigned to the individual channels are selected from among the choices of unit names represented by the softkeys along the bottom of the screen. Select a particular channel by pressing either **CH1** or **CH2** hardkeys. The displayed channel number will be indicated on the right side of the display (sixth line down). With only one channel active, Channel 1 is automatically selected. Press the softkey having the desired units name as its label to assign that name to the selected channel. Note that the name now assigned to that channel is displayed on the first line of the upper right of the display.

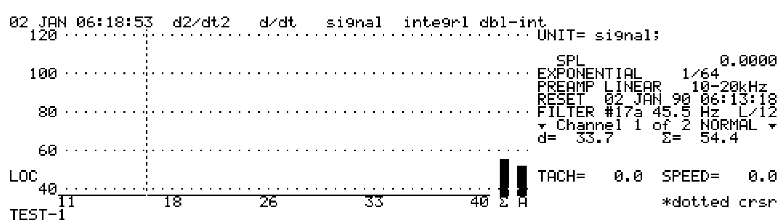
For dual channel applications, if the channels are to have different units, select the channels one at a time, assigning a name to each as described above.

If both channels are to have the same unit names, after assigning a name to one of the channels press **SAME [C]** and that same name will be assigned to the other channels.

Assignment of Integration or Differentiation

Use the numerical keypad to select the channel which is to have digital integration (single or double) or differentiation (single or double) applied to its signal. Press **SLOPE [A]** to access the Slope Menu shown in Figure 9-2.

Figure 9- 2 ***Slope Menu***



Press one of the following keys to define the integration or differentiation which is desired.

Softkeys	Softkey Functions
d2/dt2 [A]	Double Differentiation
d/dt [B]	Single Differentiation
signal [C]	No Differentiation or Integration
integrl [D]	Single Integration
dbl-int [E]	Double Integration

When the selection is made, the choice is indicated by a symbol in the first line on the right of the display.

Each digital integration is performed by dividing the level at each frequency band by $2 \cdot \pi \cdot f$, where f is the center frequency of the filter; and each digital differentiation is performed by multiplying the level at each frequency by this same factor.

1/1 and 1/3 Octave Integration and Differentiation Operations

When using the 1/1 and 1/3 octave filters, the integration and differentiation operations are purely display functions. This means that when single or double integration has been selected, the displayed spectrum will include the effect of this operation but the spectrum which is stored is neither integrated nor differentiated. Should that stored spectrum be recalled without integration or differentiation selected, there will be no integration or differentiation of the displayed spectrum. However, if the user selects an integration or differentiation operation previous to the recall operation, that function will be represented in the displayed spectrum. Note that the effect of an integration, division by $2 \cdot \pi \cdot f$, is to decrease the levels at all frequencies above 1 radian/sec. For example, the level at 1 kHz will be reduced by 76 dB, which could cause the resulting spectrum to drop below the bottom of the screen. It is often necessary to utilize the vertical offset function, described in the Section “Control of the Vertical Display” in Chapter 19 to bring the integrated spectrum back up onto the screen.

FFT Integration and Differentiation Operations

Due to the large number of spectral lines used with FFT analysis, the integration and differentiation operations are performed in the DSP processor, and are therefore not a display function. This means that the integration and differentiation operations are performed as part of the measurement and are represented in the spectrum previous to display.

The integration operations described here can be applied to FFT spectra measured using either the Standard or the Cross Analysis Modes, since the effect of the integration on phase between channels has been taken into consideration. However, due to limitations associated with the number of available bits of resolution, the integrated levels of FFT spectra are only valid for frequencies above 1% of the full scale frequency. For frequency bands at frequencies below 1% of full scale, the integration operation is not performed and the levels will not be altered. For applications where phase is not of concern and the spectra are measured using the Standard Analysis Mode, single and double integration can be performed as a display weighting function from the Digital Display Weighting Menu discussed in Chapter 10. These integrations are valid over the complete frequency range.

Upon storage, the integrated or differentiated spectrum is stored; and upon recall, without selecting integration or differentiation, the displayed spectrum will already include the integration or differentiation operation which was performed at the time of measurement and storage. Furthermore, if integration or differentiation had been selected previous to the recall, the displayed spectrum would be the same one which was stored, regardless of the fact that single or double integration or differentiation symbols are shown on the screen. Simply stated, when working with FFT spectra, the integration and differentiation operations are performed on the measurement, but not upon data recalled from memory.

Because of this, it is important that the user store the units used for the measurement along with the data. This way, by recalling the stored units at the time of recalling the data block, any integration or differentiation operations which had been performed at the time of the measurement will be indicated on the right of the screen.

Calibration

Calibration is done one input at a time. Select the input to be calibrated using the numeric keypad, then follow the appropriate calibration procedure described in the following sections.

Calibration Based on a Transducer Sensitivity Value

When a calibrator is not available, the analyzer can be calibrated by using a known value of the sensitivity of the transducer and signal conditioning system. There are two alternative procedures which may be used with the Model

3000+, depending upon whether the user wishes to express the data in logarithmic units (dB) or linear units.

Logarithmic Units Calibration (dB/Volt)

After a unit name has been assigned to the channel to be calibrated, the desired integration or differentiation defined, and that channel has been selected using the numerical keypad, press **V cal [G]**. The message “Enter dB/Volt XXX” on the right of the display will prompt the user for a numerical entry. Use the keypad to type the sensitivity in dB/Volt and press **EXIT**.

Example: The microphone being used has a sensitivity of 50 mV/Pa. No differentiation or integration is required for a sound pressure level measurement. To have an output of 1 V from this microphone, it would have to be exposed to a sound pressure of 20 Pa (e.g. 50 mv/Pa X 20 Pa = 1V). The sound pressure level, Lp, corresponding to this is calculated as follows:

$$L_p = 20 \log_{10} [P/P_{ref}] = 20 \log_{10}[20/20\times10^{-6}]$$

$$L_p = 120 \text{ dB}$$

Enter this value as the calibration level.

Logarithmic Units Calibration Microphone K-factor

A microphone which has a sensitivity of exactly 50 mV/Pa will have a logarithmic sensitivity of 120 dB/Volt. The open-circuit K-factor, K_0, is an indication of the degree to which the sensitivity of a microphone varies from this 50 mV/PA reference value. Thus when performing a calibration using decibel units, the sensitivity can be calculated from the K-factor provided on the microphone calibration chart as follows:

$$\text{dB/Volt} = 120 + K_0$$

When using Larson Davis analyzers with Larson Davis microphone preamplifiers, the system sensitivity is relatively independent of the length of the microphone extension cable up to lengths exceeding 50 feet. However, when using extension cables longer than that, it is best to correct the open circuit sensitivity of the microphone for the effect of the cable lengths when performing the calibration. Contact Larson Davis for further information.

Linear Units Calibration

When using a transducer such as an accelerometer, most users prefer to measure in units such as “g”, m/s_2, m/s, etc. rather than decibels. Since the default unit is decibels, change the scale by pressing **SHIFT**, **Y-AXIS [B]** followed by **LIN/LIN [A]** or **LIN/LOG [C]** depending upon whether the scaling of the vertical axis is to be linear or logarithmic. The uses of different units is described in more detail in Chapter 19, Control of Display Formats. After a unit name has been assigned to the channel to be calibrated, the desired integration or differentiation defined, and that channel has been selected using the keypad, press **mV cal [F]**.

The message “Enter mV/unit XXX” will prompt the user for a numerical entry. Use the keypad to type the sensitivity in mV/unit and press **EXIT**.

Example: With an accelerometer having a sensitivity of χmV/g and wishing to have the instrument read in units of g, select the name G for the units name. No differentiation or integration is required and the value of sensitivity to be entered will be χ.

The sensitivity value to be entered must correspond to the units name assigned to that channel. When the units name is the same as the units utilized in expressing the sensitivity of the transducer, as in the example above, this is straight-forward. However, when the named units are to be different from those used to express the transducer sensitivity, the entered sensitivity must be properly scaled to represent the named units.

Continuing with the above example, suppose one wishes to read the data in units of ft./sec ^{2}. First create the units name FT/SEC2 and assign it to the channel to be calibrated. Since 1.0 g = 32.2 ft/s^2, we can always multiply the given sensitivity by $\frac{1/32.2g}{ft/s^2}$ because it has a value of unity. Thus, we calculate the sensitivity/unit as follows:

$$\underbrace{50}_{\text{sensitivity value}} {}^{mV}/_{g} \times \frac{1/32.2g}{ft/s^2} = 1.553\frac{mV}{ft/s^2}$$

If we wished to express the data as a velocity, in units of ft/s, we would assign the units name of FT/SEC to that channel.

We would select to single integrate the signal to obtain velocity and would still use 1.553 as the value of mV/unit because the single integration would transform the ft/s^2 to ft/s. If we wanted the velocity expressed in in/s, the proper sensitivity in mV/unit would be calculated as follows:

$$\underbrace{\frac{1.553mV}{ft/s}}_{\text{sensitivity value}} \times \frac{1/12ft}{in} = 0.129\frac{mv}{in/s}$$

Calibration Based on a Reference Signal

The preferred method of calibration is to apply a known excitation level to the transducer and calibrate the analyzer to that value. The advantage in using this method is that it verifies the integrity of the transducer and the cable connecting it to the input module. This is common practice when using precision condenser microphones and is becoming increasingly common with accelerometers as well. Many acoustics professionals will first calibrate the system using the K-factor of the microphone. They will then use a sound level calibrator as described below to verify the K-factor calibration. An error greater than a few tenths of a decibel could be an indication of a faulty microphone, preamplifier, or cable.

When using a sound level calibrator, the known excitation level will be in decibels so the vertical scale should be logarithmic. While applying the known excitation to the transducer, perform a measurement and stop the analysis with the measured spectrum displayed on the screen (when using an acoustic calibrator, select the Long 1/3 Octave filters). Move the cursor to the frequency of the excitation, then press **level [H]**. The message “Enter Level XXX” on the right of the display will prompt the user for a numerical entry. Type the known amplitude of the excitation via the keypad and press **EXIT**. If using a Larson Davis Model CAL250 sound level calibrator, the excitation frequency will be 250 Hz, the calibration level will be 114 dB, and no differentiation or integration is necessary.

When using an accelerometer calibrator, the known excitation will typically be in units of “g” or m/s^2, so before performing the calibration set the vertical axis to an appropriate scale as described in the previous section.

Calibration Using the Test Signal

When the sensitivity of the transducer is known, the 1 kHz square wave test signal can be used to calibrate the 3000+. This procedure is simpler than that described above. Simply calculate the excitation amplitude to the transducer which would be required to produce an output of 1Volt. Perform a measurement with the test signal ON, stop the measurement and move the cursor to the 1 kHz frequency band representing the fundamental frequency. Press **level [H]**, enter the calculated amplitude value using the keypad, and press **EXIT**.

Storage and Recall of Units Information

It is important to realize that the units names, integration or differentiation and calibration are display functions. As mentioned previously, the actual input to each channel is a voltage signal which is then integrated or differentiated and scaled to produce the measurement units as defined by the user in the "Units" menu. Because these are purely display functions, when the data are stored they are stored in the same pure voltage form as measured, without the integration/differentiation and scaling operations which were performed as part of the display function.

For example, suppose the transducer were an accelerometer and the user wished to see the data in units of inch/sec. The actual measurement would produce a voltage proportional to the acceleration. The user would use the Units Menu to name the units IN/S, invoke a single integration to obtain velocity from acceleration, and then use the calibration to obtain the proper scaling. The result would be a display of a velocity spectrum, in inch/second units, as a function of frequency.

Let us further suppose that this measurement is stored and in the meantime the units are changed such that there is no integration or differentiation, and the scaling is changed. Upon recall, the spectrum which will be seen will be in the form of acceleration versus frequency, and the magnitude will be whatever corresponds to the presently active units calibration. To obtain the results which had previously been displayed at the time of the measurement, the same Units setup must be employed. Thus, it is recommended that the Units be stored in addition to the measured data so they can be recalled and used when the data are recalled and displayed.

Storage of Units Data

To store the complete set of Units softkeys which have been created, press **STORE**. The message "STORE - Units Data N" on the upper right of the screen indicates that this set of Units softkeys have been stored to the active memory files as the Nth record of Type "Units Data".

Recall of Units Data

To recall a set of Units Data from the active memory file, press **RECALL**. The message "Over Write ALL UNIT data?" on the upper right of the screen indicates that the present set of Units will be lost if the recall is continued. Press **YES [A]** to continue the recall, and press **NO [C]** to abort the recall operation.

If the recall is continued, the message "RECALL - Units Data N" on the upper right of the screen indicates that the Nth record of the Type Units Data has been recalled, and the softkey labels will change to reflect those in the stored record.

The message "*recall data" on the lower right of the screen indicates that the horizontal arrow keys are assigned to control the recall of Units Data records from the active memory file. Pressing these keys permit the user to select the particular record number which contains the desired set of Units Data. Reassign the horizontal arrow keys to control the cursor in order to prevent the recall of other records.

Frequency Weighting and Integration of FFT Spectra

Accessing the Display Menu

The Display Menu is accessed from either the Standard Analysis Menu or the Intensity Analysis Menu by pressing **DISPLAY [F]**. The display functions are not available for use with the Cross Analysis Mode. The resulting Menu will resemble either Figure 10-1 or Figure 10-2, depending upon whether the octave filters or the FFT analysis is active.

Figure 10- 1***Display Menu (octave bandwidths)***

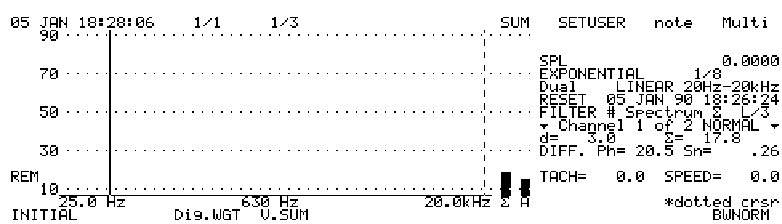
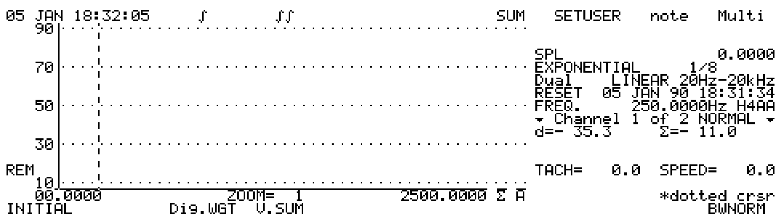


Figure 10-2 ***Display Menu (Standard Mode with FFT)***



Selecting Bandwidth for Display of 1/3 Octaves

When the measurement has been made using 1/3 octave filters, the display may be presented in either 1/1 or 1/3 octave format. The user makes this selection by pressing one of the following softkeys, **[1/1] A** or **[1/3] B**, as shown in Figure 10-1. The 1/1 octave levels are obtained by summing the three 1/3 octave levels contained within each 1/1 octave band.

Selecting Integration

When the instrument is in the Standard Analysis Mode performing FFT frequency analysis, the softkeys **[A]** and **[B]** invoke single and double integration, respectively, as shown in Figure 10-2. When either the single or double integration have been selected, this will be indicated on the right side of the screen, first line down. Each integration is performed by dividing the level in each frequency band by $2\pi f$. This means that the levels for all bands at frequencies greater than 1 Hz will be reduced in value. As a result, many of the spectrum levels previously visible when the non-integrated spectrum was being displayed may disappear below the minimum axis of the display following integration. Use the Vertical Offset function in the Shift Menu to bring the displayed levels back up to within the range of levels being displayed.

Single and Double Integration of FFT spectra can also be invoked from the Units Menu as described in Chapter 9. However, due to limitations related to the number of bits of resolution, that method of integration is invalid for frequencies below 1% of the full scale frequency. The method utilized here, while restricted to FFT spectra measured using the Standard Analysis Modes, is valid over the complete frequency range.

The integration of octave bandwidth spectra is invoked from the Units Menu as described in Chapter 9.

Digital Display Weighting

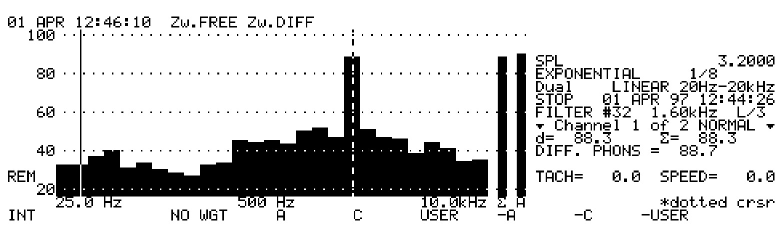
The display weighting is digital and independent from the input signal analog weighting selected from the input menu. Its effect is to weight the spectral data before presentation on the display. Display weighting is not available when the Cross Analysis Mode is active. The Model 3000+ offers two types of digital display weighting:

- Broadband Acoustic Weighting corresponding to the A and C filters
- User-defined Weighting

Accessing the Digital Weighting Menu

From the Display Menu, access the Digital Weighting Menu, shown in Figure 10-3, by pressing **Dig. Wgt [I]**.

Figure 10- 3 ***Digital Display Weighting***



Note that the softkeys associated with Zwicker Loudness on the upper left only appear when 1/3 octave filters are active. The use of Zwicker Loudness function is discussed in Chapter 6, Analysis Menus; Selection of Measurement and Display Parameters.

The A and C weighting curves are defined by the sound level meter standards IEC 60651 and ANSI S1.4 1983. The user can select to weight the displayed spectrum by positive or negative versions of either the A or the C curve. When recording environmental noise using a sound level meter with the AC output connected to a recorder, some users like to A or C-weight the AC output. This tends to increase the measurement range of the recording because environmental noise tends to have large levels at frequencies below the human hearing range which are not of interest. When playing back such a recording for analysis, the use of the negative weightings will produce an unweighted spectrum display.

In other applications, such as the analysis of hand/arm or whole body vibration, users wish to weight the spectrum by a user-defined weighting spectrum. The procedure for setting up and storing user-defined weightings is described in the following section. The default state is No Weighting.

The desired digital display weighting is selected as follows:

Softkeys	Softkey Functions
NO WGT [I]	No Weighting
A [J]	A-Weighting
C [K]	C-Weighting
USER [L]	User Defined Weighting
-A [M]	-A-Weighting (negative)
-C [N]	-C-Weighting (negative)
-USER [O]	-User Defined Weighting (negative)
Zw.FREE	Free-field Loudness
Zw.DIFF	Diffuse-field Loudness

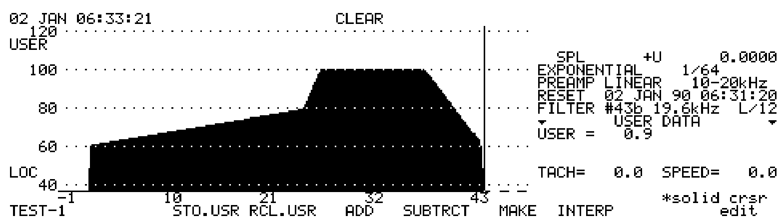
Exiting From Display Weighting

Press **EXIT** to exit from the Display Weighting Menu back to the active Analysis Menu.

User Weighting

Creation, storing, recalling and manipulation of user weighting curves are performed in the Setuser Menu, Figure 10-4, accessed from the Display Weighting Menu by pressing **SETUSER [F]**

Figure 10- 4 ***Setuser Menu***



Creating a User Weighting Curve

To create a user weighting curve, first press **CLEAR [C]** to ensure that the working buffer is zeroed.

For each bandwidth which is to have a non-zero value, move the cursor to that band, press **edit [P]** and in response to the prompt on the upper right of the display, use the keypad to type in the desired value, then press **EXIT**. The entered value will be displayed upon the screen and the cursor will move to the next filter band. It is necessary to press **edit [P]** each time before entering a value for a frequency band. However, the numerical value in the entry window when it opens is the same as the value previously entered, so a horizontal weighting line can be entered by repeatedly pressing the key sequence **edit [P]**, **EXIT**.

Interpolation Function

Rather than enter the amplitude of each band one-by-one, one may choose to define a section of the user weighting curve where the amplitude values of successive bands are to fall along a straight line (values are interpolated from a straight line). To do this, move the cursor to the frequency band which represents the low frequency limit of the section and press **INTERP [N]**. The message “AUTO INTERPOLATION IS ON” will appear on the upper right of the display. Press **edit [P]** and use the keypad to type in the amplitude for this frequency band, and press **EXIT**. Now move the cursor to the frequency band which represents the high frequency limit of the section to be defined, press **edit [P]**, type the amplitude value for that band and press **EXIT**. The amplitudes of all bands between these two bands will fall along a straight line drawn between them. If the cursor is now moved to a higher frequency and an amplitude entered, another straight line interpolated section will be defined because the interpolation function is still ON. To turn the interpolation function OFF, simply press **INTERP [N]** a second time.

Creating a User Weighting Curve from a Measured Spectrum

After a spectrum has been measured and it is being displayed, it can be made into a user weighting curve by pressing **MAKE [M]**. At the right of the display there will be a prompt “MAKE USER, ZERO AT +000.0”, which permits the user to add a dc offset in creating the user weighting curve from the spectrum. Use the keypad to enter the desired offset, then press **EXIT**. For no offset, simply press **EXIT** directly without typing a value.

The Active Register

There is an active register associated with user weighting. This register holds one user weighting curve for each of the following bandwidths; 1/1 octave, 1/3 octave, and FFT, which together make up a set of user curves. Whenever the Setuser Menu is accessed, the user weighting curve in the active register corresponding to the present analyzer bandwidth will be displayed. When the user clears, creates and edits a user curve as described in the preceding section, he is modifying the user weighting in the active register associated with that bandwidth.

When originally accessing the Setuser Menu, the user weighting curve displayed is whatever happened to be in the active file for that bandwidth at that time. This is why the clear operation is recommended before creating a new user curve. The clear operation only clears the user curve in the active register which corresponds to the active bandwidth, not the entire set of user curves.

Suppose that the 3000+ is configured to a 1/1 octave bandwidth, the Setuser Menu is used to create a user curve, and without any specific storage operation the user exits from the Setuser Menu, reconfigures the 3000+ to a 1/3 octave bandwidth, accesses the Setuser Menu and creates a user curve. Then, he exits from the Setuser Menu, reconfigures for an FFT bandwidth, accesses the Setuser Menu and creates a user curve. Although no storage operation has been performed, all three of these user curves, or a set, are contained in the active register. Should the user exit from the Setuser Menu and reconfigure to another bandwidth, upon returning to the Setuser Menu the user weighting curve which is in the active register for that bandwidth will be displayed.

Storing the Active Register into Storage Registers

There are 4 nonvolatile storage registers available into which the set of user weighting curves in the active register can be stored. This is done by pressing **STR.USR [I]**. The message on the upper right of the screen,

"USER # (1-4) TO STORE   X"

with a flashing cursor on the X prompts the user to input a register number between 1-4 using the numeric keypad and press **EXIT**. The data in these registers will remain intact unless the nonvolatile memory is reset or the data in the registers are overwritten by recall of a User Curve record, as described later in this chapter. Since the active register contains a complete set of user weighting curves, one each for 1/1, 1/3, and FFT bandwidths, the storage register will therefore contain these same four user curves.

Recalling from Storage Registers

To recall a set of user weighting curves from a storage register back into the active register, press **RCL.USR [J]**. The message on the upper right of the screen,

"USER # (1-4) TO RECALL X"

with a flashing cursor on the X prompts the user to input the number of the storage register whose curves are recalled, using the numeric keypad, and press **EXIT**. Since these are recalled into the active register, the user weighting curve based on the presently active bandwidth which was in the storage register will be displayed.

Adding Registers

The user can add the set of user curves in any of the four storage registers to the set in the active register by pressing **ADD [K]**. The message

"USER # (1-4) TO ADD? N"

prompts the user to input a storage register number using the numeric keypad and press **EXIT**. The user weighting curve resulting from the addition is displayed. User curves from the same bandwidths are added together, so each of the four bandwidth user curves in the active register reflect the result of the addition process.

Subtracting Registers

The user can subtract the set of user curves in any of the four storage registers from the set in the active register by pressing **SUBTRCT [L]**. The message

"USER # (1-4) TO SUBTRCT? N"

prompts the user to input a storage register number using the numeric keypad and press **EXIT**. The user weighting curve resulting from the subtraction is displayed. User curves from the same bandwidths are subtracted, so each of the four bandwidth user curves in the active register reflect the result of the subtraction operation.

Storage of User Curve Records

Pressing the hardkey **STORE** will result in all the user weighting curves (15) contained in the active register and the four storage registers being stored into a single record in the active memory file whose name is displayed on the lower left of the screen. The message "STORE - User Curves N" on the right of the screen indicates that these curves have been stored as the Nth record of the type "User Curves" into the active file. Like any other record, a note field may be added to the record previous to storage, and the note field can be edited from the Files Menu as described in Chapter 13, Record Operations from the Files Menu. And like any other record, User Curves can be transferred from memory to the floppy disk and recalled back into memory from the floppy disk.

Recall of User Curves

To recall a User Curve record from the active memory file, press **RECALL**. The message "RECALL - User Curves N" on the upper right of the screen indicates that the Nth record of the type "User Curves" has been recalled. The contents of the active user weighting register and the four storage registers will now contain the user weighting curves which had been recalled from Nth User Curve record. The curve from the active register corresponding to the analyzer bandwidth will be displayed.

The message "*recall data" on the lower right of the screen indicates that the horizontal arrow keys are assigned to control the recall of records from within the file. Presses of these keys permits the user to page backwards and forwards through the sequential User Weighting records to find the one which is desired. As this paging takes place, the displayed user curve will change to reflect the user curve of that bandwidth stored in the newly recalled User Record.

The user may now recall a set of user weighting curves from any one of the four storage registers into the active register as described earlier in this section.

Exiting from the Setuser Menu

Pressing the **EXIT** key will exit from the Setuser Menu back to the Display Weighting Menu. The display weighting status will be whatever it had been when the Setmenu was accessed. Thus, if it had been something other than USER, it will be necessary to change to USER or -USER before the newly created user weighting curves will have an influence on the displayed data. If USER or -USER had been active at the time of accessing the Setuser Menu, the effect of the newly created user weighting curve will be immediately apparent.

A second press of **EXIT** will exit from the Display Weighting Menu to the active Analysis Menu.

CHAPTER

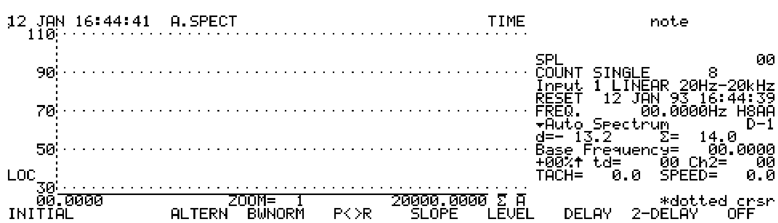
11 Trigger Functions

The 3000+ can be put into a mode whereby signal averaging is initiated by the satisfaction of certain trigger criteria. There are two types of trigger functions available: time-domain triggering (for use with FFT filtering only), and frequency-domain triggering.

Time-domain Triggering

Time-domain triggering requires that FFT filtering be selected for use with Cross Analysis and that Count Averaging be active. As the name implies, the Time Trigger is based upon the input signal as sampled in the time domain (e.g. the value of the digitized sample measured by the analog/digital converter). Only the signal applied to Channel 1 can be used for the Time Trigger. The Time Trigger Menu, shown in Figure 11-1 is accessed from the Cross Menu by pressing **T.TRIG [M]**.

Figure 11- 1 ***Time Trigger Menu***



The first two trigger criteria are the level and slope of the signal, as indicated by the expression “xx%” displayed on the right of the screen. The trigger level is in percent of full scale, and the slope is indicated by the positive or negative direction of the arrow.

Trigger Level

To increase or decrease the level, press **LEVEL [M]**. The message "*trig.level" will appear at the lower right indicating that the horizontal arrow keys are to be used in setting the level. As these keys are rotated, the change in level can be seen on the display.

Trigger Slope

The slope is toggled between positive and negative by pressing **SLOPE [L]**.

Trigger Delay

When the trigger function is Armed by pressing **RUN/STOP**, the sampling and storage of waveform data into the time buffers of both input channel begins. The size of the time buffer, in number of samples, depends upon the number of lines selected for the FFT analysis as follows:

# Lines	# Samples
100	256
200	512
400	1,024
800	2, 048

However, once the buffers are full, instead of transferring the contents to the FFT processor, as is usually done in FFT analysis, the data is shifted through the buffer sample by sample, the oldest data being lost as new data points enter. Once full, each buffer will at any instant hold a full set of samples representing the time waveform over the preceding time interval required to fill the buffer (e.g. using 400 line and a full scale of 10 kHz; 1024 samples spaced 39 microseconds apart representing a time interval of 40 milliseconds). At the instant the trigger criteria are satisfied by the data being sampled at the input of channel 1, there are a number of different ways the analysis can proceed.

A. One could elect to save all or a selected number of the most recent samples in the time buffers, then fill the remainder of the buffers with data points sampled after the trigger. When the buffers are full, they are transferred to the FFT processor and the system then continues as in a normal FFT analysis. With this option, by saving data points already in the buffers, the system has pre-trigger information describing the analog waveforms sampled before the trigger event occurred.

B. One could elect to save none of the data points within the buffers at the instant of triggering, and begin filling the buffers anew from that instant.

C. One could elect to save none of the data points in the buffers at the instant of triggering, and also to wait a selected number of samples before beginning to fill the buffers again.

The manner in which the system deals with these possibilities is determined by the Trigger Delay.

Zero Trigger Delay represents saving none of the previously measured data points, as in item (B) above.

A Trigger Delay of N samples implies item (C) above, where N is the number of samples which are allowed to pass before storage into the buffers begins.

A Trigger Delay of -N samples corresponds to item (A) above, where N is the number of most recent values in the buffers which are saved at the instant of triggering.

The Trigger Delay is set by pressing **DELAY [N]**, using the horizontal arrow keys; the right arrow to increase the value of the delay, and the left arrow to decrease it.

The message “*trig. delay” on the lower right of the display indicates the role of the horizontal arrow keys. As they are pressed, the value of the Trigger Delay is displayed on the right of the screen by the message “td= xx”, where xx is the delay in number of samples.

With FFT filtering, the time between samples is calculated from the relationship BT = 1, which is the fastest rate at which statistically independent samples can be taken. Therefore, the time between samples is a function of the baseband full-scale frequency, the number of lines used for the analysis, and the zoom factor.

The following formula can be used to determine the time between samples:

$$T = \frac{Z}{2.56\ F_{fs}}$$

T = Time between samples in seconds

$\boldsymbol{F_{fs}}$ = Baseband full scale frequency (before zooming)

Z = Zoom multiplier (1 for baseband analysis)

Example: Using a full scale of 10 kHz, and no zoom:

$$T = \frac{1}{2.56 \times 10kHz} = 39\mu s$$

To determine the total delay time, multiply T by the number of delay samples.

Channel 2 Delay

The Trigger Delay as set above applies to all channels. In some measurement situations, it is desirable to delay the beginning of sample storage in the buffer for channel 2 an additional number of samples. This is the Channel 2 Delay, which is defined as a delay with respect to Channel 1. The Channel 2 Delay is set by pressing **2-DELAY [O]**, and using the horizontal arrow keys as was done to set the Time Delay. The value of the Channel 2 Delay is indicated on the screen by the message “Ch2= xx”, where xx is the delay in number of samples. This value will always be positive.

Arming and Disabling

When the parameters have been entered, and the user exits back to the Analysis Menu, the trigger parameters will remain on the lower right of the display, to indicate that the Time Trigger is active.

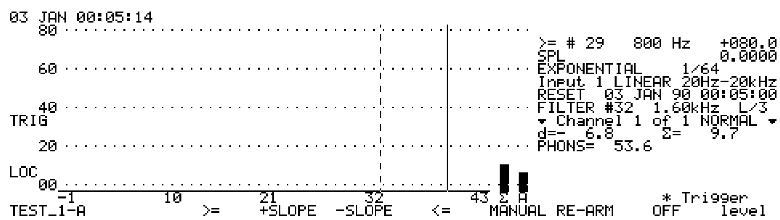
Pressing **RUN/STOP** will arm the trigger, indicated by the message “ARM” at the upper right of the display. The sampling of input data and transfer to the time buffer will begin at this point. As soon as the input signal in Channel 1 satisfies the trigger criteria, data analysis will begin. At the same time, the message on the upper right of the display will change from “ARM” to “GET”, to indicate the occurrence of triggering.

The time trigger function is turned off from the Time Trigger Menu by pressing **OFF [P]**. The time trigger parameters will then disappear from the display.

Frequency Domain Triggering

When Frequency Domain Triggering is active, the criteria used to determine when the analyzer is to be triggered into operation are the amplitude of the signal in a specified frequency band, and the slope, if desired. This method of triggering can be used with either Octave or FFT filtering (except with Count Averaging). The Frequency Trigger Menu shown in Figure 11-2 is accessed from either the Standard Analysis Menu or the Autostore Menu by pressing **F.TRIG [M]**. When doing Standard Analysis it is easiest to access this from the Standard Menu; for the other forms of analysis the Autostore Menu must be used. Since in the majority of cases the Frequency Trigger is used with Autostore, however, this is not a major inconvenience.

Figure 11-2 ***Frequency Trigger Menu***



When the Frequency Trigger Menu is first displayed, the message on the upper right of the screen will be of the following form:

[Trigger Criterion] [Trigger Frequency] [Trigger Level]

The trigger Criterion will be >=, <=, ↑, or ↓. When the frequency bandwidth is 1/1 or 1/3 octave, the band number as well as the center frequency will appear. For FFT, only the frequency will appear.

Selecting Trigger Frequency

The horizontal arrow keys are used to select the trigger frequency, as indicated by the message “*Trigger” on the lower right of the screen. The right and left horizontal arrow keys page forward and backward, respectively, through the range of available frequency values as indicated by the changing value of the trigger frequency displayed on the upper right of the screen. When paging through the frequencies, one step beyond the highest frequency is “Spectrum ∑”, meaning that the autostore will trigger based on the level of the overall level.

Selecting the Trigger Criterion

The trigger criterion and amplitude level are selected as follows:

Signal Level >= Specified Level

Press >= **[I]**, and note the message ">=" on the upper right of the display. Triggering will occur whenever the level in that band equals or exceeds the programmed value.

Signal Level <= Specified Level

Press <= **[L]**, and note the message "<=" on the upper right of the display. Triggering will occur whenever the level in that band is less than or equal to the trigger level.

Signal Level = Specified Level (positive slope)

Press **+SLOPE [J]**, and note the message "↑" on the upper side of the display. Triggering will occur whenever the signal in the specified band has both the trigger level and a positive slope.

Signal Level = Specified Level (negative slope)

Press **-SLOPE [K]** and note the message "↓" on the right side of the display. Triggering will occur whenever the signal in the specified band has both the trigger level and a negative slope.

From the Frequency Trigger Menu, the trigger level set by the user will always be displayed in decibel format at the top right of the screen, as shown in the figure depicting the Frequency Trigger Menu. However, when the vertical scale has been set to linear, upon pressing **level [P]** to input the trigger level, it will be seen from the format of the entry field that the trigger level should be input in linear units such as m/sec, g, etc. After entering this linear value, it will then be displayed in the Frequency Trigger Menu as a decibel level consistent with the calibration of the instrument.

Selecting the Trigger Level

To input the trigger level, press **level [P]** which will produce the message “TRIGGER LEVEL = XX.X” on the upper right of the screen. Type in the desired value using the numeric keypad and press **EXIT**.

Frequency Domain Trigger Setup for the SLM Mode

The previous description for establishing the trigger criteria applies when the instrument is in the Single or Dual channel Frequency Analyzer Mode. When the instrument is in the SLM Mode of operation, there is no softkey **F.TRIG [M]** available in the Main Menu. Instead, access the Autostore Menu by pressing **AUTOSTR [P]**, from which the **F.TRIG [M]** softkey is available.

When the Model 3000+ is in the SLM Mode of operation, when paging upwards through the frequency bands in order to set the Trigger Frequency, the steps beyond the highest frequency will sequentially produce the different SLM parameters prior to reaching Spectrum Â.

Arming and Disabling

The Frequency Domain Trigger is enabled as soon as the Frequency Trigger Menu is accessed and displayed, indicated by the message “TRIG” on the left axis of the screen. To disable this function, press **OFF [O]** before exiting from this Menu.

With the instrument in the Standard storage mode, pressing the **RUN/STOP** key will initiate frequency analysis, as indicated by the spectra appearing on the screen. However, spectrum averaging will not begin until the trigger criterion is satisfied. This can be verified by watching the elapsed time indication on the first line down on the right side of the screen, and noting that nothing appears until after the trigger criterion is satisfied.

With the instrument in the byTime autostore mode, pressing the **RUN/STOP** key will initiate spectrum analysis as indicated by the display of spectra along with the message “AUTO STORE IN PROGRESS” on the upper right of the screen. However, spectrum averaging and autostorage will not begin until the trigger criterion is satisfied. The use of the Frequency Domain Trigger with the byTime autostore operation is useful for the capture of data automatically based upon an even which may occur when the instrument is left unattended. Multiple events can be stored using the Automatic Re-Arming function described below.

When the 3000+ has been set in the autostore mode with a frequency trigger established and armed by pressing the **RUN/STOP** key and no trigger has occurred, a subsequent press **RUN/STOP** will disarm the frequency trigger and the message “No Data Stored” will appear on the upper right of the display. Continued presses of the **RUN/STOP** key will simply toggle the 3000+ between the armed and disarmed states.

Automatic Re-Arming

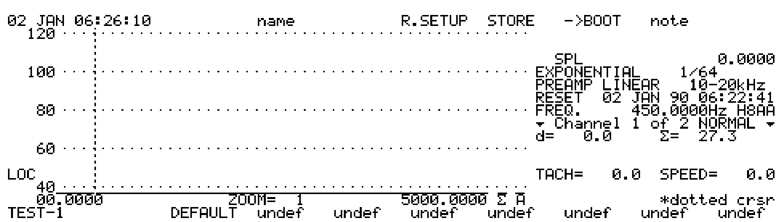
If one wishes the 3000+ to reset itself following a triggered autostore sequence, and rearm the Frequency Trigger to be ready to autostore following a subsequent event, from the Frequency Trigger Menu, before selecting the trigger criteria, press **RE-ARM [N]**. This key toggles the Rearm function on and off, as indicated by the message “Autostore rearm mode set” or “Autostore rearm mode off” on the upper right of the display.

CHAPTER

12 Storage and Recall of Instrument Setups

When the 3000+ is first delivered, there will be one default instrument setup to which the unit will configure whenever it is turned on and the software is booted up. The Instrument Setup Menu, shown in Figure 12-1 is accessed from the System Menu by pressing **SETUP [N]**, permitting the user to define a number of different Setups. The Menu can be used to change the existing instrument Setup to one of the stored Setups, and also to define to which of the stored Setups the unit will be configured when it is next booted up.

Figure 12- 1 ***Setup Menu***



The softkeys along the bottom of the display represent the eight possible instrument setups which can be defined at one time. The one represented by the key **DEFAULT [I]** is a default setup delivered from the factory and it cannot be changed by the user. The remaining ones will originally be labeled **undef** for undefined. The user can create a particular instrument setup and store it under a particular softkey labeled with an appropriate name.

Labeling and Assigning Softkeys

When the 3000+ has been configured as desired, first assign a name or label to the softkey to be used for the setup by pressing **name [B]**, then the choice of softkeys **[J]** to **[P]**. In response to the prompt on the upper right of the display, type in the desired name using the keypad, and press **EXIT**. If there is already a setup name displayed which is to be changed, press **SHIFT**, followed by **CLEAR** before typing in the new name. The name will now appear as the softkey label. To assign the present instrument setup to the softkey, press **STORE [E]**. The message "PUSH SETUP TO STORE" on the upper right of the screen will prompt the user to press the Setup softkey with the appropriate label for this setup.

Changing 3000+ Setup from Softkeys

Normally the 3000+ will boot up to its default setup as defined at the factory during production. To reconfigure the 3000+ to one of the user-defined setups created as described above, access the Setup Menu by pressing **SETUP [N],** and press the user-defined (and labeled) softkey (**[J] - [P]**) which represents the desired analyzer setup. The 3000+ will immediately be reconfigured as specified by the user-defined setup which that softkey represents.

If it is desired that the 3000+ boot up directly to one of the user-defined setups instead of to the default setup, press **>BOOT [F]** and in response to the message "PUSH SETUP FOR ATTN BOOT" press the softkey which represents the analyzer setup which is to be active after the analyzer boots up when turned on.

Reset of User-defined Setups

Pressing the softkey **R.SETUP [D]** will produce the message "*ARE YOU SURE?* on the upper right of the screen. To continue with the reset press **YES [A]**. To abort the reset, press **NO [C]**. After a reset, the labels on all the user-defined softkeys will return to **undef**.

Storage of User-defined Setups

The entire set of user-defined setups, including the softkey labels, will be stored as a record to the active memory file whose name is displayed on the lower left of the screen by pressing **STORE**. The message “STORE Setups   N” on the upper right of the screen indicates that they have been stored into the Nth record of type “Setups” in the active memory file.

Recall of User-defined Setups

Pressing **RECALL** from the Setup Menu will result in the recall of a set of User-defined setups and softkey labels from the active memory file. The message “Overwrite ALL SETUPS?” on the upper right of the screen warns the user that the recall will result in the loss of the user-defined setups presently active in the Setup Menu. Press **YES [A]** to continue with the recall. Press **NO [C]** to abort the recall operation. The message “RECALL - Setups   N” on the upper right indicates that the Nth Setup record from the active memory file has been recalled. The message “*recall data” on the lower right of the screen indicates that the horizontal arrow keys can be used to page through the Setup records available within the file. As each record is recalled, the set of softkey labels will change to correspond with the data in that record.

Exiting from the Setup Menu

When the desired setup record has been recalled, either press the appropriate softkey to change the analyzer setup or exit from the Setup Menu by pressing **EXIT**.

CHAPTER

13 Storing and Recalling Non-Autostore Data

This chapter discusses the file and record structure used for the storage of data in the analyzer, the transfer of files between the internal memory and the optional floppy disk drive, and the storage and recall of normal (non-autostored) data blocks. The storage and recall of autostored data blocks is discussed in Chapters 15 and 16.

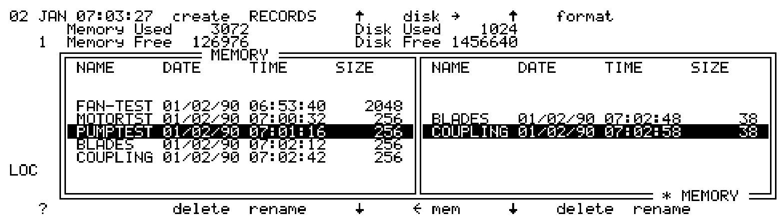
Files Operations

Data measured or generated from the 3000+ are stored to nonvolatile RAM memory as data records within user created and named data files. When the optional floppy disk drive, Model DVX003, is connected, these files can be subsequently transferred from internal memory to a floppy disk, and also from a floppy disk back into the internal memory.

Accessing the Files Menu

The Files Menu, shown in Figure 13-1, is accessed from any of the Analysis Menus by pressing **FILES [O]**.

Figure 13-1 ***Files Menu***



Files Information

The left half of the screen displays information concerning the files stored in the memory of the 3000+. This consists of the name, date and time of file creation and the size of the file in bytes.   The right half of the screen displays similar information concerning the files stored on the floppy disk.

Near the top of the display, the amount of memory already used for stored data records and the amount of free memory available for further data storage are indicated for both the internal memory and the disk memory on the left and right halves, respectively, of the screen. The capacity and volume name (user-assigned) of the disk in the disk drive is also indicated.

One of the files names displayed on each side of the screen will be highlighted by a horizontal black background strip. The user can shift the list of internal memory files up and down past the highlight using the softkeys ↑ **[C]** and ↓ **[K]**.

Similarly, the list of floppy disk files can be shifted up or down using the softkeys ↑ **[E]** and ↓**[M].**

When the Files Menu is first displayed, presses of the horizontal arrow keys will result in an up or down paging of the internal memory files on the left of the screen. However, whenever any of the up or down arrow softkeys have been used for vertical shifting of the files on either side of the screen, presses of the horizontal arrow keys will result in paging of the files on that same side of the screen.

In most of the file manipulations, operations are performed on the file which is highlighted. Thus, the use of these vertical arrow softkeys and the horizontal arrow keys are fundamental in selecting a particular file. The paging of the files up and down is also necessary when there are more files than can be displayed on the screen at one time.

Creation of Files

A new internal memory file is created by pressing **create [A].** The message “Enter new name:” on the right of the screen, accompanied by a blinking cursor, will prompt the user to type in a file name of up to eight characters using the keypad and then press **EXIT**. The new file information will then be listed at the bottom of the internal memory file listing, and it will also be highlighted. Note that when typing the file name, the hardkey **CLEAR** may be used to clear the entry field and the horizontal arrow keys can be used for editing.

Renaming Files

The highlighted internal memory file name can be renamed by pressing **rename [J]**. In response to the prompt on the right of the screen, type a new name and press **EXIT**. The new name will then appear in place of the former name in the listing. If after selecting to rename the file it is desired to abort that operation, simply use the **CLEAR** hardkey to clear the entry field and press **EXIT**. The message “Invalid name” will appear on the right of the screen and the file name will remain unchanged.

The highlighted disk file name can be similarly changed using the softkey **rename [O].**

Deleting Files

The highlighted internal memory file can be deleted by pressing **delete [I]**. The message “Delete highlighted file?” on the right of the screen prompts verification of the delete operation by the user, who will press **YES [A]** to proceed with the deletion, or **NO [C]** to abort the deletion operation.

The files management of the Model 3000+ requires that there always be at least one defined file, so when there is only a single file defined, it cannot be deleted. In such a case, simply create another file prior to making the deletion. There can be a problem in the case where the entire memory of the instrument is used for a single file. That file cannot be deleted since it is the only one, yet the user cannot create a new file because there is no memory available to store it. The solution is to delete one or more records within that file until there is sufficient memory available, 256 bytes, to create a new file and thus delete the other one as desired.

The highlighted disk file can be similarly deleted by pressing the softkey **delete [N]**.

Formatting a Floppy Disk

Place the disk to be formatted into the floppy disk drive. Note that any data already stored on this disk will be lost as a result of this operation. Press **format [F]**, which will produce the message “Enter volume: 3000+ Data” with a flashing cursor over the 2. This prompts the user to input a volume name for this disk. Use the keypad to type in the desired name and press **EXIT**. It is not obligatory that a volume name be entered; the user may choose to clear the entry field using the **CLEAR** key and enter the blank field as the name.

Upon entering the volume name, the message “OK to format this disk?” will appear on the right of the screen requesting verification of the formatting operation. To continue press **YES [A]**. To abort the formatting operation, press **NO [C]**.

File Transfers to/from Disk

Only complete files can be transferred between the internal memory and a formatted floppy disk. All data transfers to the disk are from internal memory; data cannot be stored directly to disk from the analyzer data buffers.

To transfer the highlighted memory file to the disk, press **disk** → **[D]**. Following the transfer, the file will appear highlighted with the same name at the bottom of the floppy file listing. If there is already a file on the floppy disk having the same name, the message “Overwrite this file?” will appear on the right of the screen. To approve this overwriting operation, which will cause the original file of this name on the floppy disk to be lost, press **YES [A]**. To abort the file transfer and save the original file on the disk, press **NO [C].**

The highlighted disk file can be transferred to the internal memory in a similar manner by pressing ← **mem [L]**.

Selection of the Active File

When measuring with the 3000+, only one of the internal memory files can be active. All storage of data from the analyzer to internal memory will be to the active file, and all data which is recalled from internal memory will be from the active file. The active file is determined by the internal memory file which is highlighted when the **EXIT** key is pressed, which returns the 3000+ to the Analysis Menu which had been active when the Files Menu was originally accessed.

Whenever the 3000+ is being used for measurements and analysis, the name of the active file appears on the lower left of the display. Changing the active memory file involves accessing the Files Menu, highlighting the desired memory file name (creating and naming a new one if necessary) and exiting back to the Analysis Menu.

Record Operations from the Files Menu

Classification of Record Types

When data is stored from the analyzer data buffers to the active file record, the data blocks are classified into 45 different record types based on the type of data being stored. These are the record types which are permitted:

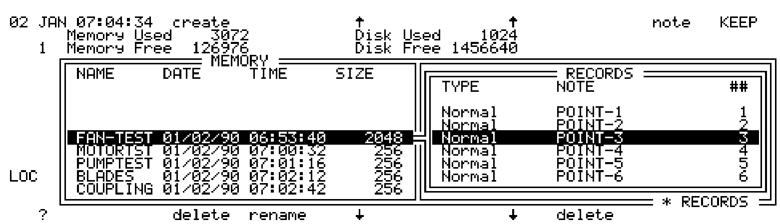
Normal, Leq, Minimum, Maximum, SEL, RT60, vsRPM Trace, Ln Trace, Ln Table, Intensity, Power, Autospectrum, AutoCorrel, Normalizatn, CrossSpectr, CrossCorrel, Transfer Fn, Impulse, Coherence, Time Wavefm, Cepstrum, Lifter, By Tach, By Time, User Curves, Setups, Units Data, SLM, (Normal + SLM), (Leq + SLM), (Minimum + SLM), (Maximum + SLM), (SEL + SLM), (MaxSpec + SLM), (ByTime + SLM) (ByTach + SLM), MaxSpectrum, Macros Data, By Tach Int, By Time Int, By Tach Crs, By Time Crs, Print Setup, Class Setup and FieldInd

See the section Record Classification later in this chapter for a more detailed explanation.

Records Listing

From the Files Menu, a listing of the data records stored in the highlighted memory file is displayed on the right half of the screen as shown in Figure 13-2 by pressing **RECORDS [B]**.

Figure 13-2 ***Records Menu***



The softkeys ↑ **[E]** and ↓ **[M]** can be used to shift the record listing vertically upwards and downwards while the highlight on the left of the screen remains on the name of the file whose records are being displayed. The listing of the records includes the Record Type and Record Number of each, along with a Note. The note field displays the first 17 characters of the note which was attached to the record at the time of storage. In order to see a note which is more than 17 characters in length, press **note [G].**

Note Editing

When the records are listed as described above, the user may create or edit the note field of the highlighted record by pressing **note [G]**. The creation and editing of notes is described in detail in Chapter 14.

Deleting Records

The highlighted record on the right side of the screen can be deleted from the memory file highlighted on the left side of the screen by pressing **delete [N]**. The message "Delete record?" on the right of the screen prompts the user to verify the deletion operation. Press **YES [A]** to proceed with the deletion, or **NO [C]** to abort the deletion operation.

Recalling a Record from the Files Menu

From the record listing, depending on the record type, the highlighted record can be recalled and displayed by pressing **KEEP [H]**. This will produce the message "KEEP record and exit?" on the right of the screen. To abort the procedure press **NO [C]**.

To continue the recall procedure, press **YES [A]**. The analyzer setup will be configured to that which was active at the time that data record was stored and the corresponding Analysis Menu will be displayed along with the data stored in the data record. The word KEEP is used to indicate that the analyzer setup configuration will be kept to that recalled from the data record.

Only records having the form of spectra or time waveform blocks can be recalled and displayed from the Files Menu as described here. The following types of records cannot be recalled in this manner:

vsRPM Trace, Ln Trace, Ln Table, User Curves, Setups, Units Data, Macros Data, Print Setup, Class Setup and Field Indicators

When the 3000+ is not in the Files Menu, stored records can be recalled from the active memory file using the **RECALL** hardkey as described later in this chapter.

Storage of Normal (Non-autostored) Data to Internal Memory

Storage of Data Blocks

We refer to the storage mode of the 3000+ as normal unless the autostore mode of storage is active. In the normal mode of storage, the displayed data block is stored each time the **STORE** hardkey is pressed. This data block is stored as a single record into the active memory file, whose name is displayed on the lower left of the screen. If the indicated file name is not the one into which it is desired to store the data block, it is necessary to return to the Files Menu and highlight the desired file before exiting so that it becomes the active file.

Record Classification

The classification of the records into types is based on the setup of the analyzer and the specific parameter which is being displayed. These are as follows:

Sound level data measured using the Wide Dynamic Sound Level Meter Mode and stored using the normal storage mode:

SLM

Sound level and spectral data measured using the Sound Level Mode and stored using the normal storage mode:

(Normal + SLM), (Leq + SLM), (Minimum + SLM), (Maximum + SLM), (SEL + SLM), (MaxSpec + SLM)

Sound Level and Spectral data measured using the Sound Level Mode and stored using the Autostore byTime or byTach mode:

(byTime + SLM), (byTach + SLM)

Spectral data measured using the Normal Analysis mode and the normal storage mode:

Normal, Leq, Minimum, Maximum, SEL or MaxSpectrum, depending on which of these parameters is being displayed.

Spectral data measured using the Cross Analysis mode and the normal storage mode; FFT and Octave band-widths:

AutoSpectrm, CrossSpectrm, Transfer Fn or Coherence depending on which of these parameters is being displayed.

Spectral and time domain data measured using the Cross Analysis mode and the normal storage mode; FFT band-widths only:

AutoCorrel, CrossCorrel, Impulse, Time Wavefm, Cepstrum or Lifter depending on which of these parameters is being displayed.

Spectral data measured using the Intensity mode and the normal storage mode:

Intensity or Power depending on which of these parameters is being displayed.

Spectral data measured using the Autostore By Time storage mode:

By Time(Standard Analysis mode)

By Time Crs(Cross Analysis mode)

By Time Int(Intensity mode)

Spectral data measured using the Autostore By Tach storage mode:

By Tach(Standard Analysis mode)

By Tach Crs(Cross Analysis mode)

By Tach Int(Intensity Analysis mode)

Curves generated using Ln and Statistics using the Standard Analysis, WDR SLM, or SLM+A mode and the normal storage mode:

Ln Trace, Ln Table

Level versus RPM curves generated and displayed from the vsRPM Graphics Menu

vsRPM Trace

Decay Time curves displayed from the RT60 Menu

RT60

Cross channel normalization curves generated by the normalization procedure to minimize cross channel amplitude/phase mismatch

Normalizatn

User Weighting Curves generated and displayed from the Set User Menu

User Curves

3000+ Setups stored from the Setup Menu

Setups

Units

Units Data

Stored macros

Macros Data

Custom Print Setups stored from Print Menu

Print Setup

Classification Lines stored from Class Lines Setup Menu

Class Setup

Field Indicators stored from Intensity Power Summation Menu

Field Ind

The user must bear in mind that ONLY the displayed data block is stored. For example, in the Standard Analysis Mode, spectra for Normal, Leq, Max, Min and SEL are calculated for each input channel. If the 3000+ is set for dual channels standard analysis and the display mode is set for Normal, then pressing **STORE** will result in the storage of one record of type Normal which contains a Normal spectrum for both channels. If the user wishes to store the Leq spectra as well, the display must be changed to LEQ mode and the **STORE** key pressed again, resulting in the storage of one data record of type Leq which contains an Leq spectrum for each of the two channels. Similarly, storage of the Max, Min and SEL spectra require selection of each of these display modes and a press of **STORE.**

In the case of complex data blocks, the data block is stored in the format of the display (real/imaginary or magnitude/phase). If the user wishes to be able to recall and display the data block in both the rectangular and polar coordinate representations, he must display and store the block twice, using each of the coordinate systems.

Storage Verification

Following the storage operation, the message “STORE - XXXX N” will appear on the upper right of the display. XXXX denotes the type classification of the stored record and N is an integer indicating that this particular record is the Nth record of that particular type which has been stored into the active file. Records of each type are numbered sequentially within a file in the order of their storage.

Setup Information

When the data block represents measured data such as a spectrum or a time waveform block, complete setup information of the analyzer (Analysis mode, averaging type and time, autostore or not, etc.) at the time of the acquisition of the data block is stored in the data record.

Notes

Information contained in the note field at the time of storage of the data block is also stored in the data record. The Note Menu is accessed by pressing the softkey **note [G]**. The user should thus create the desired note before storing the data block, as described in more detail in Chapter 14. The user can also add a note to a record after it has been stored as described earlier in the section Record Operations from the Files Menu of this chapter.

Recall and Display of Data Records (Non-autostored) from Memory

This section refers to the recall of non-autostore records from one of the operational menus of the 3000+. Recall of records from the Files Menu was discussed earlier in this chapter.

Records are recalled from the active memory file, whose name is indicated on the lower left of the screen. If this is not the file from which it is desired to recall records, it is necessary to return to the Files Menu, highlight the desired file and then exit to make that the active file. If the desired file is on a floppy disk, that file must be transferred to the analyzer memory and made the active file before recall can be performed.

Analyzer Setup for Recall

An explanation of the classification of records at the time of storage was presented above. In order to recall a particular type of record, the 3000+ must be placed in a setup configuration which corresponds to the type of record to be recalled.

Examples:

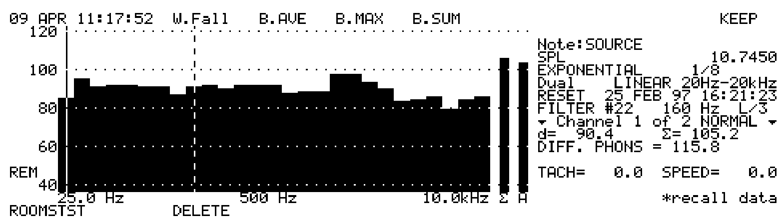
- With the 3000+ in the Standard Analysis mode and the normal storage mode, and the selected display parameter Leq, then the recall operation will recall only records of the type Leq.
- Change the display parameter to Maximum, and only Maximum type records will be recalled. Both FFT and octave bandwidths can be recalled.
- With the 3000+ in the Cross Analysis mode, normal storage mode, and the selected display parameter Coherence, only records classified Coherence will be recalled. Both FFT and octave bandwidths can be recalled. If the display parameter is changed to Impulse, only impulse response records measured with FFT filtering will be recalled.
- With the 3000+ in the autostore By Time storage mode, regardless of the Analysis mode, only By Time records will be recalled. These may be FFT or octave bandwidths.
- With the 3000+ in the RT60 Menu, only RT60 records will be recalled.
- With the 3000+ in the SLM Mode, set for normal storage, and the selected display parameter Leq, then the recall operation will recall only data records of the type (Leq + SLM). Change the display parameter to Normal and only records of the type (Normal + SLM) will be recalled. In either case, both SLM and spectral data will be recalled. However, the time history line analogous to a strip chart recording which was drawn during the original measurement is not stored with the measurement, so it will not appear after a recall. In order to store a time

history of the sound pressure level and the spectrum, the vsTime autostorage function must be used. This is described in Chapter 15.

Recall Operation

Data Records are recalled from the active memory file by pressing **RECALL**. This will result in the display of the Recall Menu, shown in Figure 13-3.

Figure 13-3 ***Recall Menu***



At the same time, one of the stored data records corresponding to the 3000+ display setup will be recalled.

Record Type and Number Indication

The message “RECALL XXXX N” on the upper right of the screen indicates that the Nth record of the type XXX has been recalled from the active memory file and is being displayed. In many cases the first record of that type will be recalled, but if some recent operations have been made on one of the records (for example if a record had recently been stored or recalled) then that record number will be recalled. The instrument setup corresponding to that recalled record will also be displayed.

Note Presentation

The note stored along with the record will also be displayed on the upper right of the display in the format “Note:XXXXXXXXX”. Only 19 characters can be displayed in this display format. If the note field is larger than 19 characters, press **note [G]** to display the entire note. Press **EXIT** to return to the data display.

Changing Displayed Record Number

After a recall operation, the message “recall data” on the lower right of the display indicates that the horizontal arrow keys now control the recall of records. Pressing them will page sequentially through the records of the same type, as indicated by the changing value of N in the message on the upper right of the screen. The right horizontal arrow recalls records placed later in the sequence (higher record numbers) and the left horizontal arrow recalls records placed earlier (lower record numbers) in the sequence. To jump faster through the record numbers, use the **SHIFT** key in conjunction with the horizontal arrow keys.

Cursor Utilization

In order to utilize the cursor to readout the data being displayed, press **CURSOR.** This will activate one of the cursors and place it under the control of the horizontal arrow keys. The use of cursors is explained in more detail in Chapter 8.

Deleting Stored Records

There are two ways in which stored records can be deleted. From the Recall Menu, the record which was last recalled, indicated by the value of N in the message “Recall - Type N” on the upper right of the screen, will be deleted upon pressing **DELETE [I]**. The message “Delete the current record?” on the upper right of the screen will appear for verification of the deletion. Press **YES [A]** to continue with the deletion, or **NO [C]** to abort the deletion.

Following the deletion, the remaining records will be repacked. Those records following the one deleted will be moved down one in sequence within the file, reducing each of their record numbers by one. As a result, the message on the upper right of the screen will still indicate the same value of N as before the deletion, but this record will now represent the record which had been stored just after the deleted record since its index has been reduced from N+1 to N. Those records located before the deleted record in sequence within the file will maintain their positions and record numbers.

Individual stored records can also be deleted from the Files Menu as explained in this chapter under the section Record Operations From the Files Menu.

Block Averaging of Stored Records

Sequentially stored records of the same type can be averaged together using the Block Averaging Function. From the Recall Menu, press **B.AVE [B]**, which will bring to the upper right of the screen the message “AVERAGE: 0001 - 0002". Use the numeric fields until they represent the range of record numbers which are to be averaged together. Upon pressing **EXIT** a single averaged spectrum will be created and displayed. Note that it will have the word “AVERAGE” on the right side of the screen, first line down, instead of the elapsed time which is usually displayed with measured spectra. In order to store this averaged spectrum, press **STORE**. The message on the upper right of the screen will indicate into which record it has been stored.

Block Maximum of Stored Records

The Block Maximum operation can be applied to sequentially stored records of the same type and bandwidth to determine the maximum amplitude which occurred in each frequency band over the entire set of sequential records. From the Recall Menu, press **B.MAX [C]** which will bring to the upper right of the screen the following message

MAXIMUM: 0001 - 0002

Use the numeric keypad and the horizontal arrow keys to enter values representing the first and the last of the sequence of record numbers over which the block maximum operation is to be performed. Upon pressing **EXIT**, the operation is performed and the resulting spectrum is displayed. Note that the word MAXIMUM appears on the right of the screen, 2nd line down, in place of the elapsed time usually displayed with a measured spectrum, to indicate that this spectrum is the result of the Block Maximum operation. Following the Block Maximum operation the spectrum is not automatically stored. To store this spectrum, press **STORE**. The record number into which the spectrum has been stored will be indicated on the upper right of the screen.

If the records contained within the specified range are not all of the same type and bandwidth, the Block Maximum operation will not be completed, and the message "NOT SIMILAR DATA" will appear on the upper right of the screen.

Block Summation of Stored Records

Sequentially stored records of the same type and bandwidth can be averaged together using the Block Summation function. This is a Root-Mean-Square summation which is appropriate for the addition of decibel levels. From the Recall Menu, press B.SUM [D], which will bring to the screen the following message:

RMS SUM: 0001 - 0002

Use the numeric keypad and the horizontal arrow keys to enter values representing the record numbers of the first and the last records to be summed. Upon pressing EXIT, the summation will be performed and the resulting spectrum displayed. Note that the word SUMMATION appears on the right side of the screen, 2nd line down, in place of the elapsed time displayed for measured spectrum, to indicate that this spectrum is the result of a block summation rather than a measurement. This spectrum is not automatically stored. Press **STORE** to store the spectrum, following which the record into which the spectrum has been stored will appear on the upper right of the screen.

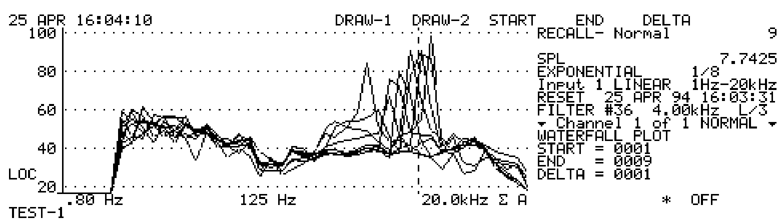
If the records contained within the specified range are not all of the same type and bandwidth, the Block Summation operation will not be completed and the message “NOT SIMILAR DATA” will appear on the upper right of the screen.

Waterfall Display of Stored Records

The waterfall display function permits the sequential display of a series of individual spectra of the same type which have been stored in sequence, each one remaining on the screen after it has been displayed. Thus, we will see drawn upon the screen one spectrum, then overlaid upon that another spectrum, then another, etc.

Once in the Recall Menu, press **W.Fall [A]** which will bring up the Waterfall Menu, shown in Figure 13-4.

Figure 13-4 ***Waterfall Menu; 2D Format***



On the right of the screen we see a table indicating the present values of START, END and DELTA. These represent the first and last records in sequence which are to be displayed, and the incremental record number between displayed spectra, respectively. For example, using the following combination

START	=	0010
END	=	0020
DELTA	=	0002

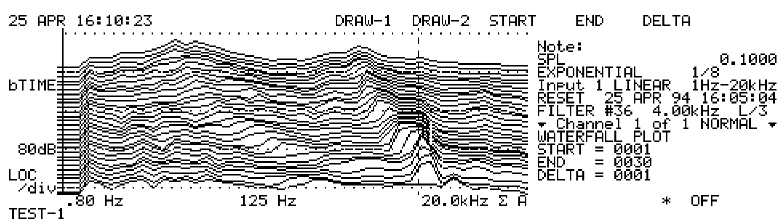
the records displayed will be numbers 10, 12, 14, 18 and 20 in sequence.

To edit any of these numbers, press **START [E], END [F]** or **DELTA [G].** This will produce the message “W.FALL sXXXX, eXXXX, ΔXXXX” with a flashing cursor to denote where inputs from the keypad will begin. The only difference between initiating this input with the **START [E], END [F]** or **DELTA [G]** softkeys is that the flashing cursor will be positioned for immediate editing of the START, END or DELTA values, respectively. Use the numeric keypad and the horizontal arrow keys to edit the values as required and press **EXIT.** The display sequence will begin immediately upon pressing that key.

There are two formats available for the presentation of the waterfall plots. In the two dimensional format, produced by pressing **DRAW-1 [C]**, the spectra are simply overlaid, one at a time, without any offsets in the vertical and horizontal directions. This produces a graphic as shown in Figure 13-4.

In the three dimensional format, produced by pressing **DRAW-2 [D]**, an offset in both the vertical and horizontal directions is added to each successive spectrum curve, providing perspective to the view.

Figure 13- 5 ***Waterfall Menu; 3D Format***



Exiting from the Recall Mode

Although the record type which was recalled from memory was determined by some aspects of the analyzer setup at the time of recall, there are other setup parameters which may be different between the recalled records. For example, within the records classified as Normal, some may use FFT filtering and others 1/3 octave. In addition, a variety of different averaging methods and times may have been used for the different measurements. Suppose the user had the analyzer configured for FFT analysis at the time the recall was initiated, and that during the recall operation a record measured using 1/3 octave was recalled and displayed. It will be noticed that when the 1/3 octave record was recalled, the setup parameters on the screen changed from those associated with FFT analysis to those associated with 1/3 octave analysis. The user has two options as to which setup the analyzer will be configured after exiting from the recall operation.

To have the 3000+ return to the setup which was active at the time the recall operation was initiated, press **EXIT.** The display of the recalled data block will be lost when this is done since the setup which had been active prior to the recall operation may not be the same as that corresponding to the presently displayed data block.

To have the 3000+ remain in the setup configuration shown on the screen (based on the record last recalled), press **KEEP [H]**. It is necessary to follow this procedure if the recalled data block is to remain on the screen.

In either case, the 3000+ will then return to the SLM or Analyzer Menu.

Memory Requirements (Non-autostore Records)

Each point of a data block (spectrum, time record, RT60 data, etc.) requires 2 bytes for storage. Complex spectra require two points per filter band. The note field requires 64 bytes.

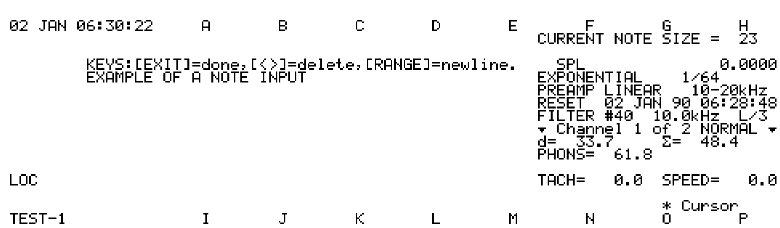
14 Annotation of Data Blocks

CHAPTER

Annotation of Data Blocks

It is possible to annotate a data block, such as adding a test number, comments concerning the measurement process, the test procedure, etc. and store them along with the data block. The softkey **note [G]** which is used for the annotation feature, is found in nearly all the Menus of the 3000+. To access the Note Menu, shown in Figure 14-1, press **note [G]**. If there was already a note attached to the data block displayed when the key was pressed, it will now be displayed on the screen along with the number of characters of the note (on the upper right of the display). Otherwise, the note field will be blank.

Figure 14- 1 ***Note Menu***



If creating a note from a blank field, simply type in the note via the keypad and press **EXIT** when finished. There are a number of editing features associated with the creation of the note field to provide flexibility and format control.

When typing into the note field with the alphanumeric keys, after 40 characters appear on a line a “+” symbol will appear at the end of the line and the blinking cursor will move to the next line down at the left margin for the beginning of a new line.

While entering characters, pressing the **RANGE** hardkey on the lower right of the front panel will terminate the present line and move the text cursor to the left margin of the next line down. Multiple presses of this key will move the text cursor down by a number of lines equal to the number of keystrokes.

When a sequence of characters have been typed onto the screen, the horizontal arrow keys may be used to move the blinking text cursor forward and backward through the text. Pressing the **CURSOR** hardkey on the lower right of the front panel will delete the character highlighted by the text cursor. All following characters will be scrolled backwards to fill in the space created by the deletion of the character. When the text cursor is within a text string, typing additional characters will cause them to be inserted to the left of the cursor position.

Note that there is a space key on the right of the front panel. When the text cursor is within a text string and the **RANGE** key is pressed, the portion of the string to the right of and including the character highlighted by the cursor will be shifted vertically downward and to the left margin to begin a new line. Repeated presses will move the cursor further downward.

Pressing **CLEAR** will clear the entire text string. In most cases when the **note [G]** key is pressed, the previously entered or recalled text string will appear. Use the **CLEAR** key to erase the previous note before beginning a new one, unless the new note will be sufficiently similar that editing would be more efficient.

After creating the note field for the data block, store the block and the note by pressing **STORE [I]**. Now, recall the data block by pressing **RECALL [J]** and observe that the first 19 characters of the note field are displayed on the upper right of the display. If the stored note field is larger than 19 characters, simply press **note [G]** to display the complete note in the center of the display.

To edit the note field of a particular data block after it has been stored, it is necessary to work from the Files Menu. This is explained in detail in Chapter 13, under the section Record Operations from the Files Menu.

CHAPTER

15 Autostore byTime

This chapter describes the byTime autostore capability of the 3000+. Before beginning this chapter, read Chapter 13, “Storing and Recalling Data” to understand the general file structure used for data storage and how to perform file operations.

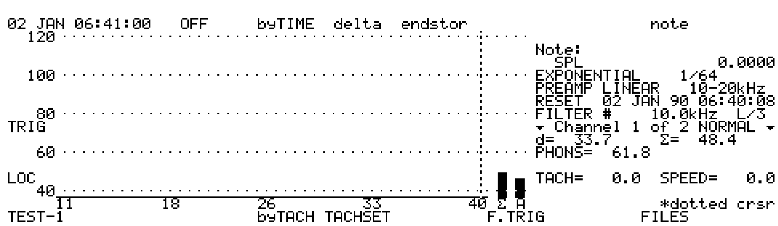
Setup for an Autostore Sequence

Before beginning an autostore operation, the data file into which the autostore data records are to be stored must be opened, as explained in the preceding chapter. Following this, return to one of the Analysis Menus.

Accessing the Autostore Menu

Access the Autostore Menu, shown in Figure 15-1, by pressing **AUTOSTR [P]**.

Figure 15- 1 ***Autostore Menu***



In the Autostore byTime storage mode, the 3000+ will measure and store spectra at equally spaced time intervals (in seconds) over a specified period of time (also in seconds). Any frequency or time domain data blocks may be autostored.

In the Analyzer mode of operation, one or two data blocks are stored each time interval, depending on whether single or dual channel analysis has been selected. With the 3000+ set for dual channel Cross analysis, both time and frequency domain data may be stored.

When the SLM mode of operation has been selected, in addition to a single channel frequency spectrum, the following SLM data are stored at each time interval: SLOW, SLOW MIN, SLOW MAX, FAST, FAST MIN, FAST MAX, IMPL MIN, IMPL MAX, Leq, SEL, Peak and Spectrum $\sum$.

Defining Delta Time and End Time

To prepare the 3000+ for a byTime autostore operation, first set the time interval desired between successive storages by pressing **delta [C]** which will produce the message “DELTA TIME = XXXXXX.XXXX” along with the flashing window cursor indicated that a numerical input is required. Type in the number and press **EXIT**. Note that the value last entered for Delta Time is already displayed, so if the same value is desired simply press **EXIT** immediately.

Next set the total time period over which the automatic storage is to continue by pressing **endstor [D]** which will produce the message “END TIME = XXXXXX.XXXX”. Type a value using the numerical keypad and press **EXIT**. As for the Delta Time, the data field for the entry of the END TIME value will already contain the value last entered, so if that same value is desired, simply press **EXIT**.

Delta Time Limitations

The time required to transfer the data to memory during the autostore sequence places some limitations on the minimum value which can be set for the DELTA TIME. This is a function of the bandwidth and the number of channels as indicated below. If the user selects a value less than the minimum permitted value, the DELTA TIME will default to the minimum value. This is easily demonstrated by inputting a zero value for this parameter, then pressing the **delta [C]** softkey and noting the value which is displayed in the parameter input field on the upper right of the display.

Table 15-1 **Minimum DELTA Time, milliseconds, using Octave and Fractional Octave Bandwidths**

	Filter Bandwidth, octaves	
# Channels	1/1	1/3
1	2.5	2.5
2	5.0	5.0

Table 15-2 **Minimum DELTA TIME, milliseconds using FFT bandwidths**

	FFT Analysis Number of lines			
# Channels	100	200	400	800
1	5	10	20	40
2	10	20	40	80

Although we have entered the autostorage parameters, the autostore function is not yet enabled. The 3000+ can still be run in a standard manner by pressing **RUN/STOP**. The autostore byTime mode is enabled by pressing **byTIME [B]**. The message “bTIME” will appear to the left of the display to indicate that the Autostore byTime mode is active.

Selection of Spectral Type to be Autostored

It is the displayed spectrum type which are sequentially stored into the autostore record. Thus, it is possible to autostore Normal, Leq, MIN, MAX, SEL, or Mx.Spec spectra. In most applications, it will be the Normal spectra which will be desired for the autostore operation. If the active display type is not desired for the autostore, exit from the Autostore Menu to the Main Menu and change to the desired display type before initiating the autostore sequence.

Count Averaging Special Considerations

When FFT filtering is used in conjunction with Count Averaging, the DELTA and ENDSTORE numbers refer to number of spectra rather than time. In that case, the prompts for data entry on the right of the screen will refer to DELTA COUNT and END COUNT, respectively. The actual rate at which spectra are stored depends upon the rate at which they are produced by the processor, which in turn depends upon the number of lines and number of channels used. Therefore, the autostorage is not strictly speaking a byTime operation, and one loses the reference to absolute time because only the spectrum number is identified with each spectrum. Nevertheless, the mechanics of the process are similar enough in procedure that the same general description should suffice.

Initiation of an Autostore byTime Sequence

An autostore sequence may be initiated manually or automatically using a Frequency Trigger.

Manual Start

Once the Autostore parameters (Delta Time and End Time) have been set, and the 3000+ has been put into the Autostore Mode by pressing **byTIME [B]**, an autostore sequence will begin as soon as the **RUN/STOP** key is pressed. Both the measurement of the spectra and their automatic storage will be initiated in this manner.

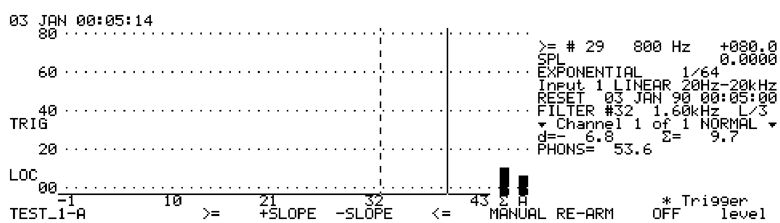
In some instances the user would like to be able to observe spectra being measured and displayed before manually beginning the autostorage sequence. This is done from the Frequency Trigger Menu, accessed from the Autostore Menu by pressing **F.TRIG [M]**, and then pressing **MANUAL [M]**. This will produce the message “Manual use

[R/S]” on the upper right of the display. Following this, pressing the **RUN/STOP** hardkey will begin the measurement and display of spectra along with the indication that the 3000+ is in the ARMED state on the right of the display. A subsequent press of **RUN/STOP** will initiate the autostorage sequence beginning with the next spectrum to arrive from the processor. Pressing **OFF [O]** from the Frequency Trigger Menu will take the 3000+ out of this particular mode of manual triggering.

Frequency Trigger Start

The Frequency Trigger is used when the user wishes to initiate the autostorage sequence based on the amplitude of one of the frequency bands measured on Channel One. The Frequency Trigger Menu, shown in Figure 15-2 is accessed from the Autostore Menu by pressing **F.TRIG [M]**.

Figure 15- 2 ***Frequency Trigger Menu***



Note that the message “TRIG” appears on the left of the display to indicate that the Frequency Trigger Mode is active.

The setup of the Frequency Trigger function is described in Chapter 11.

After the setup of the Frequency Trigger, pressing the **RUN/STOP** key will put the 3000+ in the ARMED state as indicated by the message on the right of the display. At the same time, the measurement and display of spectra will begin. Actual averaging and autostorage of spectra will not begin until the sequence is initiated by the satisfaction of the Frequency Trigger criteria. When this occurs, the state of the

3000+ will change to RUN as indicated on the right of the display.

If no event occurs to produce a frequency trigger, a subsequent press of the **RUN/STOP** key will disarm the frequency trigger and the message “NO TIME HISTORY STORED” will appear on the upper right of the display. Continued presses of the **RUN/STOP** key will simply toggle the 3000+ between the armed and disarmed states. To disable the Frequency Trigger altogether, access the Frequency Trigger Menu from the Autostore Menu by pressing **F.TRIG [M]** and then press **OFF [O]**.

For cases where the analyzer is to be used unattended, it is convenient to have the 3000+ rearm itself automatically after an autostore initiated by the frequency trigger so that data associated with a series of events can be captured. This is done from the Frequency Trigger Menu by pressing **RE ARM [N]** after selecting the frequency trigger criteria. This will produce the message “Autostore rearm mode set” on the right of the display. As usual, pressing **RUN/STOP** will arm the frequency trigger function. However, after the completion of an autostore sequence initiated by the frequency trigger, the 3000+ will return to the ARMED state so that a subsequent trigger will initiate another autostore sequence. In this mode, there is no message on the display indicating the storage of a record. Pressing the **RUN/STOP** key will disarm the frequency trigger and return the system to the manual trigger mode. The message “NO TIME HISTORY STORED” simply means that this last press of the **RUN/STOP** key did not produce an autostore, and does not indicate that the preceding frequency triggered records were not stored.

Conclusion of an Autostore byTime Sequence

Whether the autostore sequence is initiated by a manual trigger or by a frequency trigger, the 3000+ will then begin producing spectra which will be stored automatically at the time intervals selected by Delta Time until the time period corresponding to End Time is reached. A value of elapsed time is stored with each spectrum (or set of spectra if multichannels were used). In the special case of FFT filtering with Count

Averaging, the spectrum number rather than the elapsed time is stored, and the sequence ends when the total number of stored spectra specified under End Time is reached.

At the conclusion an autostore sequence (except when the RE-ARM Mode is active), the data will be automatically stored and then recalled for display to the screen to indicate into which record number the data have been stored. The message will be “STORE - By Time N” where N is the record number into which the data have been stored. Detailed descriptions of the data storage format and the display procedure are presented later in this Chapter.

When the manual start method is being used to initiate autostore sequences using the **RUN/STOP** key, at the conclusion of each sequence another sequence can be initiated immediately by another press of the **RUN/STOP** key. The user may continue to perform autostore operations in this manner until the memory is full, indicated by the message “OUT OF MEMORY” on the upper right of the display.

Disabling Autostore byTime

When the 3000+ is in the Autostore byTime mode of operation, as indicated by the message “bTime” on the left of the display, pressing the softkey **OFF [A]** will return the operating mode to the standard (non-autostore) storage mode. The message “bTime” will then no longer be displayed.

Data Storage Format

Autostorage can be used with 1 or 2 channels in all three Analysis Modes with either Octave or FFT filtering. There are a variety of data display formats available with each Analysis mode (Normal, Leq, Max, Min, and SEL for Standard; Autospectra, Cross Spectra, Transfer Function, etc. for Cross; and Intensity, Quality, Average SPL and Particle Velocity for Intensity). The display format of the sequentially acquired and stored spectra will be the same as the display format active at the time of storage.

Averaging Time Considerations

FFT Analysis

When using the Linear Repeat averaging type, the time interval between storage of spectra should be set equal to or greater than the averaging time to avoid trying to store data before it is available: A spectrum (or pair of spectra for dual channel) is read from the averaging buffer for storage at the time interval set by Delta Time. At this time the averaging buffer is reset to zero. If the time interval is less than the averaging time, zero data is stored prior to good data being available.

Example: Suppose the analyzer is configured for single channel and 800 line FFT. At this configuration, the smallest autostore time interval which can be input is 0.04 s because that is the rate at which new FFT spectra are produced. If the autostore time interval is smaller than that, say 0.01 s, an empty averaging buffer will be read three times before a full one is available.

Example: If the time interval had been selected to be 0.12 s, the average value would change every 0.04 s as a new spectrum is added to the linear average. After 0.12 s the average of three spectra would be stored, and the averaging buffer would reset and begin calculating a new average.

Example: If the time interval was not used but rather the averaging had been selected to be Count = 4, storing of data would be after four spectra are averaged: The value in the averager would be updated every 0.04 s as a new spectrum is added to the spectrum average. But after four spectra had been averaged (0.16 s), the data would be stored, and the averager would be reset and begin calculating another average.

Octave Filters

For true by-time operation of Autostore function, the user will generally select Linear Repeat, Exponential, BT/Exp or

BT/Lin averaging. The storage operations are governed by the internal clock, and whenever an integer multiple of the selected value of Delta Time is reached, whatever data is in the averaging buffer(s) is stored.

Using Linear Repeat, it is logical to set the Delta Time value to equal the averaging time. Since the averager is reset at the end of each interval, each spectrum stored would represent an average calculated over the preceding Delta Time interval. Should the Delta Time be set to three times the averaging time, the detector would be reset twice before storage of a spectrum, meaning that the spectrum stored would represent only data measured over the last third of the interval time.

Similarly, with Exponential Averaging, it is logical to set the Delta Time to be close in value to the averaging time. If the Delta Time were much less than the averaging time, there would be very little difference between the stored spectra due to the time constant of the averaging process. If it were much larger, there could be great changes in the spectra between intervals which would not be seen. One could use Linear Single, with a Delta Time less than the averaging time, but this would only display the build-up of the averaging process over a single averaging cycle.

When BT/Exp or BT/Lin are selected, the averaging times of the lower frequency filters are longer than those of the higher frequency filters because they have narrower bandwidths. This means that the averaged values of the lower frequency filter bands will be updated less frequently than the higher frequency band averaged values. At the end of any given Delta Time interval, it is only necessary to store the values corresponding to frequency bands whose averages have been updated since the last data storage operation. This is a much more efficient storage procedure than storing a full spectrum for each time interval which optimizes the use of the data memory. It is therefore recommended that one of these be used when it is necessary to store as many spectra as possible over a long time interval.

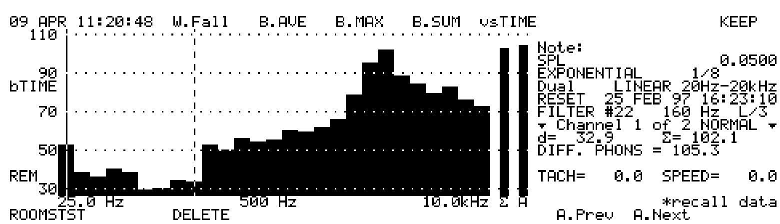
Recall and Display of Autostored Data

Pressing the hardkey **RECALL** while the 3000+ is in the autostore byTime mode will recall one of the byTime type records from the Active Memory File whose name is listed on the lower left of the screen. The message “RECALL - By Time N” on the upper right of the screen will indicate that the Nth record of the type byTime has been recalled. (If the 3000+ is in the SLM Mode, the message will be “RECALL-ByTime + SLM N”). In most cases this will be the record number which was last stored or recalled. To determine how many byTime records have been stored in a particular record, and to examine their note fields, use the Files Menu.

If the desired record is in another file, it will be necessary to access the Files Menu, change the Active Memory File and exit before performing the recall operation.

At the same time, the Recall Menu shown in Figure 15-3 will be displayed

Figure 15-3 ***Autostore Recall Menu***



and the message “recall data” on the lower right of the screen will indicate that the horizontal arrow keys are assigned to recall the individual spectra from the recalled record.

To recall an autostored record stored earlier (previous to) the one which has been recalled, press **A.Prev [N]** and note that the index N in the message on the upper right has been decreased by one, indicating that the previous record has now been recalled. Repeated presses of **A.Prev [N]** will page the recall procedure continually towards the first record stored in that file.

Similarly, pressing **A.Next [O]** will result in the recall of the autostored record which was stored later (after) the one which had originally been recalled, as indicated by a unity increase in the value of N in the message on the upper right. Repeated presses of **A.Next [O]** will page the recall procedure continually towards the last record stored in that file.

Displaying Individual Spectra

Once the desired record has been recalled, presses of the horizontal arrow keys will page through the individual spectra contained in the autostore record, bringing them sequentially to the screen. Each spectrum is tagged with the time it was stored, relative to the initiation of the autostore sequence. This is displayed on the right of the screen, first line down. An exception is when count averaging was used, in which case this field will show the spectrum number rather than a time value.

Presses of the left horizontal arrow key will produce a paging backwards in sequence toward the first spectrum stored. When there are many spectra in the record, pressing the **SHIFT** key along with the horizontal arrow key will produce a jump of more than a single spectrum, which is useful when seeking a spectrum far in sequence from the one being displayed. Use the keys **CH1** and **CH2** to select the displayed channels.

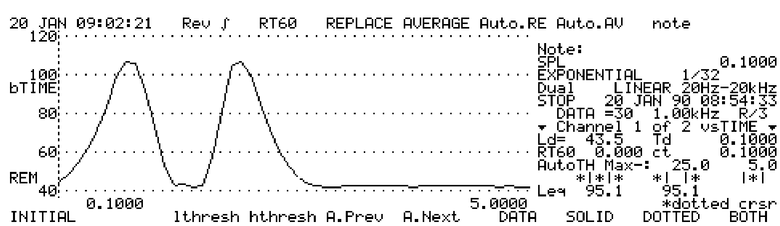
If during the data acquisition an overload condition occurs at one or more of the inputs, the inverse video message “OVERLOAD” will appear on the screen. However, the overload condition may not have affected all the inputs and may not have been in effect during the complete time period of the acquisition, meaning that some of the individual spectra in the autostored record may be accurate while others may be inaccurate due to the overload. While examining the individual spectra during the recall operation, the same message “OVERLOAD” will appear along with each spectrum which corresponded to an overload condition during the data acquisition.

To utilize the cursor for readout of the amplitude and frequency values of the displayed spectrum, press the hardkey **CURSOR** which will assign the horizontal arrow keys to control the cursor which was last active (dotted or solid). A second press of that key will bring up the Cursor Menu for selection of cursor type.

Display of Amplitude vs. Time

When a series of spectra have been autostored as a function of time, it is possible to select any single frequency band and display the level in that band as a function of time, exactly as if the original signal had been passed through a band-pass filter and then displayed upon a level recorder. To do this, recall the desired autostore record, move the active cursor to the desired frequency band, and press **vsTIME [E].** The amplitude vs. time curve will appear on the display as shown in Figure 15-4.

Figure 15-4 ***byTime Display***



The cursors can now be used to read the amplitude and time values of the displayed data. To display the broadband level versus time, see the section Broadband Level versus Time.

The amplitude versus time display mode is available for use with the following:

(1) Normal, Leq, Max, Min and SEL data records autostored byTime using the Standard Analysis mode

(2) Intensity and SPL data records autostored byTime using the Intensity Analysis mode. In this case, the softkeys INTENSITY [B] and SPL [D] are used to select which parameter is to be displayed.

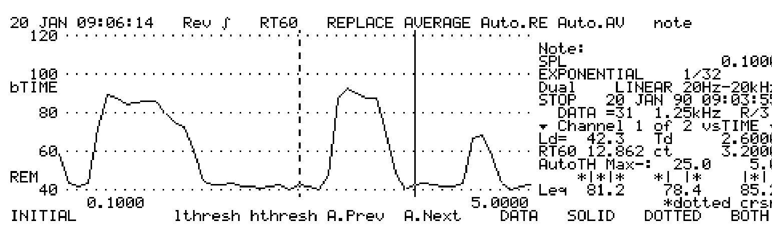
Although data records can be autostored byTime using the Cross Analysis mode, the amplitude versus time display mode cannot be used with these records.

If any of the individual spectra within the autostored record correspond to overloads, upon pressing the **vsTIME [E]** softkey to obtain a vsTime display the inverse video message "OVERLOAD" will also appear. Also, both cursors will converge together at the position along the horizontal axis corresponding to the time when the overload condition first occurred during the autostore acquisition sequence. Once in the vsTime display mode, the overload message will no longer appear when the **DATA [M]** softkey is used to change the frequency band of the display. The initial indication of an overload should be sufficient to warn that the effects of the overload will influence the data in any of the frequency bands for time values greater than that for which the overload first occurred.

Leq Measurements in the vsTime Display Mode

When data is being displayed in this mode, there are three values of L_{eq} indicated digitally on the lower right of the screen, as shown in Figure 15-5.

Figure 15- 5 ***L_{eq} Data in vsTime Display***



From left to right, these represent the following:

Leq for the entire record
(Denoted by the symbols *|*|* above)

Leq for the portion of the record outside the two cursors
(Denoted by the symbols *| |* above)

Leq for the portion of the record between the two cursors
(Denoted by the symbols |*| above)

Changing the Displayed Frequency Band

The frequency band which the displayed amplitude/time curve represents is indicated on the right of the display just above the indication of the displayed channel. To examine the amplitude/time curve corresponding to another frequency band without returning to the spectrum display and selecting another frequency band, press **DATA [M]**. The horizontal arrow keys assignment message on the lower right will become "*new data". As these keys are pressed, the indicated frequency band on the right of the display will change, and the amplitude/time curve corresponding to that indicated frequency will be displayed. This permits the user to display the amplitude/time curve for any frequency band using the horizontal arrow keys to scroll through the frequency bands one-by-one.

Broadband Level versus Time

While changing the displayed frequency band in the versus time display mode, if the right horizontal arrow key is used to move one increment beyond the highest frequency band, then the displayed curve will be that of the broadband level versus time, as indicated by the phrase "spectrum ∑" instead of a bandwidth center frequency on the fifth line down on the right of the screen.

Although the digital display weighting can be used in order that the autostored spectra be weighted before display, the broadband level is calculated before the effect of the display weighting is included. Thus, the user cannot take a series of autostored spectra and use digital weighting in order to obtain a curve of weighted broadband level (e.g. dBA) versus time. If the user wishes to examine the weighted broadband level versus time in addition to unweighted spectral data, the SLM Mode should be used with the autostore function. In that case, the SLM function can be A or C-weighted while the analyzer function is left unweighted.

SLM Data versus Time

When the 3000+ is in the Wide Dynamic Range Sound Level Meter Mode, after pressing **DATA [M]** the measured parameters can be displayed as a function of time in the following order, using the right hand arrow to page upwards through the list:

> SLOW, SLOW MIN, SLOW MAX, FAST, FAST MIN, FAST MAX, IMPULSE, IMPL MIN, IMPL MAX, Leq, SEL, PEAK

When the 3000+ is in the SLM+A Mode, after pressing **DATA [M]** the measured parameters can be displayed as a function of time in addition to the data for each frequency band. While using the right arrow key to page upwards through the frequency bands, after the highest band is reached, continue presses will display additional data in the following order:

> SLOW, SLOW MIN, SLOW MAX, FAST, FAST MIN, FAST MAX, IMPULSE, IMPL MIN, IMPL MAX, Leq, SEL, PEAK. and Spectrum Σ.

If these data were taken in the Dual Channel SLM+A Mode, use the hardkeys **CH1** and **CH2** to select the channel for which the data are to be stored.

Displaying the Same Frequency of Another Record

Suppose a number of autostore records have been stored, possibly representing reverberation decays measured at a number of different points within a room, and the user wishes to examine how the amplitude vs. time curves for a particular frequency vary from record to record. He begins as described above, recalling a record, selecting a frequency band with the cursor, and displaying the amplitude/time curve by pressing **vsTIME [C]**. If he now presses **A.Prev [K]** the amplitude/time curve for the same frequency, but calculated from the record stored previously will be displayed. Similarly, pressing **A.Next [L]** will produce the amplitude/time curve for the record stored after the one which was previously displayed. The key **DATA [M]** and the horizontal arrow keys can still be used to change the desired frequency band while examining amplitude/time curves from the various stored records.

Displaying and Storing Leq, MIN, MAX, SEL, and Mx.Spec Spectra

During a byTime autostore measurement sequence, the Leq, MIN, MAX, SEL, and Mx.Spec spectra are also calculated for the total time of the autostore sequence. Although the autostored spectra are automatically stored at the conclusion of sequence, the Leq, MIN, MAX, SEL, and Mx.Spec are not automatically stored. If the user wishes to examine these it is important that this be done immediately following the data acquisition before the data buffer is reset. For example, if a recall operation is performed immediately following the acquisition in order to examine the autostored data records, the data buffer will be reset and these other spectra will no longer be available. Although these other types of spectra can be displayed, they cannot be stored while the instrument is in the Autostore mode. So, following the conclusion of the autostore sequence, turn off the autostore mode from the Autostore Menu, access the Main Menu, and use the **Leq [B]**, **MIN [C]**, **MAX [D]**, **SEL [E]**, and **Mx.Spec [L]** hardkeys to display each of these other types of spectra, and use the **STORE** hardkey to store the displayed spectrum type.

Because the autostored spectra themselves can be of the type Normal, Leq, MIN, MAX, SEL, or Mx.Spec depending upon which display type is active at the initiation of the autostore sequence, be careful when switching between these display types while doing multiple autostore measurements. Most applications call for the autostorage of Normal spectra, but if the display type is switched to Leq, for example, to display and store the Leq spectra following the autostore sequence, and the display type is not changed back to Normal before the next autostore sequence, the next sequence will store Leq spectra instead of Normal spectra.

It is possible to create a key macro function which will in one operation perform the autostore sequence, and then sequentially display and store each of the other spectra and reset the display type to Normal in preparation for the next autostore sequence. Key macros are described in Chapter 19 of this manual.

Deleting Autostore Records

There are two ways in which autostore records can be deleted. From the Recall Menu, the record which was last recalled, indicated by the value of N in the message “Recall - By Time N” on the upper right of the screen, will be deleted upon pressing **DELETE [I].** The message “Delete the current record?” on the upper right of the screen will appear for verification of the deletion. Press **YES** [**A]** to continue with the deletion, or **NO [C]** to abort the deletion.

Following the deletion, the remaining records will be repacked. Those records following the one deleted will be moved down one in sequence within the file, reducing each of their record numbers by one. As a result, the message on the upper right of the screen will still indicate the same value of N as before the deletion, but this record will now represent the record which had been stored just after the deleted record since its index has been reduced from N+1 to N. Those records located before the deleted record in sequence within the file will maintain their positions and record numbers.

Individual autostore records can also be deleted from the Files Menu as explained in Chapter 13 under the section Record Operation From the Files Menu.

Averaging of Autostore byTime Records

Sequentially stored autostore records can be averaged together using the Block Averaging Function. From the Recall Menu, press **B.AVE [B]**, which will bring to the upper right of the screen the message “AVERAGE: 0001 - 0002". Use the numeric keys and the horizontal arrow keys to edit the two numeric fields until they represent the range of record numbers of the autostore records which are to be averaged together. The number of blocks which can be averaged in a single operation is limited to twenty. Upon pressing **EXIT** a single averaged autostore record will be created and stored into the next available autostore record number.

There is a softkey available for your convenience, which when toggled gives you the following options:

Softkeys	Softkey Functions
Start/E [H]	Shows the records from start to end
Last N [H]	Takes you to the end of the data block for averaging of the end records

As with non-autostored spectra, it is necessary that the filter type and bandwidth, and highpass and lowpass filters used for the autostored measurements be the same. In addition, it is necessary that the number of spectra in each record be the same. The average is a spectrum-by-spectrum average, meaning that if there are M autostore records being averaged, the Nth frequency spectrum in the averaged record represents the energy average of the M different Nth spectra, one per record.

Each spectrum in an autostore byTime record has associated with it a time index. When averaging is performed, the M values of time associated with the Nth spectrum in each record are averaged together to produce an averaged value to assign to the Nth spectrum in the averaged record. The user is cautioned to think carefully when performing averaging of autostored blocks in order to understand just what the result may mean physically. In the case of sound decay measurements where the same values of time interval Delta Time and time period End Time are used, and if the time delay between the beginning of the analysis and the shutoff of the time generator is consistent between tests, then the time indices for all M of the Nth spectra will be nearly the same, and the averaged value of the time index assigned to the Nth spectrum in the average record will be meaningful.

One could imagine, however, an instance where the interval Delta Time used for spectral storage of the different autostore records were vastly different due to the measurement setups being different, yet the number of spectra per record happen to be the same. The averaging process would calculate values of the Time Index for each spectrum in the average record, as described above, but the result would be meaningless.

Block Maximum of Autostored byTime Records

The Block Maximum operation can be applied to Autostored byTime records of the same type and bandwidth which have been stored sequentially (record numbers in a sequence). As explained above, the result of the Block Averaging operation is a similar autostore record where the Nth spectrum is the average of all the Nth spectra contained in the separate autostore records being averaged. The Block Maximum operation is similar, except that for each frequency band in the Nth spectrum, the amplitude is that of the highest level occurring at the same frequency across all the Nth spectra in the separate autostore records rather than their average. To perform the Block Maximum operation, from the Recall Menu press **B.MAX [C]**, which will bring to the upper right of the screen the following message:

MAXIMUM: 0001 - 0002

Use the numeric keypad and the horizontal arrow keys to enter values representing the first and the last of the sequence of record numbers over which the block maximum operation is to be performed. Use the **Last N [H]** softkey to take you to the end of the data block. Toggling this key will switch between **Start/E [H]** and **Last N [H]**. **Start/E [H]** presents the entry field with the start to end records to which the Block Maximum function will be applied. The Autostore Block Maximum operation is limited to a maximum number of sequential records of twenty. Upon pressing **EXIT**, the operation will be performed and the resulting spectrum stored.

NOTE: The word MAXIMUM appears on the right of the screen, 2nd line down, in place of the elapsed time usually displayed with a measured spectrum, to indicate that this spectrum is the result of the Block Maximum operation. If the records contained within the specified range are not all of the same type and bandwidth, the Block Maximum operation will not be completed, and the message "NOT SIMILAR DATA" will appear on the upper right of the screen.

Block Summation of Autostored byTime Records

Like the Block Average and the Block Maximum operations, the Block Summation operation can be applied to Autostored byTime records of the same type and bandwidth which have been stored sequentially (record numbers in sequence). The result is a similar autostore record where the Nth spectrum is the Root Mean Square (RMS) sum of all the Nth spectra contained in the separate autostore records being summed. This is the proper sum to utilize when adding decibels. To perform the Block Summation operation, from the Recall Menu press **B.SUM [D]**, which brings to the upper right of the screen the following message:

RMS SUM: 0001 - 0002

Use the numeric keypad and the horizontal arrow keys to enter values representing the first and last sequence of record numbers over which the summation is to be performed. Use the **Last N [H]** softkey to take you to the end of the data block. Toggling this key will switch between **Start/E [H]** and **Last N [H]**. **Start/E [H]** presents the entry field with the start to end records to be summed. The autostore block summation operation is limited to a maximum number of sequential records of twenty. Upon pressing **EXIT**, the operation will be performed and the resulting spectrum stored.

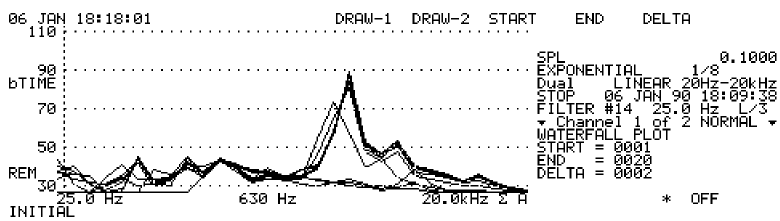
NOTE: The word SUMMED appears on the right of the screen, 2nd line down, in place of the elapsed time usually displayed with a spectrum, to indicate that this spectrum is the result of a block summation operation. If the records contained within the specified *range are not all of the same type and bandwidth, the block summation operation will not be completed and the message "NOT SIMILAR DATA" will appear on the upper right of the screen.*

Waterfall Display of Autostored Records

The waterfall display function permits the sequential display of a series of individual spectra within a byTime type autostored record, each one remaining on the screen after it has been displayed. Thus, we will see drawn upon the screen one spectrum, then overlaid upon that another spectrum, then another, etc.

Access the Recall Menu by pressing **RECALL** and use the **A.Prev [N]** and **A.Next [O]** keys to recall the record number from which the spectra are to be displayed. Then press **W.Fall [A]** which will bring up the Waterfall Menu, shown in Figure 15-6.

Figure 15- 6 ***Waterfall Menu***



On the right of the screen we see a table indicating the present values of START, END and DELTA. These represent the first and last spectra in sequence which are to be displayed, and the incremental record number between displayed spectra, respectively. For example, using the following combination:

START	=	0010
END	=	0020
DELTA	=	0002

The spectra displayed will be numbers 10, 12, 14, 18 and 20 in sequence.

To edit any of these numbers, press **START [E]**, **END [F]** or **DELTA [G]**. This will produce the message

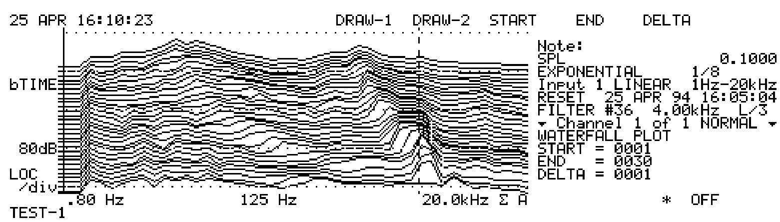
"W.FALL sXXXX,eXXXX,ΔXXXX"

with a flashing cursor to denote where inputs from the keypad will begin. The only difference between initiating this input with the **START [E]**, **END [F]** or **DELTA [G]** softkeys is that the flashing cursor will be positioned for immediate editing of the START, END or DELTA values, respectively. Use the numeric keypad and the horizontal arrow keys to edit the values as required and press **EXIT**. The display sequence will begin immediately upon pressing that key.

There are two formats available for the presentation of the waterfall plots. In the two dimensional format, produced by pressing **DRAW-1 [C]**, the spectra are simply overlaid, one at a time, without any offsets in the vertical and horizontal directions. This produces a graphic as shown in Figure 15-6.

In the three dimensional format, produced by pressing **DRAW-2 [D]**, an offset in both the vertical and horizontal directions is added to each successive spectrum curve, providing perspective to the view.

Figure 15-7 ***Waterfall Menu: 3D Format***



Usually, one begins by displaying all the spectra within the record using a large enough spectral increment number that the drawing does not take too long. Then, based on the observations of the display sequence, the range of spectra is reduced to a sequence of particular interest, with a smaller increment to produce more details of the spectral changes over that range of spectra.

CHAPTER

16 Autostore by Tach

Tachometer Input (TACH)

On the top panel of the Model 3000+ is a connector labeled TACH INPUT. This is designed to be used with a tachometer which outputs an analog pulse train whose frequency is proportional to the rotation rate of a rotating machine. The 3000+ detects the frequency of this pulse train and displays the value digitally on the lower right of the screen in the format "TACH = XXX.X". The frequency can be scaled using internal software, permitting the display of frequency in units other than Hz, such as RPM. This is also useful for cases where there are more than one pulse per revolution of the machine. We recommend the Larson Davis Model TAC100 Tachometer, which is battery operated and works with inductive, optical and piezoelectric pickups.

The user can also select to direct the signal at either of the signal inputs to the tachometer.

Second Tachometer Input (SPEED)

There is a second tachometer input available on the I/O Port pin 5 which works totally independent from the signal on the TACH INPUT connector. The software-scaled value of the frequency of this pulse train is also displayed on the lower right of the 3000+ in the format "SPEED = XXX.X". The name derives from the most common application, a vehicle test in which the TACH INPUT is a tachometer signal from the engine and the other input is from a tachometer using an optical pickup aimed at a reflective line on a tire, thereby producing a frequency proportional to vehicle speed. However, this is really just another tachometer channel, so it need not be measuring speed in spite of the label used in the display.

TACH/SPEED Display in Intensity Mode

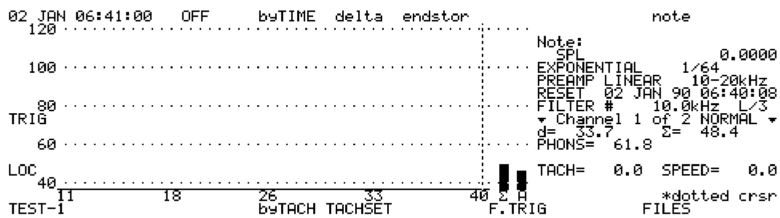
When in the Intensity mode, there is normally insufficient space on the right side of the screen for the display of the TACH and SPEED values due to the space taken up by the JOB:PART:AREA names. However, since there is not a power summation for intensity data taken in the vsTach autostore mode, there is really no practical use for these names. In order to display the TACH and SPEED data in the Intensity Mode, from the Main Menu press **SHIFT** and **TACH/JB [G]**. This will replace the display of PART: and AREA: with the field "TACH: X.X SPEED: Y.Y". Repeating this key sequence will return the right side of the screen to the usual Intensity Mode format.

byTach Autostore

Using the scaled values of RPM and Speed, the 3000+ can autostore spectra at regularly spaced increments of RPM and Speed, beginning and ending at user specified values of both parameters. The autostore may be done using either octave bandwidths or FFT frequency analysis. A particular advantage of using the octave bandwidths is that digital filter spectra are produced much more rapidly from the processor than are FFT data blocks. One can store 1/3 octave spectra as fast as 400 spectra/second while successive FFT spectra will be at least 40 milliseconds apart.

The Autostore byTach function is controlled from the same Autostore Menu used for Autostore byTime. This Menu, shown in Figure 16-1, is accessed from any of the Analysis Menus by pressing **AUTOSTR [P]**.

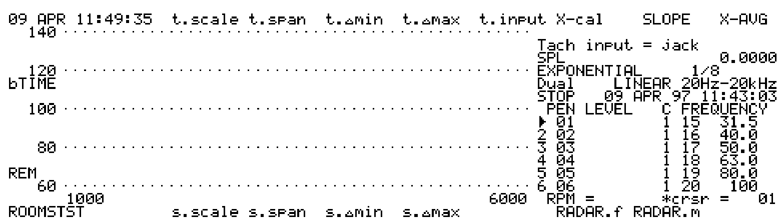
Figure 16- 1 ***Autostore Menu***



Setting the Tacho Parameters

The Tacho parameters are set from the Tacho Menu, shown in Figure 16-2, which is accessed from the Autostore Menu by pressing **TACHSET [K]**.

Figure 16- 2 ***Tachset Menu***



Selection ofTacho Input

Repeated presses of the softkey, **t.input [E]** will change the input connection to the tachometer between “jack”, “CH1”, and “CH2”, as indicated by the message on the upper right screen.

Tach/Speed Scaling

The pulse train signal (pulses/sec) applied to each of the hardware inputs, TACH and SPEED, is detected by the 3000+ as a frequency (Hz). For example, if the signal input to the TACH input represented a single pulse per revolution of a shaft, the units of TACH as read by the 3000+ would be shaft speed in Hz (rev/sec). Often one would prefer other units, such as RPM, and also there may be more than a single pulse per revolution. In the Model 3000+, the pulse rate is multiplied by a user-defined scale factor (default value of unity) to permit the use of a variety of units.

Example: To detect the shaft rotational rate, a probe is used near a gear on the shaft. The gear has 32 teeth, thereby producing a pulse train at the rate of 32 times the shaft speed. The desire is to calibrate the 3000+ so that the TACH value is measured in units of Hz (rev/sec).

Set the Scale factor = 1/N = 1/32 = 0.03125 rev/pulse

Signal into the TACH module = X Pulse/sec

Scaled value = X pulse/sec $\times$ 0.03125 rev/pulse
= 0.03125 $\times$ X rev/sec

Example: The axle rotation is monitored. Every rotation the axle moves $2 \times \pi \times r$ ft., which equals 6.28r ft. The desire is to calibrate the 3000+ so that the Speed value is measured in units of ft/sec.

Set the Scale factor = 6.28r, which has units of ft/rev

Signal into the module = X pulse/sec

Scaled value = X pulse/sec $\times$ 1 rev/pulse $\times$ 6.28r ft/rev
= 6.28r $\times$ X ft/sec

Example: The gear in Example 1 is on the axle in Example 2. The desire is to calibrate the 3000+ so that the Speed value is measured in units of ft/sec.

Set the Scale factor = 0.03125 rev/pulse $\times$ 6.28r ft/rev
= 0.196 ft/pulse

Signal into the module = X pulse/sec

Scaled value = 0.196 ft/pulse $\times$ X pulse/sec
= 0.19625 $\times$ X ft/sec

The following keys are used to set the Tacho and Speed Scaling:

Softkeys	Softkey Functions
t.scale [A]	This is the user-defined scale factor which converts the pulse rate to the desired Tach frequency units, as described in the preceding examples. The format is exponential: X.XXX E+XX
s.scale [I]	This is the user-defined scale factor which converts the pulse rate to the desired Speed frequency units, as described in the preceding examples. The format is exponential: X.XXX E+XX

Whenever the analyzer is running, the scaled values of TACH and SPEED are displayed in real-time on the lower right of the screen.

Interval and Span Settings

The purpose of the Autostore byTach function is so that a series of spectra can be stored automatically at user-defined intervals of RPM and/or Speed as a vehicle or machine is accelerating or decelerating. The intervals for RPM and Speed are set independently by the user, who can also select a slope (+,- or +/-) for the interval sequence.

The autostorage function is independent of the data acquisition. As an analyzer, the 3000+ will be operating in a normal manner during an autostore sequence, producing new spectra at regular intervals of time based on the selection of the analysis type and the averaging method and time. See the preceding chapter for more information on averaging time considerations. As each spectrum is produced from the processor, the values of TACH and SPEED are looked at to determine if a spectrum storage is called for by either of these parameters. If so, the data block in the buffer is stored along with the current values of TACH and SPEED; if not, the system waits for the next spectrum to be produced.

Spectra are produced from the math processor at discrete time intervals. When fractional octave filtering is used, the

Tach and Speed values are read at 2.5 ms intervals. Using FFT filtering, these values are read at the same rate as the FFT spectral blocks are produced (100 line, 10 ms; 200 line, 20 ms; etc.). For this reason, it is not possible to ensure that data storage will take place at precisely defined intervals of Tach or Speed because no spectra might appear which happen to have exactly these Tach or Speed values. For this reason, we define both minimum and maximum values of the increments of Tach and Speed for which data storage is to occur. t.Δmin and s.Δmin represent the increments of Tach and Speed for which we would like data storage to occur. t.Δmax and s.Δmax represent maximum acceptable values of these intervals for which the autostorage is to continue in a normal manner.

Consider a case where we have set t.Δmin to 50 RPM and t.Δmax to an extremely high value such that it is unlikely ever to occur, such as 50,000 RPM. We have set the Slope to +, and the last data storage was at 1,000 RPM. The next spectrum stored will be the first one for which the RPM value is 1,050 OR GREATER (unless it exceeds 50,000). It may be that the first spectrum satisfying that condition will correspond to an RPM of 1,058, in which case that spectrum will be stored and the system will now seek another spectrum whose interval of RPM is 50 or more greater than 1,058.

The values of t.Δmax and s.Δmax are used for situations where the Tach or Speed values change so rapidly that the interval values of Tach/Speed for which data storage actually occur could become unacceptably greater than those specified by t.Δmin and s.Δmin. Or, there could be spikes on the Tach/Speed inputs which would result in data storage corresponding to the values produced by these spikes. t.Δmax and s.Δmax are defined to limit on the high side the acceptable range of increments for Tach and Speed. Data storage requires that the measured spectrum have incremental values of Tach and/or Speed with respect to those of the spectrum last stored falling within these minimum and maximum values.

Consider the case of a vehicle acceleration where the Tach interval values have been set to t.Δmin = 5 and t.Δmax = 10. After each autostorage operation, a spectrum with a value of Tach falling between these limits must occur if the autostorage is to continue in a regular manner. Suppose that a spectrum was stored having a Tach value of 4,000 and that due to a particularly rapid acceleration the next spectrum produced corresponded to a Tach value of 4,015. No spectrum storage would occur until the driver slowed the vehicle sufficiently that a spectrum was measured having a Tach value between 4,005 and 4,010. He could then resume the acceleration and continue the autostorage sequence.

In most tests where the byTach autostore is to be used, there is a particular range of RPM or Speed over which the data is of interest. In the Model 3000+ this is defined as the span of interest, and since there are two possible independent parameters, RPM and Speed, the user can define both an RPM Span and a Speed Span. Each Span will be defined by a LOW and a HIGH value of RPM or Speed.

Influence of Slope on Test Procedure

The positive slope condition (SLOPE = +) is appropriate for a vehicle acceleration or a machine runup. In this case, it is necessary that at the moment of initiation of the test sequence either the RPM or the Speed value, or both, be below the LOW value of the corresponding Span range. As the test proceeds and the RPM and Speed values increase, autostorage will be initiated when either of these variables increases sufficiently to exceed the LOW value, thus falling within the Span. Autostorage will then continue at positive incremental values RPM and Speed corresponding to t.Δmin and s.Δmin. There will be no storage for negative incremental values of RPM or Speed. When either RPM or Speed have increased enough to exceed the HIGH limit of the corresponding Span, the autostore sequence is stopped automatically.

The negative slope condition (SLOPE = -) is appropriate for a vehicle deceleration or a machine coast down. In that case, the situation is the reverse of that for the positive slope. Either RPM, Speed or both must be above the HIGH value of the corresponding Span at the initiation of the sequence, and autostorage will begin when one of these falls within the appropriate Span range. Autostorage will then occur at negative incremental values of RPM and Speed until one of them falls below the LOW value of the corresponding Span, at which time the autostorage sequence will be stopped.

The positive/negative slope condition (SLOPE = +/-) is appropriate for a vehicle acceleration followed by a deceleration, or a machine runup followed by a coastdown. In the first phase, the procedure is the same as that for a positive slope condition, in that either or both RPM and Speed must be below their LOW Span values at the time of initiation of the test and once one of these values moves into its Span range, autostorage will occur at positive incremental values until one of them exceeds its HIGH value, thus moving out of the Span range. This completes the acceleration/runup phase. In the second phase, the procedure corresponds to a negative slope condition and the deceleration/coastdown is begun. As the RPM and Speed decrease such that one or both values fall within their Span range, data will be autostored at negative incremental values of RPM and Speed until either of them falls below the corresponding LOW value, at which time the autostorage sequence is stopped. If neither the RPM nor the Speed values exceeds the HIGH limit of their Spans before the second phase (deceleration/ coastdown) is begun, the sequence will not be stopped automatically and it will be necessary to press the **RUN/STOP** key to stop the autostore sequence.

The following keys are used to set the Span and Interval values and the Slope setting:

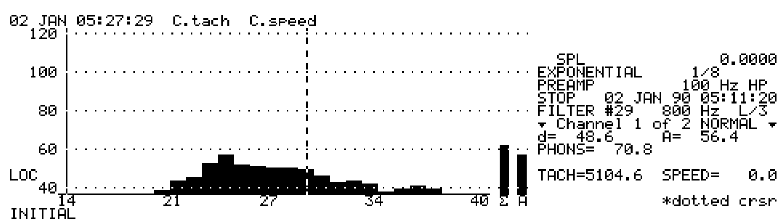
Softkeys	Softkey Functions
t.span [B]	RPM Span for which data storage based on Tach value will occur; format is “LOW/HIGH”
t.∆min [C]	Minimum Tach interval for which data storage will occur
t.∆max [D]	Maximum Tach interval for which data storage will occur
s.span [J]	Speed Span for which data storage based on Speed value will occur; format is “LOW/HIGH”
s.∆min [K]	Minimum Speed interval for which data storage will occur
s.∆max [L]	Maximum Speed interval for which data storage will occur

Softkeys	Softkey Functions
SLOPE [G]	Repeatedly pressing this key will change the sign associated with increments for the storage of data between +, -, and +-, as indicated on the upper right of the screen.

Tach/Speed Calibration

There is another manner in which the scaling of the Tach and Speed signals can be performed dynamically based on the state of an operating vehicle or machine. From the Tachset Menu access the Tach/Speed Calibration Menu, shown in Figure 16-3, by pressing **X-cal [F]**.

Figure 16- 3 ***Tach/Speed Calibration Menu***



As described in the preceding section, the fundamental readout of the Tach and Speed is in units of frequency representing the number of pulses per second seen at the two inputs. The scaling is done to convert these frequencies to alternative units such as RPM (Tach) or Miles/Hour (Speed). In the calibration method, the user specifies a value of Tach or Speed (or both) which he wishes the 3000+ to display at the instant he manually initiates the calibration procedure. The scaling is then automatically performed such that the frequency measured at the input(s) will produce the specified value(s) on the screen.

For example, suppose the user wishes to drive a vehicle at a speed of 50 miles/hour as indicated by the speedometer and perform the Speed calibration at that moment. He would press **C.speed [B]** and in response to the message “ENTER SPEED” on the upper right of the screen, type 50.0 using the keypad and press **EXIT**. Now whenever the 3000+ sees a sequence of two presses of the **RUN/STOP** key the Speed scaling will be set such that the display readout of Speed is 50. He will now press **RUN/STOP** once to begin the analysis and drive the vehicle until a speedometer reading of 50 miles/hour is obtained. At this point the readout of Speed will be in units of frequency. He then presses **RUN/STOP** a second time to perform the calibration and to stop the analysis. The 3000+ will now be calibrated to display Speed in units of miles/hour for that particular vehicle.

Use the **C.tach [A]** key to preset a value of the Tach readout to be calibrated for a particular machine condition. Then following two successive presses of the **RUN/STOP** key the Tach scaling will be set such that the readout will indicate that preset calibration value for whatever frequency was being read at the Tach input at that instant.

If both the **C.tach [A]** and **C.speed [B]** keys are used to input calibration values after accessing the Tach/Speed Calibration Menu, then both will be scaled to these calibration values following two successive presses of the **RUN/STOP** key.

Setup of Radar Communication (Passby Option required)

For those of you who have the Passby option installed on your analyzer the Tachset menu includes two softkeys, **RADAR.f [N]** and **RADAR.m [O]**. Pressing **RADAR.f [N]** will produce the message "START X = -0100". This allows you to enter the number of feet between the start of the test section where the first photoelectric switch is located and the center position of the test section. Pressing **RADAR.m [O]** allows you to enter the value in meters instead of feet. Press **EXIT** when finished entering the values.

Trigger Smoothing

In some instances, the signal from the tachometer pickup may develop some FM “jitter” due to vibration of the rotating structural element of the machine under test which is being used for detection of the tacho signal. An example of this would be a tire upon which a white target line has been drawn to trigger a signal from a photo-electric sensing probe. At high speeds, tire vibrations would produce such an effect on the signal.

To improve the performance under these circumstances, the digital treatment of the Tach/Speed signals includes averaging algorithms. Access the Tacho/Speed Averaging Menus, shown in Figure 16-4 and Figure 16-5, from the Tachset Menu by pressing **X-AVE [H]**.

Figure 16- 4 ***Tach/Speed Linear Averaging Menu***

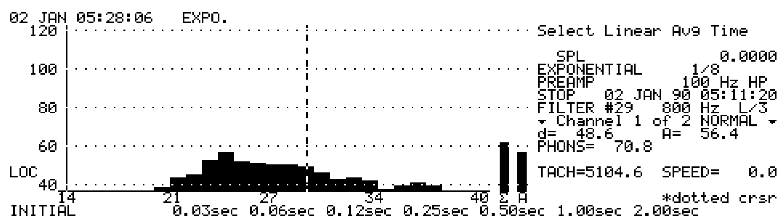


Figure 16- 5 ***Tach/Speed Exponential Averaging Menu***

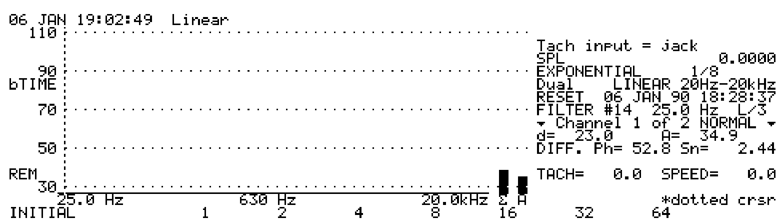


Figure 16-4 represents the Linear Averaging Mode and Figure 16-5 represents the Exponential Averaging Mode. The message at the upper right of the screen will indicate which of this is being displayed. Pressing the key **[A]** (which may be labeled **Linear** or **EXPO.**) will switch the display between the two. The message on the upper right of the screen will indicate the presently active averaging time for that averaging type. Select the Menu corresponding to the desired averaging type using the softkey **[A]**. Then, press one of the softkeys along the bottom row to select the desired averaging time and press **EXIT** to return to the Tachset Menu.

The linear averaging times are expressed in units of seconds. The exponential averaging times specify the averaging time constant by its corresponding (equivalent) base 2 exponent value (1 to 64).

In general, the user should select as small an averaging time as possible as long as stable trigger operation is obtained. The main detrimental effect of averaging is that the averaged value will lag behind the instantaneous value by a degree related to the amount of averaging and the slew rate of the tacho pulse rate. As a result, the tacho and speed values stored along with the spectra will be slightly different than the true values occurring at the instant of storage.

When the averaging of the Tach/Speed signals has been defined, press **EXIT** to return to the Tachset Menu.

Enabling Autostore byTach

From the Autostore Menu, enable Autostore byTach by pressing **byTACH [J]**.

The message “bTACH” will appear on the left of the screen to indicate that the Autostore byTach mode is active. Press **RUN/STOP** to initiate operation.

If the complete test sequence (variation of RPM/Speed) corresponding to the choice of the Slope parameter proceeds as described earlier, the autostorage sequence will be automatically stopped. Once the autostorage has begun, however, the user can stop the sequence at any time by pressing **RUN/STOP**. And in cases where the sequence has begun but the parameters do not satisfy the requirements for automatic termination of the autostorage sequence, the manual stop will be required.

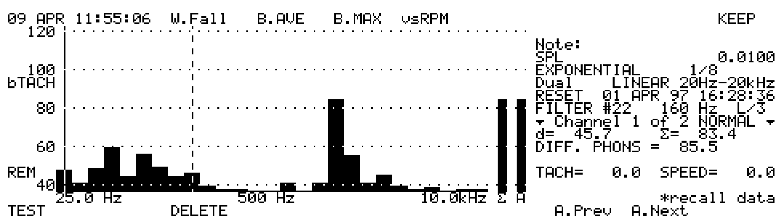
Recall of Data Autostored byTach

Pressing the hardkey **RECALL** while the 3000+ is in the autostore byTach mode will recall one of the By Tach type records from the active memory file whose name is listed on the lower left of the screen. The message “RECALL - By Tach   N” on the upper right of the screen will indicate that the Nth record of the type By Tach has been recalled. In most cases this will be the record number which was last stored or recalled. To determine how many By Tach records have been stored in a particular record, and to examine their note fields, use the Files Menu.

If the desired record is in another memory file, it will be necessary to access the Files Menu, change the active memory file and exit before performing the recall operation.

Upon pressing the **RECALL** key, the Recall Menu shown in Figure 16-6 will be displayed

Figure 16- 6 ***Autostore Recall Menu***



and the message "*recall data" on the lower right of the screen will indicate that the horizontal arrow keys are assigned to recall the individual spectra from the recalled record.

To recall an autostored record stored earlier (previous to) the one which has been recalled, press **A.Prev [N]** and note that the index N in the message on the upper right has been decreased by one, indicating that the previous record has now been recalled. Repeated presses of **A.Prev [N]** will page the recall procedure continually towards the first record stored in that file.

Similarly, pressing **A.Next [O]** will result in the recall of the autostored record which was stored later (after) the one which had originally been recalled, as indicated by a unity increase in the value of N in the message on the upper right. Repeated presses of **A.Next [O]** will page the recall procedure continually towards the last record stored in that file.

Displaying Individual Spectra

Once the desired record has been recalled, presses of the right arrow key will page through the individual spectra contained in the autostore record, bringing them sequentially to the screen. Each spectrum is tagged with the time relative to the initiation of the autostore sequence. This is displayed on the right screen, first line down. The values of Tach and Speed corresponding to the instant of spectrum storage are also displayed at the lower right of the screen.

Presses of the left arrow key will produce a paging backwards in sequence toward the first spectrum stored.

Channel Selection

When the autostore operation was performed with two channels active, there will be a complete set of spectral data for each channel. To select the input channel whose data are to be displayed, use the hardkeys **CH1** and **CH2** and note on the right of the display, sixth line down, the change in the indicated channel number.

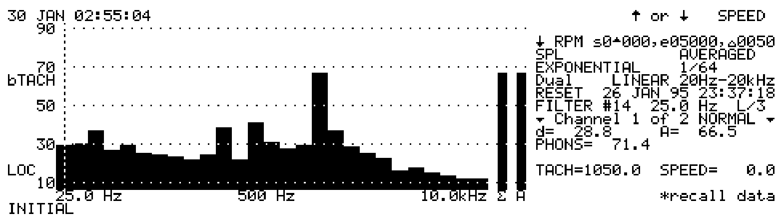
Cursor Control

To utilize the cursor for readout of the amplitude and frequency values of the displayed spectrum, press **CURSOR** which will assign the horizontal arrow keys to control the cursor which was last active (dotted or solid). A second press of that key will bring up the Cursor Menu for selection of cursor type.

Averaging of Autostore byTach Records

It is possible to average together a series of byTach autostore records when they have been stored in sequential records. To initiate this operation, recall one of these records and from the Recall Menu press **B.AVE [B]**, producing the byTach Block Averaging Menu as shown in Figure 16-7.

Figure 16- 7 ***byTach Block Averaging Menu***



The “RPM” in the message on the upper right of the screen indicates that the records will be averaged in terms of the parameters associated with the vsRPM storage format. In order to average the records in terms of the parameters associated with the vsSpeed format, press the softkey **SPEED [H]**, which will change this “RPM” to “SPEED”. To change back to “RPM”, press the softkey **TACH [H]**.

The direction of the vertical arrow at the beginning of the message on the upper right of the screen must match the direction of the autostored data being averaged (upwards if the RPM/Speed values are increasing during the test, downwards if they are decreasing), otherwise the averaging will not be performed. The softkey **[G]** is used to change the direction of the arrow in the message. If the test values increase up to the upper limit, then decrease back down to the lower limit, the averaging is performed using data for either the upwards portion or the downwards portion, corresponding to the direction of the arrow at the beginning of the message. In this manner a test which involved both directions can be divided into two data blocks, one for each direction.

In this operation the averaging is performed over a user-selected range of RPM or Speed, utilizing a user-specified increment (or step size) of RPM or Speed. The message displayed on the screen indicates the start value of RPM or Speed over which the averaging is to be done (indicated by the “s”), the end value (indicated by the “e”) and the incremental value. The default values of s, e and delta, which appear on the screen, will correspond to the RPM or Speed range and incremental value utilized at the time of data acquisition. The user is not constrained to use these default values for the averaging. the values in the three display fields can be edited by using the numeric keypad and the horizontal arrow keys and pressing **EXIT**.

In each autostored record there is a range of RPM or Speed values, over which there is one spectrum per channel stored at approximately equal intervals. In general, neither the lower or upper limits of the range, nor the interval sizes themselves, will be exactly the same for different records, even though they may have been captured using the same instrument setup. When specifying the range for the averaging, it is recommended that the start and end values of RPM or Speed be chosen such that they will fall within the range of values covered by each of the autostored records. If not, the actual range of RPM or Speed values for which data will be produced in the averaged record will be reduced such that all of the individual records have data points within that range.

When the averaging is performed, extrapolation between the actual RPM or speed values in the individual autostored records is performed which permits the user to select a value of increment Δ different than that which may have been programmed during the data acquisition process. The result is that the spectra in the averaged block will appear at precisely the requested RPM or speed increments. For example, if the data storage had been programmed to nominally store data every 100 RPM, the user can select to perform the average using an increment of 10 RPM, providing better resolution over the range of the test data. This also means that the averaging process can be utilized on a single record to “normalize” the RPM or Speed increments in the data block. As was explained earlier, in practice the data will not be stored at exactly the specified increments of RPM or Speed. By performing an average of this record using the same value of RPM or Speed increment originally specified for the autostore, the resulting spectra will appear at precisely the desired increments.

Once the values of s, e and delta have been set as desired and entered by pressing **EXIT**, the following message will appear on the screen to prompt the user to specify the sequence of record numbers over which the average is to be performed:

AVERAGE: 0001 - 0002

Use the numeric keypad and horizontal arrow keys to edit this field as desired and press **EXIT** to perform the average. In the case where one wishes to simply “normalize” the RPM or Speed increment of a single record as described above, set both fields to the same record number. The resulting record will be stored in the next available record of the type By Tach and then recalled, as indicated by the message

“RECALL byTach - N”

on the upper right of the screen, where N is the record number into which it was stored. Note that when displaying the first spectrum in sequence within the block, the note “AVERAGED” is displayed on the first line on the right of the screen to indicate that this data block was the result of an averaging process and not the result of an actual measurement.

Block Maximum of Autostored byTach Records

The Block Maximum operation can be applied to Autostored byTach records of the same type and bandwidth which have been stored sequentially (record numbers in a sequence). As explained above, the result of the Block Averaging operation is a similar autostore record where the Nth spectrum is the average of all the Nth spectra contained in the separate autostore records being averaged. The Block Maximum operation is similar, except that for each frequency band in the Nth spectrum, the amplitude is that of the highest level occurring at the same frequency across all the Nth spectra in the separate autostore records rather than their average. To perform the Block Maximum operation, from the Recall Menu press **B.MAX [C]**. The same sequence of messages, prompting for user input, will appear as they do for the Averaging operations described in the preceeding section. The Autostore Block Maximum operation is limited to a maximum number of sequential records of twenty. Upon pressing **EXIT**, the operation will be performed and the resulting spectrum stored.

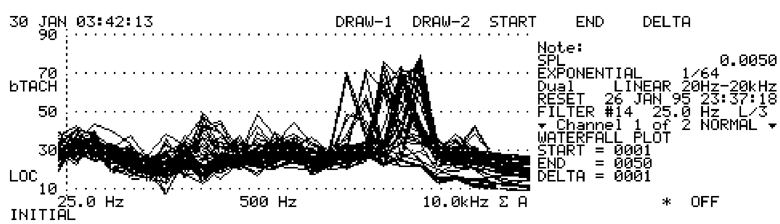
NOTE: The word MAXIMA appears on the right of the screen, 2nd line down, in place of the elapsed time usually displayed with a measured spectrum, to indicate that this spectrum is the result of the Block Maximum operation. If the records contained within the specified range are not all of the same type and bandwidth, the Block Maximum operation will not be completed, and the message "NOT SIMILAR DATA" will appear on the upper right of the screen.

Waterfall Display of Autostored Records

The waterfall display function permits the sequential display of a series of individual spectra within a By Tach type autostored record, each one remaining on the screen after it has been displayed. Thus, we will see drawn upon the screen one spectrum, then overlaid upon that another spectrum, then another, etc.

Access the Recall Menu by pressing **RECALL** and use the **A.Prev [N]** and **A.Next [O]** keys to recall the record number from which the spectra are to be displayed. Then press **W.Fall [A]** which will bring up the Waterfall Menu, shown in Figure 16-8.

Figure 16-8 ***Waterfall Menu***



On the right of the screen we see a table indicating the present values of START, END and DELTA. These represent the first and last spectra in sequence which are to be displayed, and the incremental record number between displayed spectra, respectively. For example, using the following combination:

START	=	0010
END	=	0020
DELTA	=	0002

The spectra displayed will be numbers 10, 12, 14, 18 and 20 in sequence.

To edit any of these numbers, press **START [E]**, **END [F]** or **DELTA [G]**. This will produce the message

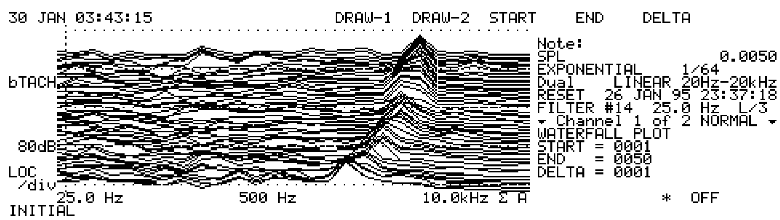
“W.FALL sXXXX,eXXXX,ΔXXXX”

with a flashing cursor to denote where inputs from the keypad will begin. The only difference between initiating this input with the **START [E]**, **END [F]** or **DELTA [G]** softkeys is that the flashing cursor will be positioned for immediate editing of the START, END or DELTA values, respectively. Use the numeric keypad and the horizontal arrow keys to edit the values as required and press **EXIT**. The display sequence will begin immediately upon pressing that key.

There are two formats available for the presentation of the waterfall plots. In the two dimensional format, produced by pressing **DRAW-1 [C]**, the spectra are simply overlaid, one at a time, without any offsets in the vertical and horizontal directions. This produces a graphic as shown in Figure 16-8.

In the three dimensional format, produced by pressing **DRAW-2 [D]**, an offset in both the vertical and horizontal directions is added to each successive spectrum curve, providing perspective to the view.

Figure 16-9 ***Waterfall Menu: 3D Format***



Usually, one begins by displaying all the spectra within the record using a large enough spectral increment number that the drawing does not take too long. Then, based on the observations of the display sequence, the range of spectra is reduced to a sequence of particular interest, with a smaller increment to produce more details of the spectral changes over that range of spectra.

vsRPM Graphics

The 3000+ provides a unique vsRPM Graphic capability which can be used with byTach autostored data. This permits the generation and display of a number of curves in the format amplitude versus RPM or Speed where each curve corresponds to a specific channel and frequency (or order).

When octave bandwidths or standard FFT spectra have been autostored, the graphic may be in the form of frequency versus RPM/Speed or order versus RPM/Speed. The latter is obtained using a Post-process order tracking.

A detailed description of the vsRPM Graphic capability is presented in Chapter 17.

CHAPTER

17 vsRPM Graphics

As explained in Chapter 16, Autostore by Tach, it is possible to read the RPM and Speed values of a vehicle or machine during a test and to autostore spectra at regular intervals of RPM or vehicle speed using the byTACH storage mode. The vsRPM Graphics capability permits the 3000+ to simultaneously display several different curves in an amplitude versus RPM/Speed format. Each curve would represent a particular channel and frequency band (or order value), user-definable.

When octave bandwidth or standard FFT analysis is utilized, the spectral data can be used directly to produce curves of frequency band versus RPM/Speed. However, by using a procedure which we refer to as post-process order tracking, these curves can also be generated in the form of order versus RPM/Speed. This procedure is explained in detail later in the chapter under the section Post-process Order Tracking.

There are two different modes of operation of the vsRPM Graphics:

Real-time vsRPM Graphics

A. Without utilizing the autostore capability directly, the system can be set up to measure and plot the curves representing selected data as a function of RPM or Speed live on the display as the test takes place. For example, if a machine is run-up from a low to a high value of RPM, the curves would be drawn on the display from left to right as the test proceeds. Only spectra measured in the Standard Analysis Mode can be handled in this manner. The data corresponding to these curves can be stored to memory for subsequent recall, viewing and printing. The spectra may be autostored in the byTach mode at the same time, but this is not required.

vsRPM Graphics from byTach Autostored Records

B. When the byTACH autostore capability have previously been utilized to measure and store Standard and Intensity spectra as a function of RPM/Speed, the vsRPM Graphics can be used in conjunction with the stored data to generate sets of curves as a function of RPM or Speed. Each set of curves generated by this procedure can be stored to memory for subsequent recall, viewing and printing. Note that the Real-time vsRPM display mode can be active during a test in which the byTACH autostorage procedure is used to store data as a function of RPM/Speed. During the test the data can be observed in real time on the screen as the test proceeds, permitting the test engineer to verify that the test has proceeded as desired.

Real-time vsRPM Graphics

The first step in performing Real-time vsRPM Graphics is to setup the 3000+ to the desired measurement configuration, including number of channels, analysis type, filter type, averaging type and time, etc. The 3000+ must be in the Standard Analysis mode. It is not necessary to activate the byTach autostore mode to perform real-time vsRPM Graphics. However, the user may select to perform a vsRPM autostore at the same time as generating a real-time vsRPM display. The two procedures are totally independent.

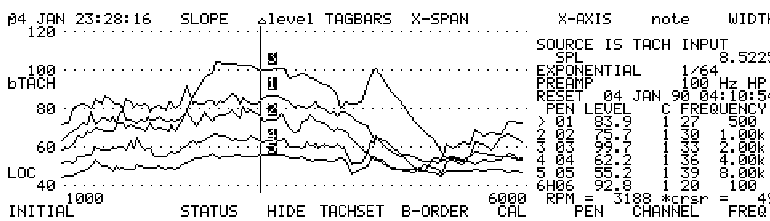
LCD Display Pen Format

When working with the LCD display of the 3000+ the user can generate only a single set of up to six different traces at a time. Because there is only one set of traces, there is no reason to assign a particular pen to more than one trace. As a result, there really is no justification for selecting pen numbers for each trace which are different than the trace numbers themselves. For this reason, in the following we shall make it a standard procedure to keep the pen numbers the same as the trace numbers. The preceding section was presented simply because it is possible to define them differently, and if the user chooses to do so he may.

Parameter Selection

From the Standard Analysis Menu, access the vsRPM Graphics Menu, shown in, by pressing **vsRPM [J]**.

Figure 17- 1 ***vsRPM Graphics Menu***



The parameters which define the individual traces are contained in the table on the lower right of the screen. Each row represents a specific trace (numbered 1-6) along with the presently assigned values of PEN, channel (C), and frequency (FREQUENCY) or order (ORDER) arranged in rows.

Accessing a Trace

To modify the parameters associated with a particular trace, use the numeric keypad and press the key corresponding to the number of the trace to be modified. The symbol “>” will appear to the left of that trace number to indicate that it is now the trace to which modifications will be assigned.

Pen Selection

As mentioned above, we will adopt the convention that the pen numbers shall be made the same as the trace numbers. If upon accessing the table the pen numbers do not correspond to the trace numbers, modify them to do so as follows. Access each trace and press **PEN [N]**. Use the horizontal arrow keys to modify the pen number until it corresponds to the trace number.

Channel Selection

After accessing the desired trace number, press **CHANNEL [O]** repeatedly and note that this will toggle between channels 1 and 2. Use this method to obtain the desired input channel number for each trace.

Frequency Band Selection

After accessing the desired trace number, press **FREQ [P]** and use the horizontal arrow keys to sequence through the filter center frequencies corresponding to the analysis type presently active for the 3000+. Beyond the highest center frequency will appear SUM, which represents the summation of the energy contained in all the frequency bands. Set the frequency in the table to the desired value.

If instead of **FREQ [P]** the softkey **ORDER [P]** is displayed, press **B-FREQ [L]** to change it to **FREQ [P]** before selecting the frequency band.

Order Selection

Whether the 3000+ has been configured for octave bandwidths or standard FFT filtering, it is possible to present the vsRPM curves in the form of orders rather than frequency. To do this, press **B-ORDER [L]** to change the parameter represented by the last row of the table from FREQUENCY to ORDER. Set the order number for each trace as described in the preceding paragraph. When creating order plots from standard frequency analysis measurements, it may also be desirable to set the peak search parameters. It is suggested that the user read the section Post-process Order Tracking before selecting orders when working with octave bandwidth or standard FFT filters.

RPM/Speed Selection

By pressing **X-AXIS [F]** a message on the upper right of the screen will state either “SOURCE IS TACH INPUT”, meaning that the format of the plot will be amplitude versus RPM, or “SOURCE IS SPEED INPUT”, meaning that the format will be amplitude versus Speed. Pressing **X-AXIS [F]** again will toggle between these two, as indicated by the change in the message.

Horizontal Scale Selection

Select the lower and upper horizontal scale limits by pressing **X-SPAN [D]** which will produce the message “LOW/HIGH XXXXX/YYYYY” on the upper right of the screen where XXXXX represents the left end of the screen and YYYYY the right end of the screen in either RPM or Speed scaled units, whichever is active at the time. Use the numeric keypad to enter the desired values and press **EXIT**. These lower and upper values will be displayed on the screen just below the axis.

Note that data for both RPM and Speed are saved during each test, which means that both a vsRPM and a vsSpeed plot may be displayed following a test, although only one of them may be active during the test.

Slope Selection

Repeated presses of **SLOPE [A]** will toggle the slope parameter between +, - and +/- as indicated by the message on the upper right of the screen "DISPLAY SLOPE IS SET TO XX" where XX is either +, - or +/-.

When the slope parameter is +, the curves will be drawn from left to right across the screen in a continuous manner only for positive increments of RPM/Speed. Should the RPM/Speed decrease temporarily during a test and then begin increasing, the curve generation will be seen to pause during the period the RPM/Speed is less than the maximum value previously achieved, and then will be renewed when the RPM/Speed values exceed that previous maximum value.

When the slope parameter is -, the inverse applies, and the curves will be generated from right to left corresponding to decreasing values of RPM/Speed.

When the slope parameter is +/-, the generated curves may move to the left or right across the screen, and in instance where the RPM/Speed value changes both positively and negatively during a test a loop pattern may be generated.

In some tests, it is desired to observe data produced only by increases or decreases in the RPM or Speed values. For example, during a machine run-up there may be a temporary reduction in RPM which would cause the drawn curves to loop back to the left on the screen before continuing to be drawn toward the right. One may wish not to show that portion of the curves. Or, one may wish to generate curves during an automobile acceleration and be certain that the curves will not reverse back toward the left when the automobile decelerates after the test is completed. Setting the slope parameter to + would provide the desired results.

Incremental Control of the Trace

New data points which could be used by the graphics routine are generated every time a new spectrum is produced by the processor. To limit the density of these points on the screen, new points for each curve are only generated whenever the data satisfies user selected criteria for display.

Graphically, in order to avoid having a large number of points for each curve plotted very close together when either the level values or the RPM/Speed values (or both) are not changing rapidly, the user establishes a minimum variation of each, one of which must be exceeded if a new point is to be drawn. For example, if the minimum level variation is 2.0 dB and the minimum RPM variation is 10 RPM, then a new point will not be plotted until the new data point has either a level 2.0 dB or more above or below the point plotted previously for that curve, or an RPM value 10 or more above or below the previously plotted point.

The incremental RPM/Speed value required for the generation of a new point is determined by the value of t.Δmin and s.Δmin as set from the Tachset Menu. In addition, to minimize the possibility that a sudden spike in the value of RPM/Speed might create discontinuities in the curves, the values of t.Δmax and s.Δmax are also used to define the maximum increment of RPM/Speed for which a new point is to be generated.

The incremental amplitude value required for the generation of a new point is set by pressing **Δlevel [B]**, which results in the message “ENTER dB THRESHOLD XXX.X” on the upper right of the screen. Use the numeric keypad to enter the desired value and press **EXIT**.

Control of Trace Status

The 3000+ allows 4 kilobytes of memory to the generation of the traces in the Real-time vsRPM Graphics mode. These are divided among the number of traces which are active. Each point requires 6 bytes; 2 each for the trace amplitude, the RPM value and the Speed value.

To make a trace inactive previous to a test in order to increase the amount of memory available for the remaining pens, access it via the numerical keypad and press **STATUS [I]**. The letter “U”, denoting Unassigned, will appear to the left of an inactive trace in the parameter table. Repeated presses of **STATUS [I]** will toggle the status between active and unassigned. Any combination of traces may be made inactive.

Calibration

While in the vsRPM menu a calibration facility has been added for your convenience. Upon pressing the **CAL [M]** softkey, the analyzer recalls the Setup Menu, specifically the Setup defined under the **[P]** softkey. It then enters the UNITS menu so that levels can be calibrated. To get back to the vsRPM menu simply press the **EXIT** hardkey.

Performing a Test

When the parameters have been input as required, simply press the **RUN/STOP** hardkey to begin a test. The scaled values of RPM and Speed can be read on the lower right of the display as the test proceeds. Whenever the RPM/Speed values fall between the lower and upper limits of the screen, the points for each of the curves will be drawn across the screen corresponding to the incremental RPM/Speed and amplitude levels and the slope parameter. In a typical machine runup or automobile acceleration test, the RPM will begin at a value less than the lower limit of the screen. As soon as the RPM reaches the lower limit value, the curves will begin to appear and will be drawn from left to right across the screen.

If the limit of the memory is exceeded during a test, the oldest data points will be replaced with newer ones, although the portions of the curves corresponding to the older points already drawn of the screen will remain.

At the conclusion of the test, press **RUN/STOP** to stop the data acquisition and graphics generation.

Examination of the Traces

At the conclusion of the test, the cursor can be used to examine the data point by point. Level values for each trace corresponding to the cursor position are presented in the parameter table. The value of RPM/Speed and the point number for the cursor position are displayed on the lower right of the screen. Numbers travel with the cursor on the screen to identify the individual traces. Pressing **X-AXIS [F]** will switch the format between vsRPM and vsSpeed.

Hiding Traces

In many cases it may be desirable to improve the readability of the Trace Display by removing, or hiding, one or more traces from the screen. A trace is hidden by accessing it with the numeric keypad and pressing **HIDE [J]**. The fact that a pen is hidden is indicated by a letter "H" to the right of the trace number in the parameter table. A second press of **HIDE [J]** will "unhide" the trace, or cause it to reappear on the screen.

Storage of Trace Displays

At the conclusion of a test, pressing **STORE** will result in the storage of the Trace Display presently on the screen. To store both vsRPM and vsSpeed Trace Displays, they must each be displayed and stored. The message "STORE - vsRPM Trace N" on the upper right of the screen will indicate the Trace Display has been stored to the active memory file as the Nth record of the type vsRPM Trace.

The data representing all traces, hidden or not, are stored along with the parameter table.

Recall of Trace Displays

To recall a Trace Display, the autostore function must be off and the 3000+ in the vsRPM Menu. If either byTime or byTach autostore is active at this time these will override the fact that the vsRPM Menu is being displayed, the data records for that type will be recalled instead.

Press **RECALL** to recall a Trace Record, which will produce the message “RECALL - vsRPM Trace N” on the upper right of the screen to indicate that the Nth record of type vsRPM Trace has been recalled from the active memory file. Use the horizontal arrow keys to recall other vsRPM Trace records from the active memory file.

vsRPM Graphics from byTach Autostored Records

Standard Mode Data

The procedure for generating vsRPM Graphics from byTach autostored records is not greatly different from that used for the Real-time vsRPM Graphics. The first step is to activate the byTach autostore mode, press **RCL** and use the **A.Prev [N]** and **A.Next [O]** keys to recall the desired autostore records. Then press **vsRPM [D]** to access the vsRPM Graphics Menu as shown in Figure 17-1.

Upon accessing the vsRPM Graphics Menu, the RPM axis will be scaled to the X-Span used for the autostored data and a graphic will be generated corresponding to the frequency/order values and channel numbers already programmed into the table. Use the keys to modify the parameters in the table as desired, and press **REDRAW [M]** to obtain a new graphic display. The user can press the key and use the cursor to read out the levels of the different traces and to hide them as desired.

The user can now use the cursor to readout the values of the individual traces and hide traces exactly as was done with the graphic produced in the Real-time vsRPM Graphic mode.

Modification of the Graphic Parameters

In the Real-time vsRPM Graphics mode, the only data which are stored in the graphics memory buffer are those associated with the specific channel numbers and frequencies assigned to the traces previous to performing the test. Once the graphic has been created it cannot be modified, with the exception that traces can be hidden.

With the vsRPM Graphics based on autostored records, the entire set of complete spectra for each channel are available for use in the graphics routine. Thus, the user can now modify most of the graphics parameters, such as the channels and frequencies (or orders) defined for each trace, the horizontal axis endpoints, the slope parameter, and switch between Frequency and Orders in those cases where this is consistent with the data type. Simply press **REDRAW [M]** after modifying the parameter table as desired.

Storage and Recall of Trace Records

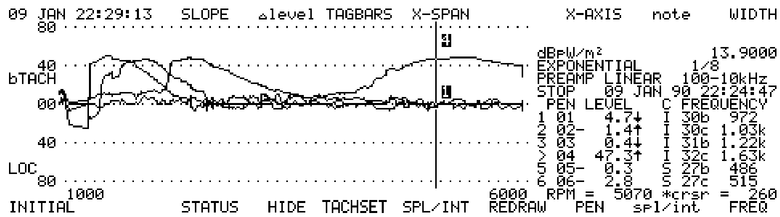
The storage and recall of the displayed vsRPM graphic is performed exactly as explained for the Real-time vsRPM Graphics.

Because of the versatility of this routine for displaying data using different combinations of channel and frequency (or order) for each trace, the user may choose to generate and store a variety of graphic displays as vsRPM Trace records.

Intensity Mode Data

To recall intensity spectra autostored vsRPM, set the 3000+ to the Intensity Mode with vsRPM autostore active and press **RECALL**. Use the **A.Prev [M]** and **A.Next [N]** softkeys to recall the desired record number of that type. In this case, the vsRPM Graphics Menu will look somewhat different, as shown in Figure 17-2.

Figure 17-2 ***vsRPM Graphics Menu (Intensity Data)***



In this mode, the user can select to graph either intensity or sound pressure level spectra. In the graphic parameter table, the column beneath the C will no longer represent the channel number. Instead there will appear either an “S” or an “I” to represent Sound Pressure Level or Intensity, respectively. The softkey **spl/int [O]** will toggle the value for the selected trace between these two. When the selected value is “I”, an upward or downward vertical arrow will appear to the right of the amplitude value in the first column, to represent positive (upward) or negative (downward) intensity corresponding to the alignment of the intensity probe.

The softkey **SPL/INT [L]** will toggle the graphic between a display of sound pressure level versus RPM/Speed and a display of intensity versus RPM/Speed. When the display represents sound pressure level, the units indicated on the right side of the screen, second line down, will be “dB SPL”. When the display represents intensity, the units will be “dBpW/m2”. If some of the traces have been defined to represent intensity and others pressure, then graphic data will appear for either the Intensity or the Sound Pressure display choice. Note that the table will present both intensity and sound pressure level amplitudes corresponding to the cursor position regardless of which graphics mode has been selected.

Post-process Order Tracking

Post-process Order Tracking permits the user to generate curves of order versus RPM/Speed when octave bandwidths or standard FFT filtering are utilized.

The key to this is the availability of the Tach and Speed data for each spectrum. In the real-time mode, values of Tach and Speed are being read with each spectrum. When byTach autostored data are being used, values of Tach and Speed have already been stored along with each spectrum. When performing Post-process Order Tracking, either Tach or Speed will be selected by the user to serve as the reference frequency.

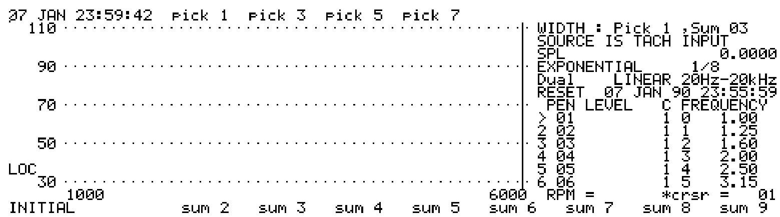
The methodology for Post-process order tracking is to note the value of the reference frequency associated with the spectrum, determine into which filter band that frequency would fall, and take the amplitude of that bandwidth as the amplitude of the first harmonic. A similar procedure is followed for each multiple of the reference frequency to determine the amplitude of the higher orders.

As far as setting up the 3000+ to perform Post-process order tracking, the user proceeds as explained above in the sections describing Real-time vsRPM Graphics and vsRPM Graphics from byTach autostored data. The main difference is that the parameter to be setup in the last column for each trace will be ORDER instead of FREQUENCY. If the vsRPM Graphics Menu is indicating a softkey **FREQ [P]**, press **B–FREQ [L]** to change FREQ to ORDER. When setting this parameter, pressing the horizontal arrow keys will page through a sequence of order numbers instead of frequency.

Peak Hunt Procedure

In cases where the rate of change of the reference signal is high, the time delay inherent in the digital filters may cause the peaks associated with the different orders to fall into one of the frequency bands adjacent to the one where the calculation predicts it ought to be found. Fortunately, in many cases where order analysis is used, most of the dominant components are order related. In such instances, the use of a peak hunting routine can correct this problem. From the vsRPM Graphics Menu, press **WIDTH [H]** which will produce the Peak Hunt Menu, shown in Figure 17-3.

Figure 17-3 ***Peak Hunt and Bandwidth Averaging Menu***



The choices of the Peak Hunt parameter are represented by the row of softkeys above the screen:

pick 1[A] p ick 3[B] p ick 5[C] pick 7[D]

With pick 3 selected, the program examines the amplitude of not only the frequency band which is calculated to represent a particular order, but also those of each adjacent sideband. Of these three bands, the one whose amplitude is the largest is taken to represent that order.

With pick 5 the search includes the two adjacent bands on each side, and with pick 7 the three adjacent sidebands on each side. With pick 1, no sidebands are examined.

Note that when this analysis is being performed from autostored spectral data, the user can generate a variety of vsRPM/Speed graphics using different choices of the peak hunt parameter.

When using octave bandwidths for Post-process order tracking, selecting the SHORT filter algorithm rather than the LONG one will reduce the filter delay. Since in most cases the signal components of concern will be harmonically related, the loss in bandwidth selectivity will not affect the accuracy significantly.

Bandwidth Averaging Procedure

When working with fractional octave filters, the amplitude at the crossover point between adjacent filters is -3 dB relative to the passband. Therefore, when tracking a constant amplitude signal which is changing in frequency, the order vsRPM/Speed curve will dip by 3 dB each time the signal falls between two filter bands. This effect can be reduced by averaging the levels in more than one bandwidth.

The number of bandwidths over which the averaging is to be performed is selected from the Peak Hunt and Bandwidth Averaging Menu shown in Figure 17-3. The choices are represented by the softkeys below the screen.

Another reason for utilizing the Bandwidth Averaging is to increase the effective bandwidth of the analysis for the purpose of comparing the data with that measured using a different bandwidth, such as that measured using a different analysis system.

CHAPTER

18 Statistics and L_n Calculations

Statistics and Ln values (n = 1-99 in integer steps) can be calculated using either 1/1 or 1/3 octave bandwidth. Statistical values are calculated for each frequency band and, in the SLM mode, all the sound level meter parameters. An Ln value from a set of measurements represents the amplitude level which was exceeded "n" percent of the time over the measurement period. For example, suppose that the level in the 250 Hz frequency band has been sampled 1,000 times, and that the value calculated for L_{90} is 85 dB. This means that 90% of the samples (900 samples) had level values above 85 dB. In the Model 3000+, Ln values are determined in integer steps from L_{01} to L_{99} over a user-positioned measurement range of 120 dB. Autoranging must be utilized to obtain this large measurement range, since the dynamic range of the 3000+ is approximately 80 dB. Other statistical values which are calculated for each frequency band are the maximum value, the minimum value, the median value, the mean value and the standard deviation. In the SLM and single channel Standard mode, the resolution of the statistics is 0.5 dB. In the dual channel Standard mode, the resolution of 1.0 dB.

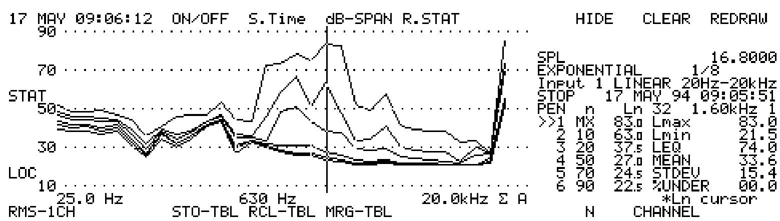
Setup for Statistical Analysis

Statistical analysis can be performed with the 3000+ in either the SLM or the Analyzer Mode. In the Analyzer Mode statistics are calculated in 1/1 or 1/3 octave frequency bandwidths. In the SLM Mode, statistics are calculated for the sound pressure level (both Slow and Fast response) in addition to those for the 1/1 or 1/3 octave bandwidths. Because the weighting of the SLM function is selected independently from the weighting of the Analyzer function, the user can select A or C-weighting for the SLM statistics and linear weighting for the frequency band statistics.

If using the 3000+ in the Analyzer Mode, access the Statistics Menu, shown in Figure 18-1, from the Analyzer Menu by pressing **STAT [K]**.

If using the 3000+ in the SLM Mode, access the Statistics Menu from the SLM Menu by pressing the sequence **DISPLAY [F], STAT [B].**

Figure 18-1 ***Statistics Menu***



Setting the Update Interval

Data is taken from the measurement buffer to the Statistics Table at regular time intervals specified by the user. Before turning on the statistics mode, set this interval by pressing **S.Time [B]**, which will bring to the upper right of the screen the message “UPDATE DELTA = XXXXXX.XXXX”. Use the numeric keypad to type in the desired value, in seconds, and press **EXIT**. The statistics mode must be off when modifying the update interval. If it is on, the message “To edit, turn stats OFF” will appear briefly on the upper right of the screen.

Setting Measurement Range

The measurement can be made over a measurement range of 120 dB. The statistics mode must be off to modify the measurement range. To observe the selected range, or to modify it, press **dB-SPAN [C]** and note that the presently selected range is indicated on the lower right side of the screen in the form “*tbl XX/YYY”, meaning that the statistics table is configured to handle sampled level values ranging from XX up to YYY. Note that XX may have negative values. While this message is displayed, use the horizontal arrow keys to move this 120 dB range to encompass the desired range of input values for the measurement to be made.

Use of Autoranging

The autoranging function is meant to deal with situations where the general sound level increases or decreases significantly, yet slowly, over time such as may occur during 24 hour measurements where the night time levels are substantially lower than the day time levels. During the short time period (typically several seconds) when the autoranging process is taking place, data will not be available for updating of the statistical table, so there will be a loss of information. As a result, the autoranging function may not be able to deal effectively with short term events which initiate an autoranging operation. For example, should the instrument gain be set such that a vehicle passby produces an overload, therefore initiating an autorange operation, data corresponding to the passby would be lost during the overload and the autoranging operation.

The duration of the autoranging operation will depend upon both the averaging time and the highpass filter selection. The fastest response results from the use of the lowest averaging time and the highest value of highpass filter.

When the application is such that significant noise events of short duration are of major interest, and the general background levels are very low, it is best that autoranging not be utilized. The instrument range should be set to handle the events without overload. In such a case, much of the background noise may fall below the lower limit of the measurement, as indicated by significant values of the displayed parameter “%UNDER”.

Turning the Statistics Analysis On and Off

Repeated presses of the softkey **ON/OFF [A]** will toggle the state of the statistics mode between on and off. If the statistics mode is off, pressing this key will result in the message "Statistics mode is ON" on the upper right of the screen. If the statistics mode is on at the time this key is pressed, the message on the upper right of the screen will be "*ARE YOU SURE?*" to remind the user that the data presently in the Statistics Table will be lost if the statistics mode is turned off. To proceed, press **YES [A].** To abort the turn-off procedure and leave the statistics mode on, press **NO [C]**.

Selecting the Ln Values for Calculation and Display

The Statistics Table which is generated during a measurement when the statistics mode is on is capable of producing Ln values between 1 and 99, in integer steps, for each frequency and for the broadband level. Up to six statistics curves, each corresponding to a particular value of n, can be calculated and displayed at one time. The statistics parameter table on the right of the screen is used to assign a value of n to each of the six curves which may be drawn.

Select the trace number (1-6) whose parameter value is to be changed by pressing that numerical value on the keypad. The symbol ">>" will appear to the left of the trace number indicating that it has been addressed and may be modified. To change the value of n corresponding to the desired Ln, press **N [N]**, and press the horizontal arrow keys until the desired value appears. Note that one increment below the value n=1 the word "Max" will appear, indicating that the parameter is set for the maximum value. Similarly, one increment above the n=99 the word Min will appear, indicating that the parameter is set for the minimum value.

If it is desired that less than six traces are to be used, addressing any trace and pressing **CLEAR [G]** will disable that trace, indicated by a space where the parameter value would normally appear. No data is calculated or displayed for a trace which has been cleared. Note that when a previously cleared trace is addressed, the parameter value which had been displayed at the time the trace was cleared will reappear.

Running the Statistics Mode

When the statistics mode has been turned on and the statistics parameter table set as desired, press **RUN/STOP** to begin the analysis. The state of the analyzer, as indicated on the fourth line down on the right side of the screen will change to RUN to indicate that the analysis has begun and that the Statistics Table is receiving input and being updated at the regular time intervals set by the user. However, no curves will be drawn until either the analysis is stopped by pressing **RUN/STOP** or until the softkey **REDRAW [H]** is pressed. When either of these keys is pressed, the Ln data and curves representing them are generated from the data in the Statistics Table at that instant. There is no mode whereby the Ln values and curves are generated and displayed in real-time.

It is not necessary to keep the Statistics Menu on the screen during the analysis. Most users would probably prefer to return to the Standard Analysis Menu in order to observe the spectral display during the analysis period.

Calculation and Display of Data

After the analysis has been running, from the Statistics Menu press either **RUN/STOP** or **REDRAW [H]** to generate and display the curves corresponding to the parameters in the statistics parameter table.

When the statistics are being calculated from the analyzer mode of operation, the complete horizontal scale is used to represent the frequency range of the analysis. The horizontal arrow keys are used to move the cursor across the frequency range. The cursor position is indicated on the right of the screen, fifth line down. The Ln values for each pen corresponding to the cursor position are presented in the table.

When the statistics are being calculated from the SLM mode of operation, the left three-quarters of the horizontal scale represent the frequency range and the remaining portion represents the sound pressure level statistics. As the cursor is moved upwards through the frequency range, after passing through the highest frequency band, continued movement to the right will produce in sequence the statistics associated with the Slow detector of the SLM, those for the Fast detector and the spectral sum. The last parameter is not produced from the sound level meter function, but calculated from the frequency analysis as the sum of the energy of all the frequency bands.

The mean and standard deviation are calculated as follows:

$$\text{mean:}\bar{x} = \frac{1}{n}\sum_{i=1}^{n} x_i$$

$$\text{STDDEV} = \sqrt{\frac{1}{n-1}\sum_{i=1}^{n}(x_i - \bar{x})^2}$$

The instrument cannot, of course, measure levels which are below the lower limit of the selected measurement range. In order to provide an indication that levels are frequently falling below this lower limit, the parameter "%UNDER" is displayed. In cases where the "%UNDER" is high, and the Lmax levels are much lower than the upper limit of the measurement range, the input gain should be increased, thus shifting the measurement range downward to include more lower level values.

Selecting the Display Channel Number

With the Model 3000+ set to dual channel mode (STAND 2), independent Statistics Tables are generated for each channel. Ln statistics and curves can be displayed for only one channel at a time. The number of the channel whose statistics are being displayed (1 or 2) is displayed on the right of the screen, 5th line down, to the right of the frequency corresponding to the cursor position. The default selection is channel 1. Pressing the softkey **CHANNEL [0]** will toggle this between the two channels as indicated by changes in the 5th line on the right of the screen.

Modifying the Parameter Table Values

The parameter value, n, for any trace may be changed in the statistics parameter table by simply addressing the desired trace using the numeric keypad, pressing **N [N]** and using the horizontal arrow keys in the same manner as originally used to set the statistics parameter table values. The entire set of curves will be regenerated and drawn immediately upon changing any parameter value. Even though a redraw is in progress, the user can continue to use the horizontal arrow keys to modify a parameter. Eventually, after a series of redraws, the display will correspond to the final value entered into the table.

Hiding a Trace

For reasons of clarity, the user may wish to display only one or several of the six traces at a time. Any trace may be hidden by addressing it and pressing **HIDE [F]**. When this is done, the curve corresponding to that trace will not appear on the screen. The status of a hidden trace is indicated by an asterisk * to the left of the trace number. Any number of traces may be hidden at one time. To unhide, or to again include a trace in the set of curves being displayed, simply address that trace and once again press **HIDE [F]**. The asterisk will then disappear to denote that the trace is no longer hidden.

Clearing the Statistics Table

Repeated presses of the **RUN/STOP** key will simply start and stop the analysis; there will be no reset of the Statistics Table. Whenever the analyzer is running, the Statistics Table will continue to be updated and will represent the statistical characteristics associated with all measurements since the Statistics mode was turned on, or was last cleared. In order to clear the Statistics Table to begin a new independent measurement sequence, press **R.STAT [D]** which will produce the message “*ARE YOU SURE?*” on the upper right of the screen. To continue and clear the Statistics Table, press

YES [A]. To abort the clearing operation and preserve the active Statistics Table, press **NO [C]**.

Storing the Ln Trace

To store the data associated with the statistics parameter table presently displayed, including hidden traces, can be stored to the active memory file by pressing **STORE**. This will be sufficient data to regenerate the curves presently displayed on the screen. The message displayed on the upper right of the screen, "STORE - Ln Trace N" indicates that this data have been stored as the Nth record of type Ln Trace in the active memory file. It is recommended that a descriptive note be created before storage of the Ln Trace. Although the Ln Trace notes are not displayed during the recall procedure, the record listing in the Files Menu will permit the user to observe the notes attached to each stored Ln trace record.

Storing the Statistics Table

The complete Statistics Table can be stored to memory by pressing **STO-TBL [I]**. Note that storage of the Statistics Table requires a large amount of memory, in excess of 22 KB. It is recommended that a descriptive note be created prior to storing the Statistics Table. Although the Statistics Table notes are not displayed during the recall procedure, the record listing in the Files Menu will permit the user to observe the notes attached to each stored Statistics Table record.

Recalling Ln Traces

To recall an Ln Trace, from the Statistics Menu press **RECALL**. The message on the upper right of the screen, "RECALL - Ln Trace N" indicates the Nth record of type Ln Trace has been recalled from the active memory file. If a different Ln Trace record number is desire, use the horizontal arrow keys to access it.

The cursor can be used to readout the data values corresponding the statistics parameter table and the displayed curves. Hidden traces can also be unhidden. However, attempts to modify the values in the statistics parameter table by pressing **N [N]** will produce the message “Illegal with recall data!”. This is because the complete Statistics Table must be available to calculate statistics for values of n different than those already present in the statistics parameter table.

Recalling a Statistics Table

To recall a Statistics Table, press **RCL-TBL [J]**. The message on the upper right of the screen will prompt the user to enter the record number of the Statistics Table to be recalled using the numeric keypad and press **EXIT**. This will produce the message “* ARE YOU SURE?*” on the upper right of the screen, warning the user that the recalled Statistics Table will overwrite the Statistics Table presently active in the 3000+. To proceed press **YES [A]**. To abort the recall and maintain the present Statistics Table intact, press **NO [C]**.

Once the Statistics Table has been recalled, data and curves will be produced as described above. The user may then modify the statistics parameter table and the displayed curves as desired.

Merging Statistics Tables

A stored Statistics Table can be merged with the active statistics table by pressing **MRG-TBL [K]**. This will produce the message “Enter RECORD number XX”, prompting the user to enter the record number of the stored Statistics Table which is to be merged with the active Statistics Table using the numeric keypad and press **EXIT**. This will be followed by the message “*ARE YOU SURE?” warning that the newly merged Statistics Table will overwrite the presently active Statistics Table. To continue press **YES [A]**. To abort the merge and preserve the present Statistics Table, press **NO [C]**.

An example of the use of the merge function would be where one wishes to know the noise statistics for the morning rush hour each day during the workweek, and also the statistics for the entire weeks morning rush hour periods. The Statistics Table measured each morning will give the daily information. If the Statistics Table is stored daily, then at the conclusion of the week all five of these tables could be merged together to produce a single table representing the week long morning statistical data for use in producing statistics for that entire sample period.

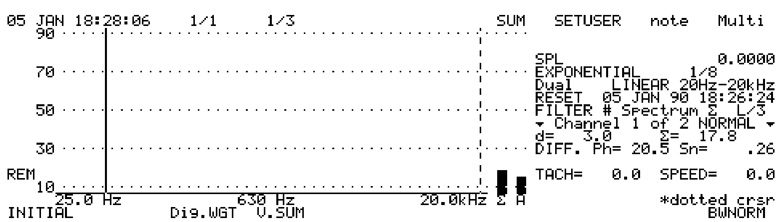
CHAPTER

19 Control of Display Formats, Cross-Channel Normalization and Use of Key Macros

Accessing the Display Menu

Many of the functions discussed in this Chapter are initiated from the Display Menu, shown in Figure 19-1, which is accessed from the Main Menu by pressing **DISPLAY [F]**.

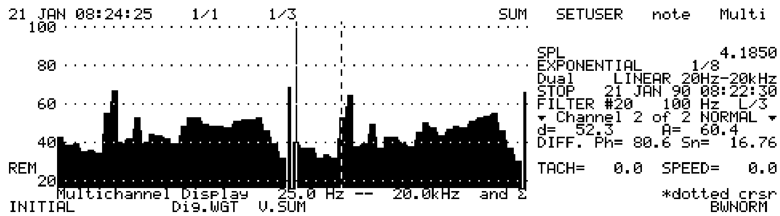
Figure 19- 1 ***Display Menu***



Dual Channel Side-by-Side Display Mode

When the Model 3000+ is configured for dual channel measurements in the Standard Analysis Mode, it is possible to display the spectra for both channels simultaneously in a side-by-side configuration as shown in Figure 19-2. The spectrum for channel 1 is displayed on the left side of the screen and the spectrum for channel 2 is displayed on the right side of the screen.

Figure 19-2 ***Multi Display Format***



This function is enabled from the Display Menu by pressing **Multi [H]**. Repeated presses of this softkey will toggle between the single and the dual channel display formats. The cursor and the data readouts on the right side of the screen will correspond to one of the two displays (channels) as indicated by the message “Channel 1 of 2 NORMAL” or “Channel 2 of 2 NORMAL” on the right side of the screen, sixth line down. Use the hardkeys **CH1** and **CH2** to select which of the two channels are to be accessed by the cursor and readout on the right side of the screen.

Displaying 1/3 Octave Spectra in 1/1 Octave Format

When a spectrum has been measured using 1/3 octave bandwidths, it is possible to sum these in groups of three in order to produce a spectrum having 1/1 octave bandwidths. This is displayed from the Display Menu, Figure 19-1, by pressing **1/1 [A]**. To return to the 1/3 octave bandwidths representation, press **1/3 [B]**.

When 1/1 octave bandwidths have been used for the measurement, only the 1/1 octave format is possible, so neither of the softkeys **1/1 [A]** or **1/3 [B]** will appear in the Display Menu.

Digital Reading of A-Weight and Summation Bands

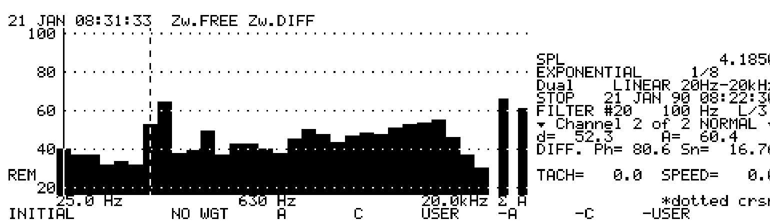
Also located in the Display Menu is the key **SUM [E]** which controls the digital readout of the broadband levels represented by the two vertical bars on the right of the spectrum display. When solid or dotted cursors are active, these represent the A-Weighted and Linear broadband levels calculated from the sum of the energy measured 1 Hz and 20 kHz. These are identified below the bars by the letter “A” and the summation symbol “Σ”, respectively. Repeatedly pressing **SUM [E]** will cause the digital value displayed on the lower right of the screen to toggle between these two, as indicated by the “A” or summation symbol on the line below the channel indication.

When the "*both" cursor mode is active, these bands represent the A-Weighted and Linear broadband levels calculated from the sum of the energy between the two cursors, rather than between the highpass and lowpass filters.

Digital Display Weighting

Digital Display Weighting is controlled from the Digital weight Menu, shown in Figure 19-3, which is accessed from either the Standard or Intensity Analysis Menu by pressing **DISPLAY [F]** then **Dig.WGT [I]**

Figure 19- 3 ***Digital weighting Menu***



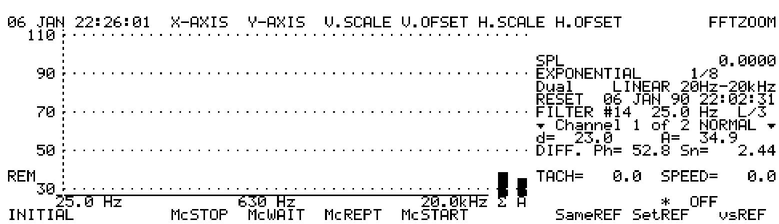
This Menu permits the user to select an A, C, User, –A, –C or –User Weighting Function to be applied to measured spectral data before it is displayed. This will be in addition to analog weighting which may have been applied at the input. Digital display weighting is described in detail in Chapter 10. The key **SETUSER [F],** from the Display Menu, is used to create the User Weightings. Display weighting is not possible when using Cross Analysis.

Display of Spectra Relative to a Reference Spectrum

In some application it is desirable to compare two spectra, or to display spectra relative to some user-defined spectrum. Although the 3000+ can only display a single spectrum on the LCD screen at one time, it is possible to display relative to a reference spectrum. This is quite useful when using logarithmic amplitude scales (dB) because this format produces a spectrum which represents the difference between the selected and the reference spectrum.

The spectrum which is to be defined as the reference spectrum must first be displayed. In most cases it will be a spectrum already measured and stored, so simply recall it. If it has not already been stored, it is best to store it at this time since one will probably need to recall it at a later time to show just what the reference spectrum looked like. Access the Shift Menu, shown in Figure 19-4, by pressing **SHIFT** and then press **SetREF [O]**. The message “vsREF” on the left of the screen along the vertical axis indicates that the amplitudes correspond to a spectrum being displayed relative to the reference spectrum. Because we are now displaying the same spectrum which was selected as the reference spectrum, (a spectrum relative to itself) all points will have zero amplitude, resulting in a horizontal line.

Figure 19-4 ***Shift Menu***



Any spectra now displayed, whether just measured or recalled from memory (remember to press **KEEP [H]** when exiting from the Recall Menu) will be displayed relative to the reference spectrum. If the message “Reference may not match” appears on the upper right of the screen, this indicates that the displayed and reference spectra have different bandwidths and a display of this spectrum versus the present reference spectrum is not appropriate.

Dual Channel Measurements

When the Model 3000+ is configured for dual channel analysis, the data block to be used as a reference will represent a dual channel measurement and the spectra for channels 1 and 2 will in most cases be different. However, the spectrum for only one of these two channels can be displayed at the time the reference spectrum is defined.

By pressing **sameREF [N**], the displayed spectrum will be defined as the reference spectrum for both channels. This makes it very easy to display the difference between two spectra measured simultaneously in the dual channel mode.

By pressing **SetREF [O]**, the spectrum corresponding to channel 1 will be defined as the reference spectrum for channel 1 and the spectrum corresponding to channel 2 will be defined as the reference spectrum for channel 2.

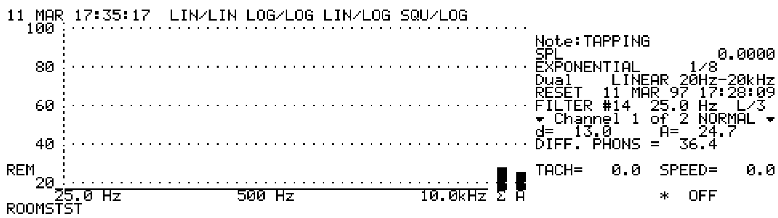
Returning to Normal Display Format

All spectra are measured and stored in their absolute format; the versus reference display mode is purely a display function. Return to the normal spectral display format from the Shift Menu by pressing **vsREF [P]** a second time. Repeated presses of **vsREF [P]** will toggle the vsREF display mode on and off.

Control of Vertical Display

There are four different vertical display formats possible with the Model 3000+. The default format active upon turning on the instrument (unless the boot setup has been modified) is log/log, meaning that the numbers along the vertical scale (gradations) and the cursor readout are in logarithmic (dB) units, and the scaling format of the screen is logarithmic as well. This is the format used most often for acoustic measurements. To modify the vertical scaling, access the Y-Axis Menu, shown in Figure 19-4, by pressing the key sequence **SHIFT**, **Y-AXIS [B]**.

Figure 19-5 ***Y-Axis Menu***



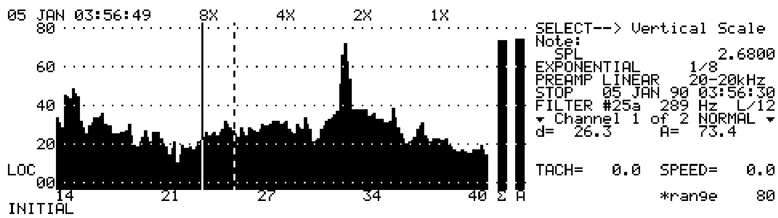
To select a new scaling format, press one of the following:

Softkeys	Softkey Functions
LIN/LIN [A]	Linear gradations and cursor readouts on a linear scale
LOG/LOG [B]	Log gradations and cursor readouts on a log scale
LIN/LOG [C]	Linear gradations and cursor readouts on a log scale
SQU/LOG [D]	Squared linear gradations and cursor readouts on a log scale; useful for display of power spectral density

Control of Display Range

Normally the screen will display an amplitude range of 80 dB using the logarithmic format. In linear format, this will be seen as 100% to 0% from top to bottom, expressed in units of percent of full scale. This display range can be reduced from the Shift Menu (accessed from the Analysis Menu by pressing **SHIFT**) by pressing **V.SCALE [C]**, which will bring to the screen the Vertical Scale Menu shown in Figure 19-6.

Figure 19- 6 ***Vertical Scaling Menu***



Press one of the following keys to obtain the full screen range as indicated below:

Table 19-1 **Vertical Scales**

Key	*Log Full Display Range	*Linear Full Display Range	† Normalized Display Range
8* [A]	10	100% to 87.5%	0.12 to -0.12
4* [B]	20	100% to 75.0%	0.25 to -0.25
2* [C]	40	100% to 50%	0.50 to -0.50
1* [D]	80	100% to 0%	1.00 to -1.00

*** Display of frequency domain functions will be either log or linear.**

† The Normalized display range is used for display of time-domain functions such as autocorrelation, cross correlation, coherence, coherent output power, time, weighted time and averaged time.

After this is done, note that the value of the full scale on the display remains the same, but that the numbers below are reduced as a result of the decrease in the display range.

When the displayed amplitude range is made less than the dynamic range of the measurement, the resolution seen on the screen is increased but the entire valid range of the measured data can no longer be seen at one time. For example, when the displayed amplitude range is decreased from 80 dB to 20 dB, only the upper 20 dB of the data will be visible on the screen.

This conflict is resolved by offsetting the position of the display window relative to the full scale of the measurement. The offset is dynamically adjustable by the user, providing a moveable display window of high resolution which can be shifted up or down through the range of the measured data. This is done from the Shift Menu by pressing **V.OFSET [D],** which will produce the message “*V.Offset xx” indicating that the horizontal arrow keys have been assigned to control the position of the vertical display window. Use the horizontal arrow keys to shift the window in steps of 10 dB. As the window is shifted, the value of xx in the message will change to indicate the position of the window, in multiples of 10 dB, relative to the full scale of the original display. For example, the message “*V.Offset-30” indicates that the displayed full scale value is 30 dB below the full scale of the actual measured data. To reset the window to the zero position, simply press **V.OFSET [D]** a second time (or move it back with the horizontal arrow keys, and then assign the horizontal arrow keys to another role, such as controlling the cursor.

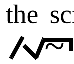
Note that the vertical display range and offset are set uniquely for each display function. This means, for example, that in the Cross Analysis Mode different ranges and offsets can be set for the display of Autospectum, Transfer Function, and Coherence data.

The use of very small display ranges, such as 10 dB, can lead to confusion if the user forgets that it has been selected as such. For example, if a new measurement is initiated it may appear that there is a malfunction in the measurement system when no data appears on the screen. It may simply be that the highest component in the spectrum is more than 10 dB below the full scale and therefore not visible without moving the viewing window.

Bandwidth Compensation (Power Spectral Density)

When displaying spectra, whether in octave or FFT bandwidths, the amplitude of each filter band represents the RMS values of the energy measured contained within that band. This is not typically a problem when using octave bandwidths because their bandwidths and center frequencies are established by international standards.

However, when performing FFT analysis the bandwidth of each filter depends upon the following factors: number of lines, baseband full scale frequency and zoom factor. Since a variety of each of these are available with most analyzers, it could be very difficult to compare measurements made with different combinations of these parameters. One way of dealing with this is to compensate for the bandwidth by dividing the energy within each band by the bandwidth of the filter and use this as the amplitude value for the display. The units would then be in the form of energy/Hz. This form of data presentation is often referred to as power spectral density. A very common application is in the measurement of random vibration, where the desired amplitude units are g^2/Hz.

To display spectra in the bandwidth compensated format, access the Display Menu (from the Analysis Menu press **DISPLAY [F]**) and press **BWNORM [P]**. The fact that bandwidth compensation is active is indicated on the right of the screen, first line, after the units name, by the symbol . Repeated presses of **BWNORM [P]** will turn bandwidth compensation on and off.

When using the LIN/LIN or LIN/LOG vertical display format with the bandwidth compensation active, the units will be ($\text{linear unit} / \sqrt{\text{Hz}}$). When using the SQU/LOG vertical display format with the bandwidth compensation active, the units will be [(linear unit)^2/Hz]. Thus, with the system calibrated to units of "g", the LIN/LIN and LIN/LOG vertical displays will $g / \sqrt{\text{Hz}}$ while the SQU/LOG vertical display will provide g^2 / Hz.

Control of Horizontal Display

Selection of Logarithmic/Linear Format

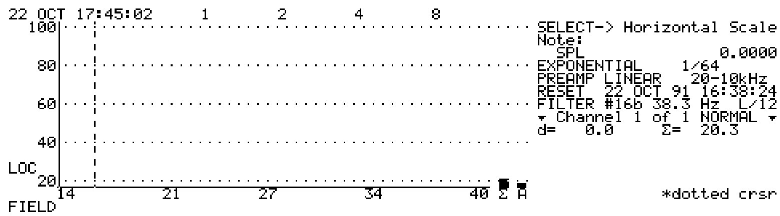
When displaying spectra measured using fractional octave filters, the horizontal axis representing frequency will be logarithmic. When displaying time-domain functions, the horizontal axis representing time will be linear. Neither of these can be changed.

When displaying spectra measured using FFT filtering, the default setting of the horizontal axis representing frequency will be linear. However, from the Shift Menu shown in Figure 19-2, repeated presses of **X-AXIS [A]** will toggle the format of this scale between logarithmic and linear.

Control of Display Range

In the default condition, the measured data block will be displayed such that the frequency (or time) range used for the measurement extends completely across the screen from left to right. This represents a horizontal scale factor of unity. The user can change this value. When a scale factor of eight is selected, for example, the total horizontal width of the data block is multiplied by eight, so that only one-eighth of the width of the block can be seen at one time, but the screen resolution will be eight times greater than with unity scale factor. This is done from the Shift Menu (Accessed from the Analysis Menu by pressing **SHIFT**) by pressing **H.SCALE [E]** which will bring to the screen the Horizontal Scale Menu shown in Figure 19-7.

Figure 19- 7 ***Horizontal Scaling Menu***



Select the Horizontal Scale Factor by pressing the corresponding softkey.

Only a portion of the total data block will be visible at one time when the horizontal scale factor is greater than 1. In order to be able to pan the displayed portion horizontally, from the Shift Menu, press **H.OFSET [F]** which will produce the message "*H. Offset xx" on the lower right of the screen indicating that the horizontal offset of the displayed portion of the horizontal axis is now under control of the horizontal arrow keys.

Normalization of Amplitude and Phase Between Channel 1 and Channel 2

When using the Cross Analysis Mode, channel 1 is the reference channel. Each cross channel measurement such as cross spectrum, transfer function, and coherence are made between channels 1 and 2. In order to minimize the effect of amplitude and phase mismatch between the channels, a normalization procedure can be utilized to correct for such mismatches.

When the Model 3000+ is used in the Intensity Analysis Mode, cross spectrum measurements are used for the determination of intensity, so normalization to correct for mismatch between channels may also be desirable to obtain the best possible accuracy for intensity measurements.

The procedure is to input the same broadband noise into both input channels and measure the transfer function between them. Any variation from a unity value of magnitude and a zero value of phase represents the effect of mismatch. Once measured, this transfer function can then be used to build a correction function which will normalize the cross channel data taken in subsequent measurements to correct for these errors.

If only the amplitude and phase of the analyzer itself are to be normalized, the broadband noise source is applied directly to the input connectors. When using Larson Davis side-vented 1/2" or 1/4" microphones such as supplied with the Model 2260 Sound Intensity Probe, the user can utilize the Model CAL291 Residual Intensity Calibrator in conjunction with the noise source to amplitude and phase match the complete measurement system including the microphones and preamplifiers. The CAL291 applies the same amplitude acoustic signal, with zero phase difference between them, to both microphones.

Connection of the Noise Generator

The internal noise generator of the Model 3000+ will be used for the normalization procedure. If normalizing just the instrument, use a BNC “T” connector and several cables to direct the output of the generator to both inputs using an ADP012 BNC-to-5-pin Switchcraft adapter.

If normalizing through measurement microphones, connect the output of the generator to the input of the CAL291 and press the microphones firmly into the microphone openings.

Normalization in Cross Mode, Using FFT Filtering

Normalization using FFT filtering must be done using the same values of highpass and lowpass filters and choice of time weighting window as will be used for subsequent measurements. If any of these are changed after the normalization has been done, the normalization must be redone. Therefore, setup the analyzer accordingly.

Selection of 100 Line Resolution

FFT normalization curves are always measured using 100 line resolution. If during a subsequent analysis a larger number of lines is selected, the correction function will be extrapolated from the one created using 100 lines.

Selection of Baseband Full Scale Frequency

The system memory permits the measurement and storage of a unique normalization function for each permissible value of baseband full scale frequency. Following the normalization procedure, with normalization active, during subsequent measurements the 3000+ will utilize the stored normalization function which corresponds to the value of baseband full scale frequency selected for the analysis. Thus, it is best to measure and store a normalization curve for each permitted baseband full scale frequency to be certain that the results will be correct for any value of full scale frequency which may be selected. Otherwise, there may be normalization curves in memory which do not correspond to the measurement being made.

Noise Generator Setup

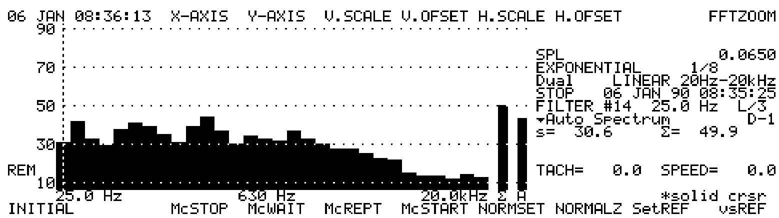
Set the noise generator to white noise, and turn it on. White noise is used because its equal energy per constant bandwidth spectrum produces a nominally flat spectrum shape with FFT filtering.

Measurement

Set the analyzer to Count Single Averaging and a sufficiently large number of spectra that a very stable, accurate measurement of transfer function is made. Observe the display of the transfer function during the measurement to verify that it has converged to a stable value. If there is any doubt, average over a larger number of spectra.

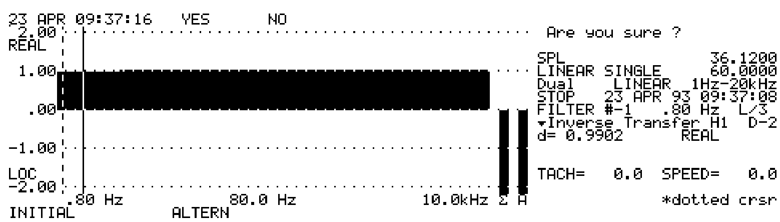
Access the Shift Menu, shown in Figure 19-8, by pressing **SHIFT**, after selecting Cross channel analysis.

Figure 19-8 ***Shift Menu***



Quickly press **NORMSET [M]** and the Normalization Menu will appear as shown in Figure 19-9**.**

Figure 19-9 ***Normalization Menu***



The real part of the inverse transfer function will be displayed on the screen at this time. To display the imaginary part, press **ALTERN [I]**. The message "are you sure?" on the upper right of the screen will prompt the user to verify that he wishes to replace whatever correction function may have previously been stored for that baseband frequency value with the newly calculated correction function.

To store the correction function press **YES [A]**. The display will return to the Menu which had been active prior to accessing the Shift Menu, and the Normalization Mode of the 3000+ will be active, as shown by the message **NORM** on the left of the screen.

To abort the storage procedure, press **NO [B]**. The screen will revert to the Menu active prior to accessing the Shift

Menu, but the Normalization Mode of the 3000+ will not be active.

If it happens that the 3000+ already had the Normalization Mode active at the time of attempting to set the normalization correction function, the message “Normalization must be OFF” will appear on the upper right of the screen. It will be necessary to turn off the normalization mode and repeat the transfer function measurement before completing the normalization procedure.

Toggling Normalization ON and OFF

From the Shift Menu shown above, the normalization mode is turned on and off by repeated presses of the softkey **NORMALZ [N]**. The corresponding messages on the screen will be “Normalization is ON” or “Normalization is OFF”.

Thus, whenever the Normalization is OFF the measurement is a simple cross channel measurement, and when it is ON, the stored correction function corresponding to the selected baseband full scale frequency will be used to correct the measured results.

Normalization in the Cross Mode, using Octave Bandwidths

The normalization procedure used with Octave Bandwidths is nearly identical to that used with FFT bandwidth. The main differences are as follows:

Step 1 Pink noise is recommended rather than white noise.

Step 2 Use Linear Averaging for the measurement, with an averaging time sufficiently long that a stable value of transfer function is obtained.

Correction functions for 1/1 and 1/3 octave bandwidths are stored separately. This means that once the proper correction function has been stored for each bandwidth, the user can switch between bandwidths without re-calculating the correction function. However, the user must be certain that the correction function being used has been generated using the same highpass and lowpass filters as the analyzer setup.

The octave bandwidth normalization functions created and stored while in the Cross Analysis Mode are used only when normalization is activated from the Cross Mode.

Key Macros

The operation of the Model 3000+ can be simplified by the use of user-defined key macros. A key macro permits the user to define up to fifty sequential keypresses and, upon executing the macro, have the operation of the analyzer proceed as if each of these keypresses were being performed manually in the order programmed. A simple example would be where the user is doing dual channel structural dynamics measurements, and wishes to store the excitation and response autospectra, the transfer functions H1 and H2 in both rectangular and polar coordinates, and the coherence. Under manual operation, each variable must be recalled in the desired format, then stored. By stringing together the keypresses representing the selection of display parameter, the coordinate system (where applicable), and the store command within a macro, simple execution of the macro would perform the complete sequence, and it would be available for user execution whenever required. Up to eight different macros can be available at one time. Macros can also be stored to memory and recalled for keyboard use.

Creating Macros

To create a macro, press the key sequence **SPACE**, **CREATE [A]** which will display the Macro Menu. As directed by the message on the upper right of the screen, press one of the keys **[I] – [P}** to select which key is to be used later to execute the macro, type in a label to identify it and press **EXIT**.

Following this sequence, it will record sequential keypresses performed by the user until either the hardkey "**-**" is pressed again or fifty keypresses have been performed.

While making key presses during the creation of a macro, if the **SPACE** key is pressed previous to pressing a key, that key press will be included in the macro but it will not be executed during the programming of the macro. This is very useful when utilizing the keys **STORE** and **RUN/STOP** during the definition of a macro.

McSTOP, McWAIT, McREPT and McSTART Softkeys

The are three softkeys in the Shift Menu which can be used in the creation of macros. **McSTOP [I]** will cause the macro processor to pause during its execution until the analyzer is in the STOP state, at which time the execution will continue. This is useful for defining macros which will initiate a measurement and also perform operations after the measurement has been completed. This implies the use of an averaging method where the measurement sequence stops without the requirement of manual intervention (e.g. linear single or count single) or an autostore measurement.

The softkey **McWAIT [J]** is used to provide a wait state during the execution of a macro. Upon pressing this key, a menu will offer the user a choice of wait periods in a binary sequence beginning with 0.25 second.

The softkey **McREPT [K]** is used to generate a repeat of the macro. When this is used, the macro will continue to be repeated until the analyzer is stopped manually by pressing the hardkey **SPACE**.

The softkey **McSTART [L]** sets the analyzer up with a time and date for the macro to start.

Resetting Macros

From the Reset Menu, pressing **R.MACRO [F]** will clear all the macros presently active.

Executing Macros

To execute a macro, simply press **SPACE** to display the Macro Menu, followed by the softkey corresponding to the desired macro. The message “MACRO: Executing N”, where N is the letter (I-P) of the softkey corresponding to the macro being executed, will appear on the upper right of the screen. When executing a macro the menu displayed on the screen must be the same one which was displayed when the macro was created. The message “MACRO MENU MISMATCH” will appear on the upper right of the screen when the wrong menu is being displayed when attempting to execute a macro.

A macro can also be executed via the opto-isolated ports, such as from one of the keys on the intensity probe. When programming the role of Key A or Key B from the I/O Menu, use the key sequence **-**, **SPACE**, [I-P].

For example, to program the Key A to execute the macro K, use the sequence **KEY A [D]**, **-**, **SPACE**, **[K]**.

Delayed Macro Execution

In order to schedule the execution of a macro at a future time, press **SHIFT, McSTART [L]**, which will bring to the upper right of the screen the message “Macro M on DD at HH:MM:SS”. “M” is the number of the macro to be executed, “DD” is the day for the execution and “HH:MM:SS” is the time for the execution. The letter “M” will be flashing to indicate that the cursor is centered on that letter waiting for an entry.

Storing Macros

To store a set of up to eight macros which are active in the analyzer, press the key sequence **SPACE**, **STORE**.

The message “Macros Data N” on the upper right of the screen will indicate that these macros are stored in the Nth record of type Macros Data. In order to differentiate between the stored Macros Data records, it is recommended that the note field be used to tag each with a descriptive word or phrase.

Recalling Macros

Records of type Macros Data are recalled from the Files Menu. The records are listed on the right side of the screen. Highlight the desired Macros Data record, and press **KEEP [H]** to perform the recall.

CHAPTER

20

Sound Intensity Measurements

Selection of the sound intensity operating mode permits the Model 3000+ (equipped with OPT 80) and used with the Larson Davis Model 2260 Sound Intensity probe, to determine the flow of acoustic energy between the two microphones in a direction parallel to the axis between them. During the measurement, sound pressure data is being sampled at each microphone in a synchronized manner. The sound intensity is then calculated in software based upon a knowledge of the spacing between the microphones and the temperature and static pressure of the medium.

The intensity display depends upon the filters selected for the analysis; 1/1 and 1/3 octave, or 100, 200, 400, or 800 line FFT. When exponential weighting using a short time-constant has been selected, the probe can be moved and rotated to probe around a sound source, permitting the user to observe in real time the changes in the intensity spectrum. This can be useful for rapidly identifying the physical location of dominant sound radiators.

Most often the parameter of interest is the acoustic power flowing across a selected surface element, determined by multiplying the surface area by the average intensity measured in the direction normal to the surface. In a typical project, one measures the acoustic power associated with many separate area elements which together form an envelope in space totally enclosing the device under test. Once the intensity at each point has been measured and stored, along with the area value, then power values can be calculated not only for each area element, but for groups of elements which together make up larger sections of the overall surface. To offer maximum flexibility in these calculations, the individual elements are denoted by AREA names. Groupings of these AREAs, which together make up larger surface areas, are denoted by PART names. Finally, the totality of PARTs are denoted by a JOB name.

When ByTime or ByTach autostore are used with the 3000+ in the Intensity Analysis Mode, both Intensity and SPL spectra will be stored simultaneously. Upon recall, either of these data types may be displayed in the vsTime or vsRPM format as explained in Chapters 15 and 16.

Sound Intensity Standards

There are two types of applicable standards for sound intensity; instrument standards which present minimum performance requirements for the instruments to be used to perform the measurement and application standards which provides a methodology to be followed in order to obtain accurate results. In North America most users will follow standards approved by the American National Standards Institute (ANSI) while in other countries most users will follow standards approved by the International Electrotechnical Commission (IEC) and the International Standards Organization (ISO).

All of these standards are intended to provide sound intensity measurements in one-third octave bands or octave bands. There are no standards which define the measurement of sound intensity in narrow band (FFT) frequency format.

Instrument Standards

ANSI S1.9-1996 Instruments for the Measurement of Sound Intensity

IEC 1043:1993 Instruments for the measurement of sound intensity -Instruments which measure intensity with pairs of pressure sensing microphones.

Both of these standards address the accuracy requirements of the sound intensity measurement system in terms of the pressure-residual intensity index. They also define the requirements for the following devices:

- Residual intensity testing device
- Sound intensity calibrator

When calibrated as recommended in this manual, the combination of the Model 3000+ analyzer and the Model 2260 Sound Intensity probe meets or exceeds the specifications of both these standards for a sound intensity measurement system.

The Larson Davis Model CAL291 Residual Intensity Calibrator meets or exceeds the specifications as a residual intensity testing device as defined in both these standards. To drive the Model CAL291 the Model 3000+ must be equipped with either the OPT 10 Noise Generator or the OPT 11 Signal Generator.

Application Standards

ANSI S12.12-1992 Engineering Method for the Determination of Sound Power Levels of Noise Sources using Sound Intensity. This standard addresses both discrete point and scanning measurement techniques.

ISO 9614-1 Acoustics-Determination of sound power levels of noise sources using sound intensity. Part 1: Measurement at discrete points

ISO 9614-2 Acoustics-Determination of sound power levels of noise sources using sound intensity. Part 2: Measurement by scanning

Setup and Calibration of the Measurement System

In the following we describe the calibration of a measurement system consisting of a Model 3000+ analyzer and a Model 2260 Sound Intensity Probe.

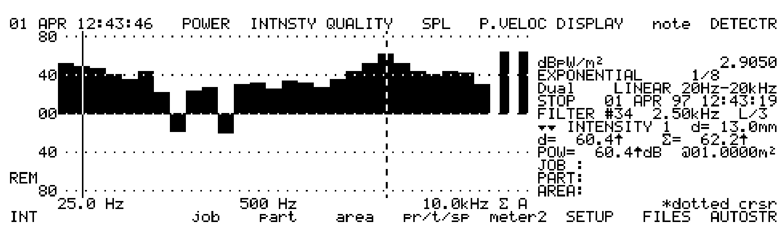
Sound Pressure Level Calibration

Set the Model 3000+ to the dual channel Standard Mode of analysis. Set both channels to the frequency range 1 Hz-10 kHz. Select SPL units for both channels. Use a Type 0, 1 or 1L calibrator to calibrate each of the input channels as described in Chapter 9, Selection of Units and Calibration.

Setup for 1/3 octave Intensity Measurement

From the Main Menu, set the Model 3000+ to the Intensity Mode by pressing **SYSTEM**, **INTENSY [E]** and press **EXIT** to display the Intensity Analysis Menu, shown in Figure 20-1.

Figure 20-1 ***Intensity Analysis Menu***



Select the 1 Hz - 10 kHz frequency range by pressing **SYSTEM**, **INPUT [K]**, **1-10K [M]**, **EXIT**. Select the long 1/3 octave filters by pressing **SYSTEM**, **FILTER [G]**, **1/3oct [B]**, **long [F]**, **EXIT**. The 1/1 octave filters could be selected, but data measured using 1/3 octave filters can also be displayed in 1⁄1 octave format.

Pressure, Temperature and Spacer Length Input

Determine the atmospheric pressure in millibars and the temperature in °C at the location where the measurement is to be made. Select the microphone spacer to be used based on data provided with the intensity probe and note the length. Pressing **pr/ t/ sp [l]** will open the appropriate data entry field on the upper right of the screen. Enter these parameters using the numeric keypad and press **EXIT**.

Amplitude and Phase Normalization; 1/1, 1/3 Octave Measurements

For sound intensity measurements, it is essential to have the best possible amplitude and phase match between channels in order to meet the pressure-residual-intensity specifications of the standard. It is recommended that the Larson Davis Model CAL291 Residual Intensity Calibrator be used for this procedure. In order to use the CAL291, the Model 3000+ must be equipped with either the 2800-OPT 10 Noise Generator or the 2800-OPT 11 Signal Generator.

Loosen the connection of the adjustable arm of the intensity probe, remove the spacer, lay the probe on the surface supporting the calibrator with the handle positioned above the calibrator and press the microphones firmly into the two microphone openings. Set the detector of the 3000+ to exponential averaging with a 1 second averaging time.

If using the CAL291, set the Noise Generator or the Signal Generator to provide a pink noise signal. Press **RUN/STOP** to begin an intensity measurement. Use the vertical arrow keys to select as low a value of full scale amplitude as possible without overloading the input. Press **RUN/STOP** to stop the measurement.

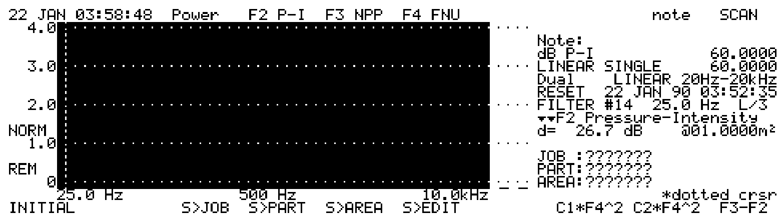
Change the detector to a one minute linear single average by pressing **DETECTR [H]**, **LIN.S [A]**, **AV.TIME [H]**, **EXIT**, **EXIT**. Perform a one minute measurement by pressing **RUN/STOP** key and waiting until the instrument state changes from RUN to STOP.

The data on the screen will be the residual intensity of the measurement system without normalization. Internally, the Model 3000+ is measuring the transfer function between the signals at the two microphone inputs. Because a residual intensity testing device is being used as the acoustic source for both microphones, we know that they are exposed to equal amplitude signals with zero phase difference between them (within the limits of the standard for the residual intensity calibrator). Thus, any variation of the magnitude of the transfer function from unity (0 dB) and of the phase from zero is the result of errors in the measurement system, which includes the microphones, intensity probe electronics and the analyzer. Having measured the transfer function, a correction function can be generated which can be used to improve the precision of measured cross channel data such as sound intensity. We refer to this procedure as normalization.

To perform the amplitude and phase normalization, press **SHIFT**, **NORMSET [M]**, **YES [A]**. This creates the correction function and activates its use in subsequent measurements as indicated by the message “NORM” on the left of the screen. The use of this correction function may be toggled on and off by subsequent presses of the key sequence **SHIFT**, **NORMALZ [N]**. The correction function will remain as last measured until changed by a new normalization procedure.

To verify that the system has been properly normalized, perform a second one minute measurement using the residual intensity testing device with pink noise as the source. To uniquely define this measurement as one associated with the verification of the normalization procedure, assign it the JOB name “NORM” by pressing **job [I]**, using the keypad to enter the name “NORM” into the field on the upper right of the display and pressing **EXIT**. Store the measurement as JOB:NORM by pressing **STORE**. Of course some other name may be used for this purpose, as long as the user is careful not to use a similar JOB name during subsequent measurements. Having completed this naming/storing sequence, the pressure-residual-intensity index associated with this measurement can be displayed by pressing the key sequence **POWER [A]**, **S>JOB [I]**, **F2 P-I [B]**. The results should resemble those in Figure 20-2, measured with the instrumentation system set for a 25 mm spacer distance.

Figure 20- 2 ***Pressure-residual-intensity display***



The Class 1 minimum pressure-residual intensity requirements for a sound intensity measurement system specified by ANSI S1.9-1996 and IEC 1043:1993 are as follows:

Frequency, Hz	ANSI S1.9-1996	IEC 1043:1993
50	13.0	12.0
63	13.0	13.0
80	13.0	14.0
100	13.0	15.0
125	13.0	16.0
160	13.7	17.0
200	14.7	18.0
250	15.7	19.0
315	15.7	19.0
400	15.7	19.0
500	15.7	19.0
630	15.7	19.0
800	15.7	19.0
1,000	15.7	19.0
1,250	15.7	19.0
1,600	15.7	19.0
2,000	15.7	19.0

Frequency, Hz	ANSI S1.9-1996	IEC 1043:1993
2,500	15.7	19.0
3,150	15.7	19.0
4,000	15.7	19.0
5,000	15.7	19.0
6,300	14.9	19.0
8,000	13.7	-
10,000	12.9	-

If the system is working properly and calibrated as specified, the pressure-residual-intensity index should be significantly better than the minimum values specified by the standards.

Press **EXIT** to return to the Intensity Menu.

Sound Intensity using Narrow Band (FFT) Analysis

Although there are no standards governing the measurement of sound intensity using narrow band (FFT) analysis, there are applications where such a measurement may be of great values, particularly when it is important to know the frequency content of the sound intensity or sound power with greater frequency resolution than provided by 1/3octave bands. With the Model 3000+, sound intensity measurements can be performed using FFT analysis. Setup the 3000+ as described above for 1/3 octaves, but instead of 1/3 octave filters select the number of lines desired for the FFT analysis (100, 200, 400 or 800), Hanning weighting and the baseband full scale frequency desired for the analysis.

When using FFT analysis for sound intensity, the normalization procedure must be performed from the Cross Mode as described in Chapter 19. When the normalization has been completed, return to the Intensity Mode to perform sound intensity measurements. Note that the normalization procedure must be performed for the same set of analysis parameters (frequency range, time weighting window and baseband full scale frequency) to be used for the subsequent measurements. To avoid problems, it is best to repeat the normalization prior to utilizing the FFT analysis mode for sound intensity analysis.

Definition of Surface Area (m^2) for the Power Calculation

The intensity measurement produces a spectrum representing values of acoustic power flowing through a unit surface area perpendicular to the direction of alignment of the intensity probe. The linear units of intensity are Watt/square meter. The intensity level, in logarithmic units, is expressed as dB relative to 1 pW/m^2 (1.0×10^{-12} W/m^2).

In many measurement projects, we typically take a single measured intensity spectrum as being the average value over some representative surface area, and then calculate the actual power flowing across that surface by multiplying the measured intensity by the surface area. The units of acoustic power are thus Watts. The units of the acoustic power level are dB relative to 1 pW (1.0×10^{-12} W).

When an intensity spectrum is displayed, both the intensity and the acoustic power levels corresponding to the position of the active cursor are displayed on the right side of the display. The intensity level is shown as normal for the cursor readout, with an “s” or a “d” to denote the solid or dotted cursor. The acoustic power level is displayed below the intensity level in the format POW = XX @ xxxx m^2; XX is the acoustic power level based upon the value used for the Surface Area, xxxx. The value of surface area is entered from the Setup Menu by pressing **meter2 [M]**, which will bring to the upper right of the screen a field indicating the present value of Surface Area. If acceptable, press **EXIT**; otherwise type in the desired value using the keypad before pressing **EXIT**.

In the following section it will be shown how each individual measurement can have associated with it an AREA name as well as a value of surface area. Upon entering the AREA name there will be a prompt to enter a value of surface area as well, in which case it is not necessary to use the **meter2 [M]** key to input a value of surface area.

Job, Part, Area Labels

As explained in the preceding paragraph, one generally associates a value of surface area with the measurement of an acoustic intensity spectrum in order that the acoustic power flowing through that surface may be calculated and displayed.

Furthermore, as described in the introduction to this chapter, it is especially convenient to build upon these measurements and areas a structure which permits the summation of the acoustic power over logical groups of areas which are meaningful in terms of the test object itself. In the 3000+, we do this by permitting the user to define an area label for each measurement in addition to a numerical value of surface area. We call this the AREA label. We may then decide that a designated number of AREAs are to make up a larger grouping which we label as a PART. Finally, we may decide that a designated number of these PARTs are to make up an even larger grouping which we label as a JOB.

It is quite common in acoustic power measurements to imagine a rectangular envelope in space enclosing a test object which is placed on the ground. We might choose to call the entire surface by the JOB label SOURCE. The well-defined surfaces making up the totality of this envelope could be assigned PART labels TOP, FRONT, REAR, LEFT and RIGHT. If we choose to subdivide each of these PARTS into four separate areas, we could label them as UP LEFT, UP RIGHT, LOW LEFT and LOW RIGHT.

The complete test would involve the measurement of four spectra for each of the five surfaces making up the total envelope, a total of 20 measurements. As each measurement is made, the proper value of Surface Area is entered. Then, the position of that measurement in the hierarchy of labels is defined by assigning AREA, PART and JOB labels to each. Following our example, the JOB:PART:AREA labels for the measurements would be as follows:

SOURCE:TOP:UP LEFT

SOURCE:TOP:UP RIGHT

SOURCE:TOP:LOW LEFT

SOURCE:TOP:LOW RIGHT

SOURCE:FRONT:UP LEFT

SOURCE:FRONT:UP RIGHT

SOURCE:FRONT:LOW LEFT

SOURCE:FRONT:LOW RIGHT

SOURCE:REAR:UP LEFT

SOURCE:REAR:UP RIGHT

SOURCE:REAR:LOW LEFT

SOURCE:REAR:LOW RIGHT

SOURCE:LEFT:UP LEFT

SOURCE:LEFT:UP RIGHT

SOURCE:LEFT:LOW LEFT

SOURCE:LEFT:LOW RIGHT

SOURCE:RIGHT:UP LEFT

SOURCE:RIGHT:UP RIGHT

SOURCE:RIGHT:LOW LEFT

SOURCE:RIGHT:LOW RIGHT

There is no obligation that the numerical values of surface areas used for the individual measurements be equal. Once the measurement has been made, the Surface Area defined, and the JOB:PART:AREA labels assigned, the measurement is stored.

In the Power Summation section of this chapter, we will see how we can use the internal programming of the 3000+ to sum the power from groups of AREAs to obtain power for each PART, and for the entire JOB.

Entering Label Names

To assign the labels to each measurement, press each of the following keys and respond to the prompt on the upper right of the display by typing the desired name using the keypad and pressing **EXIT**.

Softkeys	Softkey Functions
job [I]	The total envelope surface being measured. It is composed of PART surfaces.
part [J]	A subset of the total envelope surface. It is composed of AREA surfaces.
area [K]	A subset of the JOB surface.

AREA has both a label and a numerical value of surface area. After the label has been typed and entered, an entry field will open at the upper right of the screen for input of a numerical value of surface area as indicated by the message “SQUARE METERS = XXXXX”. The value XXXXX in the field will be taken from the AREA field on the lower right. If the value is to be the same, simply press **EXIT**; otherwise input a new value from the keypad before pressing **EXIT**.

As each label is entered, it will appear on the lower right of the display alongside the appropriate designation JOB, PART or AREA. The numerical value of surface area for the AREA label can be edited without changing the AREA label by pressing **meter2 [M]**, inputting a new value and pressing **EXIT.** Be sure to store each measurement after defining the labels and making the measurement.

The user is not obliged to assign labels to measurements, but without them the power summation capability of the 3000+ cannot be utilized. It is possible to assign labels or to modify them after they have been stored, but it may be more convenient to do this at the time of data acquisition and storage.

Selection of Display Parameters

In the Intensity Analysis Mode, the user can choose to display the intensity level spectrum, the SPL level of Channel 1, the particle velocity spectrum or the quality spectrum. Quality is calculated as the ratio of the intensity to the average sound pressure level or in logarithmic format the difference between the intensity level and the average sound pressure level. Quality is used as an indication of the degree to which the intensity data can be taken as an accurate representation of the true intensity.

Selecting Displayed Parameters

To select the spectrum to be displayed, press one of the following:

Softkeys	Softkey Functions
INTNSTY [B]	Displays the Acoustic Intensity Spectrum
QUALITY [C]	Displays the Quality Spectrum
SPL [D]	Sound Pressure Level Spectrum of Channel 1
P.VELOC [E]	Displays the Particle Velocity Spectrum in units of dB re.1 nm/s

These parameters will be associated with a single microphone pair connected to channels 1 and 2.

Readout of Broadband Levels

The two vertical bars displayed to the right of the spectrum (INTNSTY, SPL and P.VELOC displays only) represent the total energy between the high-pass and low-pass filters (denoted by the summation symbol Σ beneath) and the total A-weighted energy between the high-pass and low-pass filters (denoted by the symbol "A" beneath).

Either of these values may be displayed digitally. At any particular time, the digital indication on the right of the screen alongside the indication of the cursor amplitude will be assigned to represent one of these two, as indicated by the symbol “A” or “Σ” alongside the corresponding digital value. Pressing the softkey **SUM [E]** in the Display Menu will toggle this display between these two.

Reducing the Frequency Display Range

In the intensity mode, the default frequency display range is 25 Hz-10 kHz. Depending upon the nature of the sound source being measured, there may be very little sound power emitted above 5 kHz. And, unless the 50 mm spacer is being used, the sound intensity data measured at the lower frequencies will be invalid. In these instances, the horizontal display scaling can be changed to decrease the display range and increase the resolution as described in Chapter 19, section “Control of Horizontal Display”. For example, if the cursor is placed at 1.25 kHz and the horizontal scaling changed to 2, the frequency display range will be 315 Hz-5 kHz.

Reducing the Amplitude Display Range

Due to the relatively small valid dynamic range of intensity measurements, it may be desirable to select a smaller display range (<80 dB) when viewing intensity. The amplitude display range is changed from the Shift Menu by pressing **V.SCALE [C]** and making a selection as described in detail in Chapter 19.

Performing the Intensity Measurement

The most common standards which define the procedure to be followed in performing a measurement of the sound power radiated by a noise source based on sound intensity measurements are the following:

- ISO 9614-1 Acoustics-Determination of sound power levels of noise sources using sound intensity - Measurement at discrete points.

- ISO 9614-2 Acoustics-Determination of sound power levels of noise sources using sound intensity. Part 2: Measurement by scanning.

- ANSI S12.12-1992 Engineering method for the determination of sound power levels of noise sources using sound intensity.

The ANSI (American National Standard Institute) standard is typically followed in the United States, while the ISO standard is more commonly followed in countries other than the United States. Each of these standards are very detailed and complex. It is strongly recommended that the user become familiar with the complete standard appropriate to their measurement situation, as it is outside the scope of this manual to cover all the aspects of these standards.

Since both of these standards call for the use of 1/1 or 1/3 octave bandwidths, from the Filter Menu press **1/1 oct [A]** or **1/3 [B]** to select the desired bandwidth.

Each of these standards employ a single measurement of sound intensity as the average intensity across a representative surface. The ISO standard establishes a number of Field Indicators, one which is a function of time and the others of spacial position, and a detailed procedure which must be followed to achieve a desired grade of accuracy. Later in this chapter is a section describing how these indicators can be evaluated by the Model 3000+ and displayed in a particularly useful format for field applications. The ANSI standard presents an Appendix B (which is not considered to be a part of the standard) in which a number of indicators are described which may be used for evaluating data quality. However, these are included with the standard for guidance and informational purposed only.

The averaging time of the intensity measurement should be set according to the requirements of the particular standard being followed. When using exponential averaging, both the ISO and ANSI standards are satisfied by selecting Constant Bandwidth averaging using a BT product of at least 400. To setup the Model 3000+ for this, from the Main Menu press the softkey sequence **DETECTR [H], BT/EXP [D], AVE.TIME [H], 512 [J], EXIT** to set the instrument for BT = 512 (the 3000+ does not include the value of 400 among the choices of averaging times in this mode). This is a continuously running average. It is up to the user to determine when to stop the measurement. If the user is watching the instrument display, it would be sufficient to run until the displayed spectrum is seen to be stable over the frequency range of interest. Alternatively, the averaging should continue until the run time is equal to or greater than the value of T corresponding to the lowest frequency of interest, T = 512/B, where B is the bandwidth of the lowest frequency filter. The bandwidth is calculated from the center frequency of the filter band as follows:

1/1 Octave Bandwidths: 0.707*Fc

1/3 Octave Bandwidths: 0.23*Fc

The ANSI standard also permits the use of fixed time period averaging. In such a case one could set the instrument to utilize linear averaging with an averaging time equal to or greater than the value of T calculated in the preceding paragraph. This is done from the Intensity Main Menu by pressing the softkey sequence **DETECTR [H], LIN.S [A], AVE.TIME [H]**, using the numeric keypad to enter the value of T, then pressing **EXIT**. When using this linear averaging method, the averaging will stop automatically when the programmed averaging time has passed. When using a Larson Davis intensity probe, a change in the status of the LEDs will inform the user when the measurement has been completed.

The ANSI standard also permits the use of scanning during the measurement, meaning that the intensity probe is moved uniformly over each area segment during the measurement. The result is a single intensity spectrum which has been averaged both in time and spatially over the defined measurement area. Scanning can be combined with either of the averaging methods described in the above paragraphs.

Storage and Recall of Intensity Spectra

To store the displayed Intensity spectrum, press **STORE** which will produce the message “STORE-INTENSITY n” on the upper right of the screen to indicate that the spectrum has been stored into the active file as the nth record of type Intensity. Both the Intensity and the SPL spectra are stored, along with the **JOB**, **PART**, and **AREA** names, the surface area associated with the AREA, and the setup as displayed on the right of the screen. The Quality and Particle Velocity spectra are calculated from the Intensity and SPL spectra, so it is not necessary to store them.

Intensity spectra are recalled from the Intensity Menu by pressing **RECALL** which will produce the message "RECALL-Intensity N" on the upper right of the screen to indicate that the Nth record of type Intensity has been recalled and is being displayed. The message "*recall data" on the lower right of the screen indicates that the horizontal arrow keys are in control of the recall process. Pressing the left arrow key will sequentially recall records lower in number than the one presently recalled, while the right arrow key will sequentially recall records higher in number than the one being displayed.

When performing a recall from the Intensity Menu, the Intensity spectrum will always be displayed, even though the display parameter may have been different (SPL, Quality or Particle Velocity) at the time the recall was initiated. However, following the use of the **KEEP [H]** key to maintain the selected record on the screen, any of those alternative parameters can be displayed by pressing **QUALITY [C]**, **SPL [D]** or **P.VELOC [E**].

Note that Intensity, SPL, Quality and Particle Velocity spectra can also be stored automatically as a function of time or RPM/Speed using the byTime and byTach autostore functions as described in Chapters 15 and 16. Upon recall these data can be displayed in the formats level versus time or level versus RPM/Speed.

Editing the JOB, PART and AREA Names, the surface Area value and the Note Field of a Stored Intensity Spectrum

First recall the spectrum whose parameters are to be edited by pressing **RECALL** and using the horizontal arrow keys to bring to the screen the desired record. Then press **edit [K]** and respond to the message "*ARE YOU SURE?" on the upper right of the screen by pressing **YES [A]**. To abort the edit operation instead, press **NO [C]**.

Each of these parameters can now be edited by pressing the associated softkey, **Note [G]**, **job [I]**, **part [J], area [K]** or **meter2 [M]**, typing in the new entry using the alphanumeric keypad and pressing **EXIT**. To return to the Recall Menu press **EXIT** another time. At this time the modified parameters will replace the original parameters. To return to the Intensity Menu, press **EXIT** one more time.

Power Summation

Stored along with each measurement are the following:

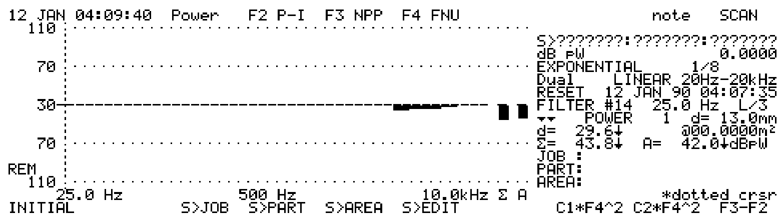
- The labels for JOB:PART:AREA (blank if not assigned)
- A numerical value of surface area corresponding to the measurement for purposes of power calculation
- The measured intensity spectrum
- The SPL spectrum for Channel 1

The Quality and Particle Velocity are not stored because they can be calculated from the intensity and average SPL spectra.

Accessing Power Summation Menu

To perform summations of the stored power spectra, from the Main Menu press **POWER [A]** which will access the Power Summation Menu shown in Figure 20-3.

Figure 20- 3 ***Power Summation Menu***



Search Field Concept

At the upper right of the screen will be the power summation search field “???????:???????:???????”, which defines how the stored intensity spectra are to be summed. The format of the search field is as follows:

JOB_name:PART_name:AREA_name.

If names are inserted into the JOB, PART, and AREA fields in the search field, then all stored spectra which have these same three labels can be summed together to produce a single power spectrum. Since each measurement is unique, there should not be more than one stored spectrum with the same JOB, PART and AREA labels. In that case, the displayed power spectrum will be for a single measurement.

However, if the user specifies names for JOB and PART only, leaving ???????? in the AREA field, for example FAN:FRONT:???????, all spectra having those specified JOB and PART labels (in this example FAN and FRONT), regardless of the AREA labels, will be summed and the resulting power spectrum displayed.

If only a name for JOB is specified, for example FAN:???????:???????, then the summation spectrum will be based upon all the spectra having that specified JOB label (in this example, FAN), regardless of their PART and AREA labels.

Manually Entering Labels into the Search Field

Names can be typed directly into the search field by pressing the key **S>EDIT [L]**. This will result in a flashing cursor below the first character of the JOB field. Use the alphanumeric keypad and the horizontal arrow keys to type the desired characters into the field and press **EXIT**. Note that the names must be literally identical to those used as JOB, PART and AREA names, including blank spaces. For example, if the JOB field before entry of a name is ??????? and the name to be entered is CAR, then the three letters must be followed by four spaces.

Entering Labels by Recalling Spectra

There is a much easier way to enter names into the search field without the necessity of typing them. Before accessing the Power Summation Menu, recall one of the stored spectra which already has the desired labels stored with it. Don't forget to press **KEEP [H]**. These labels will appear in the fields at the lower right of the display as each spectrum is recalled. When the Power Summation Menu is accessed, they will remain there, requiring only the use of the **S>JOB [I], S>PART [J]** and **S>AREA [K]** softkeys to transfer them into the upper right search field.

Performing a Power Summation

When the desired names have been entered into the power summation search field, press **Power [A]** to perform the summation.The spectrum representing the power sum will then be displayed. When the summation has been completed, the search field will no longer be displayed on the upper right of the screen, but the JOB, PART and AREA names associated with the search, and whose spectra have been summed to produce the displayed power spectrum, are indicated on the lower right side of the screen.

NOTE: When in the Power Summation Menu the units indicated on the right of the screen will be POWER in units of dB re. 1 pW (dBpW on the screen) rather than the units INTENSITY in *dB re. 1 pW per square meter (dBpW/m^2 on the screen) which appear when in the Intensity Main Menu.*

As with other spectrum displays, the frequency and amplitude corresponding to the cursor position are displayed digitally on the right of the screen when the SOLID or DOTTED cursor is active. When the BOTH cursor mode is active, the digital display will indicate the total power between the cursor frequencies in both unweighted (to the right of the summation sign) and A-Weighted (to the right of the letter "A") formats, and the cursors will move together in response to presses of the horizontal arrow keys. The level displayed to the right of the delta symbol on the right of the screen is the difference between the amplitude corresponding to the dotted cursor position and that corresponding to the solid cursor position. The ability to readout the power associated with a frequency range larger than a single frequency bandwidth is important when studying the characteristics of broadband noise sources.

Storage of Power Spectra

Power spectra are stored from the Power Summation Menu by pressing **STORE** which will produce the message "STORE-Power N" on the upper right of the screen to indicate that the displayed spectrum has been stored into the active file as the Nth record of type Power.

Recall of Power Spectra

Power spectra are recalled from the Power Summation Menu by pressing **RECALL**, which will produce the message "RECALL-Power N" on the upper right of the screen to indicate that the Nth record of type Power has been recalled from the active files and is being displayed. The message "*recall data" on the lower right of the screen indicates that the horizontal arrow keys are programmed to control which record is being recalled and displayed. Presses of the left arrow key will sequentially recall records lower in number than the one presently recalled, while the right arrow key will sequentially recall records higher in number than the one presently recalled. Once the desired record has been recalled, press **KEEP [H]** to stop the recall operation and return the system to the Power Summation Menu with the recalled spectrum on the display. Pressing **EXIT** instead of **KEEP [H]** will abort the recall process, clear the display and return control to the Power Summation Menu without a recalled spectrum being displayed.

Power Summation Example

Consider a sound intensity measurement project consisting of a JOB named MACHINE that radiates noise from PARTs named FRONT and REAR. To accurately define the noise radiation characteristics of the MACHINE, FRONT and REAR are each divided into four AREAs named UP-LFT, UP-RGT, LOW-LFT, AND LOW-RGT. The intensity of each AREA is measured, and the data is stored in eight Intensity records as depicted in Table 20-1.

Table 20-1 **JOB Named MACHINES**

RECORD NUMBER	PART NAME	AREA NAME
1	FRONT	UP-LFT
2	FRONT	UP-RGT
3	FRONT	LOW-LFT
4	FRONT	LOW-RGT
5	REAR	UP-LFT
6	REAR	UP-RGT
7	REAR	LOW-LFT
8	REAR	LOW-RGT

Three Level Search

To display the power spectrum of the measurement associated with the labels MACHINE:REAR:UP-LEFT, from the Intensity Analysis Menu recall the intensity spectrum from record number 5 because it has those particular label names. Then access the Power Summation Menu and copy the labels into the Power Search Field by pressing **S>JOB [I], S>PART [J]** and **S>AREA [K].** The Power Search Field will now read MACHINE:REAR:UP-FRONT. Press **Power [A]** to perform and display the summation spectrum, which in this case is not really a summation because only record 5 satisfies the search field criteria.

Two Level Search

To calculate and display the power spectrum representing the sum of all the measurements made on the PART named FRONT, from the Intensity Analysis Menu recall record 1, 2, 3 or 4 because any of these have stored with them the desired JOB and PART labels MACHINE and FRONT. To perform the summation, access the Power Summation Menu and press **S>JOB [I]** and **S>PART [J]** so that the Power Search Field now reads MACHINE:FRONT:???????. Pressing **Power [A]** will calculate the summation of the intensity spectra stored in records 1,2,3 and 4 because they all satisfy the search field criteria.

Single Level Search

To calculate and display the power spectrum corresponding to the entire JOB labeled MACHINE, from the Intensity Analysis Menu recall any one of the records 1-8, since all of them have the desired JOB label. Access the Power Summation Menu and press **S>JOB [I]** which will make the Power Search Field read MACHINE:???????:???????. Pressing **Power [A]** will calculate the summation of the intensity spectra stored in records 1 through 8 because all of them satisfy the search filed criteria.

Field Indicators Specified in the Standard ISO 9614-1: 1993 (E)

Temporal Variability Indicator (F_1)

The temporal variability indicator is calculated from a series of short-time-average measurements of intensity, at a single position, as follows:

$$F_1 = \frac{1}{\overline{In}}\sqrt{\frac{1}{M-1}\sum_{K=1}^{M}[Ink - \overline{In}]^2}$$

where:

$\overline{In}$ is the mean value of In for M short time average samples Ink calculated by the formula

$$\overline{In} = \frac{1}{M}\sum_{K=1}^{M_1} Ink$$

With the Model 3000+ we utilize the vsTime autostore capability to evaluate F_1. With the analyzer setup to measure intensity using the desired selection of filter type and bandwidth, pressure/temperature/spacer length and a representative value of surface area, select Linear Repeat Averaging by pressing the following key sequence: **DETECTR [H]**, **LIN R [B]**. Select the desired short-time-average value (the standard recommends 8 - 12 seconds or an integral number of cycles for periodic signals) by pressing **AV.TIME [H]**, entering the desired number using the numeric keypad, and pressing **EXIT**.

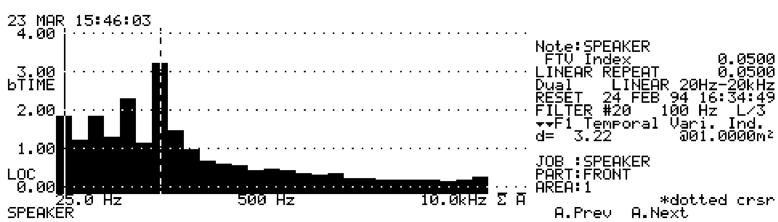
Return to the Intensity Main Menu and access the Autostore Menu by pressing **AUTOSTR [P]**. Set the time interval between autostored spectra to equal the value of the short-time-average value by pressing **delta [C]**, entering the value using the numeric keypad and pressing **EXIT**. The number of short-time-averaged values to be stored during a measurement sequence, M, is established by setting the total time period for the autostore equal to the short-time-average multiplied by M. The standard recommends a value of M = 10. For example, if the short-time-average value is 10, set the value of endstore to M*10 = 100. This is done by pressing **endstor [D]**, entering the value using the numeric keypad, and pressing **EXIT**. Activate the byTime autostore mode by pressing **bytime [B]**, which will produce the message “bTIME” along the vertical axis on the left of the screen.

Place the intensity probe in position relative to the test specimen where the temporal variation is to be determined and press **RUN/STOP** to begin the measurement. A sequence of M spectra, each measured over the short-time-average period, will be stored automatically in a single block at the conclusion of the sequence. Note the record number under which this measurement is stored. This is displayed on the upper right of the screen in the format “STORE By Time Int N” where N is the record number.

If F_1 is to be evaluated at a number of different positions, it is recommended that the Note Field or the **JOB:PART:AREA** fields be defined prior to each measurement to facilitate differentiation between records upon recall.

To display F_1, make sure the instrument is set to the Intensity Mode and that the autostore mode is active (which it will be if the display is performed immediately following the measurement). Press **RECALL** to initiate a recall operation and use the **A.Prev [N]** and **A.Next [O]** softkeys to access the desired record number. When the desired autostore record has been recalled, press **F1 FTV [G]** to display F_1 as a function of frequency, as shown in Figure 20-4. The standard recommends that the values of F_1 not exceed 0.8 over the frequency range of the intensity measurement.

Figure 20- 4 ***F1 Display***



Surface Pressure-intensity Indicator (F_2), Negative Partial Power Indicator (F_3), and Field Non-uniformity Indicator (F_4)

These parameters, calculated from a sequence of measurements made at different spacial positions over a defined surface enclosing the test specimen, are used as indicators of the accuracy of the measurement procedure. The formulas are as follows:

Surface Pressure—Intensity Indicator

$$F_2 = \overline{L_p} - \overline{L_{|In|}} \qquad \textit{eq. (1)}$$

where:

$\overline{L_p}$ is the surface sound pressure level calculated from the equation:

$$\overline{L_p} = 10\log\left[\frac{1}{N}\sum_{i=10}^{N} 10^{0.1L_{pi}}\right]\mathrm{dB} \qquad \textit{eq. (2)}$$

$\overline{L_{|In|}}$ is the surface normal unsigned intensity level calculated from the equation:

$$\overline{L_{|In|}} = 10\log\left[\frac{1}{N}\sum_{i=1}^{N}[|I_{ni}|/I_0]\right]\mathrm{dB} \qquad \textit{eq. (3)}$$

where $|I_{ni}|$ is the unsigned normal sound intensity at measurement position i.

Negative Partial Power Indicator

$$F_3 = \overline{L_p} - \overline{L_{In}} \qquad \textit{eq. (4)}$$

where $\overline{L_p}$ is the surface sound pressure level calculated from equation (2);

$\overline{L_{In}}$ is the surface normal signed intensity level calculated from the equation

$$\overline{L_{In}} = 10\log\left|\frac{1}{N}\sum_{i=1}^{N} I_{ni}/I_0\right|\mathrm{dB} \qquad \textit{eq. (5)}$$

I_{ni} is the signed magnitude of the normal sound intensity component measured at position i on the measurement surface:

I_0 is the reference sound intensity, 10^{-12} Wm^{-2}

NOTE: Where the normal sound intensity component level L_{Ini} at position i is expressed as XX dB, the value of I_{ni} shall be calculated from the equation $Ini = I_0 \times (10^{(XX)/10})$; where the normal sound intensity component level L_{Ini} at position i is expressed as (-) XX dB, the value of L_{Ini} shall be calculated from the equation $Ini = -I_0 \times (10^{(XX)/10})$.

If $\sum I_{ni}/I_0$ is negative, the measurement array does not satisfy the requirements of the standard.

Field Non-Uniformity Indicator

$$F_4 = \frac{1}{\overline{I}_n}\sqrt{\frac{1}{N-1}\sum_{i=1}^{N}[I_{ni}-\overline{I}_n]^2} \qquad \textit{eq. (6)}$$

where $\overline{I}_n$ is the mean value of M short time average samples I_{nk} of I_n calculated from the equation:

$$\overline{I}_n = \frac{1}{M}\sum_{i=1}^{M} I_{nk} \qquad \textit{eq. (7)}$$

In the previous section, Power Summation, it is shown how the sound power can be determined over a surface area named Part by performing a two level search using the field JOB:PART:???????. The search identifies all the intensity measurements stored in the database which correspond to the Area elements which together makeup the larger surface area named Part. From the Power Summation Menu, the indicators defined above can also be determined and displayed as a function of frequency over the same surface area by pressing the following keys:

Softkeys	Softkey Functions
F2 P-I [B]	F_2, Surface Pressure-intensity Indicator
F3 NPP [C]	F_3, Negative Partial Power Indicator
F4 FNU [D]	F_4, Field Non-uniformity Indicator

The resulting display will resemble Figure 20-5, Figure 20-6 and Figure 20-7.

Figure 20-5 ***F2 Display***

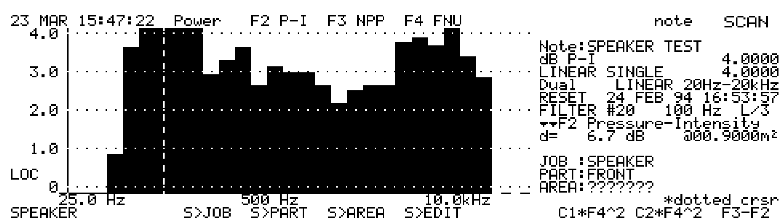


Figure 20-6 ***F3 Display***

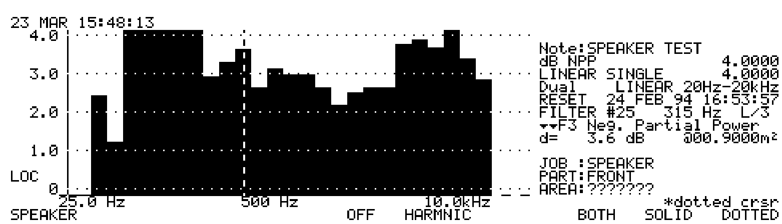
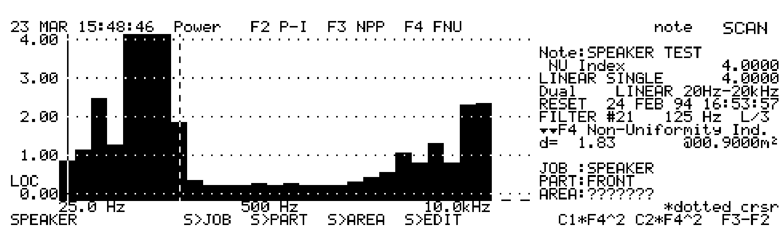


Figure 20-7 ***F4 Display***



In order to calculate these parameters over the complete set of intensity measurements, which represent all the separate area elements contained within all the Parts, use the single level search "JOB:???????:???????". This global evaluation of these parameters should only be used when equal surface area values have been utilized for every intensity measurement.

Alternate Presentation Format for F_2, F_3 and F_4

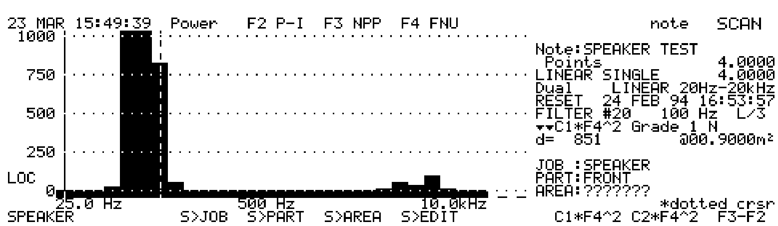
The manner in which these Field Indicators are applied to a sound intensity project requires that further calculations be performed. In the following section, we present several other parameters which can more directly provide the user with the information needed to assess what actions may be required in order to satisfy the requirements of the standard.

The parameter F_4 is used to determine the adequacy of measurement positions (referred to in the standard as criterion 2) as follows: “The number N of probe positions uniformly distributed over a chosen measurement surface is regarded as sufficient i f $N > CF_4^2$ “

The value of N is a function of frequency as well as the degree of precision desired (Precision/Grade 1 or Engineering/Grade 2). To evaluate the adequacy of measurement positions used over a particular Part, perform a two level search “JOB:PART:??????” using that Part name.

To determine and display N as a function of frequency corresponding to the Precision/Grade 1 requirement, press **C1*F4^2 [N]**. The result will resemble Figure 20-8.

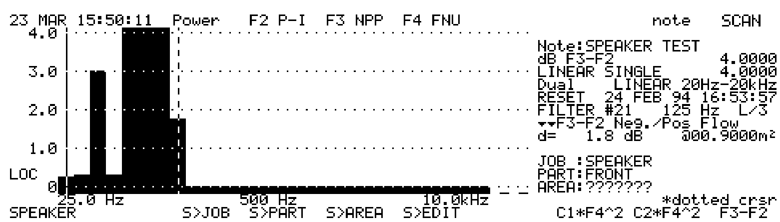
Figure 20- 8 ***C1*F4² Display***



To obtain the same information corresponding to the Engineering/Grade 2 requirement, press **C2*F4^2 [O]**. In either case, if the largest value of N obtained across the frequency range used for the measurement is less than the number of separate measurements (Areas) used within that Part, then this aspect of the standard is satisfied. If that is not the case, repeat the test using a number of measurements (Areas) equal to or greater than N and examine this criterion again.

Another parameter of importance is (F_3 - F_2), which is used in conjunction with criterion 2 to suggest actions to be taken to improve the accuracy of the measurement. Using the same search field (JOB:PART:???????) utilized above to determine the value of N, press **(F3-F2) [P]** to obtain the display shown in Figure 20-9.

Figure 20-9 ***(F3-F2) Display***



The manner in which the standard is written places emphasis on which of the following is obtained:

```
       (F3 - F2) < 1 dB
1 dB < (F3 - F2) < 3 dB
       (F3 - F2) > 3 dB
```

It is a simple matter, using the display format of Figure 20-10, to determine which of these situations corresponds to the measured data over the frequency range of interest.

Field Indicators Specified in the Standard ISO 9614-2: 1994

The section, "Field Indicators Specified in the Standard ISO 9614-1:1993(E)" described the field indicators used for a measurement at discrete points. This section covers the field indicators as specified for a measurement by scanning.

Surface Pressure-intensity Indicator (F_{Ip}), Negative Partial Power Indicator ($F_{+/-}$), and Partial power repeatability check (L_1 - L_2)

Surface Pressure-intensity Indicator (F_{Ip})

The sound field pressure-intensity indicator is calculated as follows:

$$F_{pl} = [L_p] - L_w + 10log\left(\frac{S}{So}\right)dB$$

where:

$[L_p]$ is the surface sound pressure level calculated from the equation:

$$L_p = 10\log\left[\frac{1}{S}\sum_{i=1}^{N} S_i 10^{0.1L_{pi}}\right]\text{dB}$$

where:

S is the total area of the measurement surface =

$$\sum_{i=1}^{N} S_i$$

and S_o is the reference surface area (= 1 m^2).

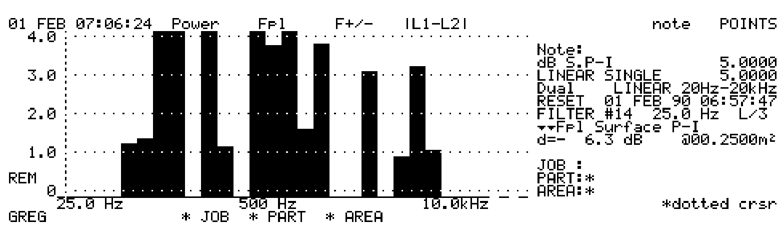
With the analyzer setup to measure intensity using the desired selection of filter type and bandwidth, pressure/temperature/spacer length and a representative value of surface area, select Linear Single Averaging by pressing the following key sequence: DETECTR [H], LIN. S [B]. Select the desired time-average value (the standard recommends at least 20 seconds) by pressing AV.TIME [H], entering the desired number using the numeric keypad, and pressing EXIT.

Place the intensity probe in position relative to the test specimen where the sound field pressure-intensity is to be determined and press **RUN/STOP** to begin the measurement. The analyzer will run for the average time selected and then stop. Press the **STORE** hardkey to store the measurement. This is displayed on the upper right of the screen in the format "STORE - Intensity 1".

If F_{lp} is to be evaluated at a number of different positions, it is recommended that the Note Field or the **JOB:PART:AREA** fields be defined prior to each measurement to facilitate differentiation between records upon recall.

To display F_{lp}, make sure the instrument is set to the Intensity Mode. Press **RECALL** to initiate a recall operation. When the desired autostore record has been recalled, press **Fpl [G]** to display F_{lp} as a function of frequency, as shown in Figure 20-10.

Figure 20-10 ***F_{lp} Display***



Negative Partial Power Indicator ($F_{+/-}$)

This parameter, calculated from a sequence of measurements made at different spacial positions over a defined sur

face enclosing the test specimen, is used as an indicator of the accuracy of the measurement procedure given by:

$$F_{+/-} = 10\log\left[\frac{\Sigma |P_1|}{\Sigma\ P_i}\right] \qquad \textit{eq. (8)}$$

$P_i = < I_{ni} > S_i$

where:

$< I_{ni} >$ is the signed magnitude of the estimated segment-average normal sound intensity measured on the segment i of the measurement surface;

S_i is the area of the segment i;

Also

$|P_i|$ is the magnitude of P_i

Partial power repeatability check (L_1 - L_2)

The following check should be made on the suitability of the measurement conditions:

$|L_{wi}(1) - L_{wi}(2)| \le s$

where:

$L_{wi} = 10\log[|P_i|/P_o]dB$

where:

P_o is the reference sound power = $(10^{-12}W)$

Sound power can be determined over a surface area named Part by performing a two level search using the field JOB:PART:???????. The search identifies all the intensity measurements stored in the database which correspond to the Area elements which together makeup the larger surface area named Part. From the Power Summation Menu, the indicators defined above can also be determined and displayed as a function of frequency over the same surface area by pressing the following keys:

Softkeys	Softkey Functions
Fpl [B]	Surface Pressure-intensity Indicator
F+/- [C]	Negative Partial Power Indicator
L1 - L2 [D]	Partial power repeatability check

The resulting display will resemble Figure 20-11, Figure 20-12 and Figure 20-13.

Figure 20-11 F_{lp} ***Display***

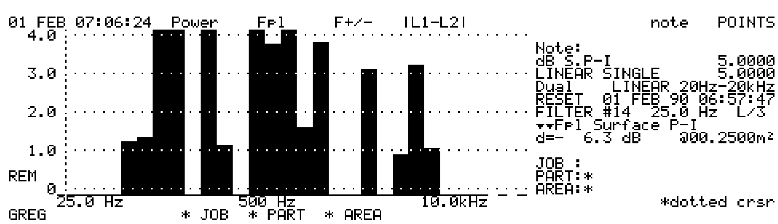


Figure 20-12 ***F+/- Display***

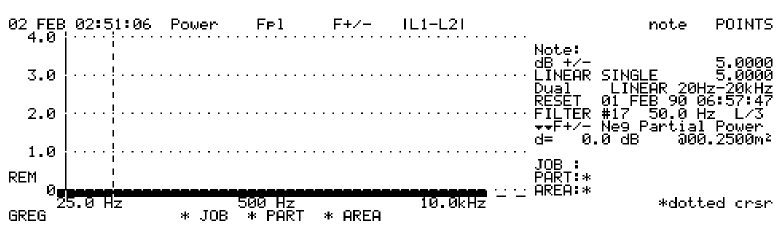
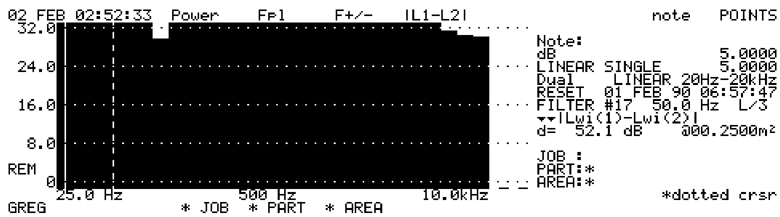


Figure 20-1 3 ***L1 - L2 Display***



In order to calculate these parameters over the complete set of intensity measurements, which represent all the separate area elements contained within all the Parts, use the single level search "JOB:???????:???????". This global evaluation of these parameters should only be used when equal surface area values have been utilized for every intensity measurement.

CHAPTER

21 Room Acoustics Measurements

Sound Decay Measurements

One of the most common acoustics tests performed for the evaluation of room acoustics is the measurement of reverberation time, or early decay time. This is done by injecting acoustic energy into the room, usually with loudspeakers, an exploding balloon or a starter pistol, and then examining the decay of the sound pressure level as a function of time. One generally wishes to evaluate the reverberation time in 1/3 octave frequency bands, since the result of the measurement can be used to determine the sound absorption in the room, which is a function of frequency.

To make such a measurement with the 3000+ it should be configured for Standard Analysis, using 1/3 octave filtering, and with the Autostorage Mode active so that a series of spectra are measured and stored during the sound decay at regular time intervals which are short with respect to the reverberation time to be measured. Typical values of reverberation times are 0.1 to 3 s.

The time decay process is a statistical one, such that there will be a variation of results between tests measured at the same position in the room using apparently identical sound excitation, and between tests made at different measurement points in the room. To achieve good statistical accuracy, numerous tests are generally performed with the microphone at different positions in the room, often performing a number of tests at each position.

In any case, the data for each individual decay measurement will be stored in a unique autostore record. Several autostore decays may be averaged together with the block averaging function. (See Chapter 15, Averaging of Autostore Records.) This will give a smoother decay plot to use for RT60 measurements.

Use of the Noise Generator

When using electronic amplifiers and speakers to excite the room for a reverberation test, the noise generator built into the Model 3000+ is ideal.

Typically one would select Pink Noise in order to provide approximately equal acoustic power per 1/3 octave bandwidth. One would like to achieve a fairly flat sound spectrum over the frequency range of interest in the room previous to beginning the decay measurement, because that provides a good dynamic range at each frequency for the decay measurement. In some cases it is desirable to use a spectrum shaper between the noise generator and the amplification system to optimize the initial sound spectrum. In this description we refer to the Noise Generator. Instruments equipped with the Signal Generator could use either Wideband Pink Noise or 1/3 Octave Bandlimited Noise, which would permit the utilization of the autolevel function.

Procedure

Step 1 Using a microphone input, configure the 3000+ as follows: Standard analysis, 1/3 octave filtering, no autostorage, exponential averaging of 1/8 s.

Step 2 Access the Noise Menu from the System Menu by pressing **NOISE [J]**.

Step 3 Select pink noise by pressing **PINK [M]**, and turn the generator on by pressing **ON [A]**. The noise generator will now deliver a pink noise signal to the amplification system.

Step 4 The horizontal arrow keys will control the output level of the noise generator, indicated by the message “*noise –X.X” where X.X is the output level with respect to the maximum output.

Step 5 Begin measuring by pressing the RUN/STOP key.

Step 6 Set the gain appropriately, and then adjust the noise generator and the sound reproduction system until the room is well excited by a spectrum which is fairly flat in the frequency range of interest. If the level is not high enough, there may not be sufficient range between the exited levels and the background noise to make a meaningful measurement.

Step 7 Change the averaging to Linear Repeat.

Step 8 With respect to the anticipated decay time, set a very short averaging time (say 0.05 s).

Step 9 Put the analyzer into the autostore mode, using an interval equal to the averaging time and an Endstore longer than the anticipated decay time (possibly 2 s).

Step 10 Access the Noise Menu again, and press **OFF/RUN [D]** to put the noise generator into the Off-with-Run mode. Set a delay time long enough for the lowest frequency filter to respond before the noise shuts off.

Step 11 Return to the Standard Analysis Menu and press the **RUN/STOP** key to begin analysis and the autostorage of data.

Step 12 At the completion of the autostore sequence, the autostored record will be recalled and displayed.

Move the cursor to a frequency band of interest and press **vsTIME [E]** to examine the decay curve. Press **DATA [M]** and use the horizontal arrow keys to examine the decay curves for the other frequency bands (See Chapter 15 for a more detailed description of the use of the autostore mode and the control of the display).

Based upon the results of this test, the user may decide to modify the values of averaging time, Endstore and initial excitation sound level to improve the measurement. Autostore records whose data are not satisfactory for later analysis may be deleted.

Use with Impulsive Excitation

When an external source of impulsive noise is to be used for the tests, the Frequency Domain Trigger capability can be utilized.

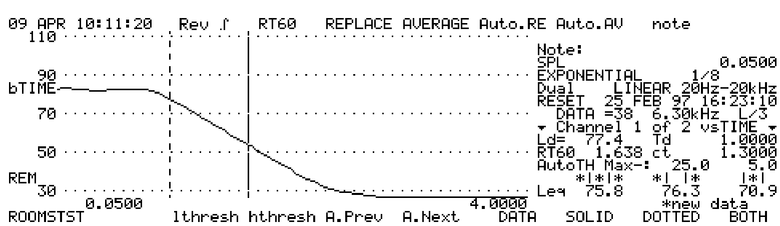
Move the active cursor to a frequency band which will be strongly excited by the source. Set the trigger criteria to be +SLOPE. It may be necessary to experiment with the value of the trigger amplitude until a value is found which triggers reliably, but not too long before the decay begins.

The 3000+ will be configured for Standard Analysis, 1/3 Octave, and autostore mode as described in the preceding section. In some situations, such as music hall acoustics studies, there is much more information to be obtained from the time history of the level in each band than just the reverberation time, such as the timing of arrivals of reflections, the existence of flutter echoes, etc. The 3000+ is ideally suited for these tests as well.

Evaluation of Reverberation Time

A typical decay curve will resemble Figure 21-1.

Figure 21-1 ***Typical Decay Curve***



In this log amplitude versus time format, the major portion of the curve will be linear (with some variations), especially over the first 10 to 20 dB of the decay. Eventually the level will approach the background noise level of the room, which will limit the actual dB range over which the level will decay. The value which is referred to as the reverberation time, RT60, is by convention the time which would be required for the level to decay by 60 dB. In most cases, there is not a sufficient difference between the initial level and the background level for a 60 dB decay to be measured. In practice, one measures the slope of the upper portion of the decay curve over a time interval where the curve is linear (the early decay portion), then extrapolates this to calculate the time which would be required for a 60 dB decay. For example, if the slope of the early decay portion of the curve is 33 dB/s. the RT60 is = 1.8 s.

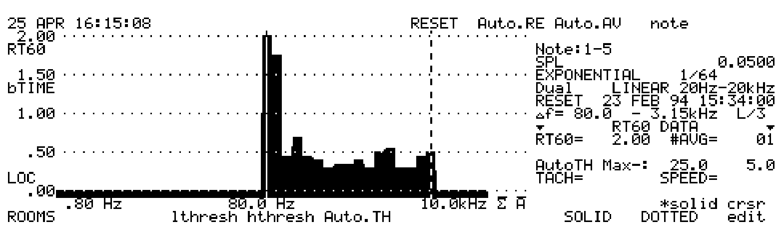
RT60 Register

There is an RT60 register in the 3000+ which can store a value of reverberation time for each frequency band. There are a number of ways by which reverberation time values can be calculated and stored into this register. It is possible to perform averaging in this file, such that a newly calculated value is averaged together with the previously stored value. The RT60 file may be stored to non-volatile memory.

Reading Current RT60

The RT60 Menu, shown in Figure 21-2, is accessed from the vsTime display of a byTime autostore record by pressing **RT60 [B]**.

Figure 21- 2 ***RT60 Menu***



The display indicates the values last stored in the RT60 register.

The cursor can be used to read the value of RT60, and the number of averages which have been used to determine that RT60 value (#AVG).

When evaluating a new set of RT60 values, begin by setting all RT60 values to zero by pressing **R.RT60 [D]**.

Manual Entry of RT60 Values

Suppose the value of RT60 is known from previous measurements for one or more frequency bands, and the user wishes to simply enter these values into the register. Move the cursor to the desired band, press **edit [P]** and respond to the prompt on the upper right of the display by typing in the value using the keypad and pressing **EXIT**. Note that the entered level will appear as a vertical bar on the screen, and that the number of averages will be unity.

One reason the user may wish to enter values manually is to make use of the Transmission Loss calculation software, which requires RT60 values which the user may have previously determined.

Manual Determination of RT60 Using the Cursors

The evaluation of RT60 values from autostored decay time records is done from the Autostore Menu. From the Autostore Menu with the vsTime autostore mode active, recall an autostored decay measurement by pressing **RECALL**. Press **CURSOR**, move to a frequency band of interest, and press **vsTIME [C]** to obtain a decay curve for that frequency band. Use the softkeys **SOLID [N]**, **DOTTED [O]** and **BOTH [P]**, and the horizontal arrow keys to move the solid and dotted cursors such that they define the portion of the curve which is to be used for the determination of the slope of the decay and from that the value of RT60 for that frequency, as shown in Figure 21-1.

When either the solid or the dotted cursor is active, the level and the relative time (with respect to the beginning of the autostore record) of the cursor intersection with the displayed curve are displayed on the right of the screen. When both cursors are active, the level and time differences between the intersections of the two cursors are displayed.

Although the overall form of the early decay curve is linear, there will always be statistical variations of the measured curve with respect to a straight line. In the 3000+, a linear regression is made using the points on the decay curve between the cursors to determine a curve, and the slope of that straight line is used for the determination of the RT60.

The result is displayed on the right of the screen in the format: “RT60 XX.XX ctr Y.YY” where XX.XX is the decay time and Y.YY is the time corresponding to the center of the portion of the curve used for the extraction of the decay time. Both are in units of seconds.

If the RT60 register has not been cleared before beginning this sequence of curvefitting and evaluation of RT60, access the RT60 Menu by pressing **RT60 [B]** and press **RESET [D]**. Respond to the message “Erase RT60 database?” by pressing **YES [A]** unless you have made an error, in which case abort the clear operation by pressing **NO [C]**. Exit back to the vsTime display by pressing **EXIT**.

To enter the value of RT60 for that frequency into the RT60 register, press **REPLACE [C]**. The name implies that this value will replace whatever value was previously in the register at that frequency. One could then access the RT60 Menu (press **RT60 [B]**) to verify that the value has been accepted into the register. Place the active cursor on the band of interest and note the expression “RT60= X.XX #AVE= 01" indicating that the value of RT60 at the cursor position is X.XX and that only a single curvefit has been used to determine this value.

To continue with the manual method of determining RT60 values, press **DATA [M]** and use the horizontal arrow keys to change the frequency of the displayed decay curve and repeat the procedure for curvefitting and storing the result for this frequency into the corresponding RT60 register. With one autostore set of data, the user could thus calculate and store an RT60 value for each frequency of interest without leaving the vsTime Display Menu.

If the user has made several decay measurements, after calculating and storing the RT60 value from one of the autostore records, he could press **A.Prev [K]** or **A.Next [L]** while still in the vsTime Display to bring to the screen the decay curve for the preceding or subsequent measurement, without changing the frequency. Again he could use the cursors to determine the RT60 value, but in this case he would press **AVERAGE [D]** which would average the new RT60 value with the previously stored value, and store the average value in the register. In the RT60 Menu, the value of #AVE which would now be displayed for that band will be two. Continuing in this manner, the user could manually average RT60 values for each frequency band over the entire set of decay measurements. The number of averages used for each frequency band need not be the same.

Although this manual method is slower than the automatic method described below, there is an advantage in that each decay curve can be examined by the user, and the best time segment of each used for evaluating the RT60 value. With the automatic method, there can be particular problems obtaining satisfactory decay curves at the low and high frequency limits. The manual method permits the careful selection of data to be used.

Automatic Determination of RT60 Using Max–based Thresholds

A common practice in the determination of RT60 values from sound decay curves is to apply the curvefit to the portion of the decay curve beginning when the sound level has decayed to 5 dB below the initial noise level and ending when the level has decayed even further below the initial noise level, 25 dB for example. This is achieved automatically in the Model 3000+ using Max–based threshold levels

From the vsTime Menu, if the text on the 9th line down on the right side of the screen reads “AutoTH Max–: XX YY”, then the Max–based Threshold mode is already active. If the message reads “THRESHOLDS: XX YY”, then the Fixed Thresholds (described in the following section) are active. To modify the mode of the thresholds, access the RT60 Menu by pressing **RT60 [B]** and note that repeated presses of **AutoTH [K]** toggles the threshold mode between Max–based and Fixed as indicated by the text on the right of the screen. Select the Max–based mode and press **EXIT** to return to the vsTime Menu.

The beginning of the portion of the decay curve used for the curvefit is defined by the upper threshold. Select the upper threshold by pressing **hthresh [J]**, which will produce the message “ENTER THRESHOLD nnn.n” on the upper right of the screen. Use the numeric keypad to enter a value which will define the beginning of the curvefit time interval as the instant when the sound level has decayed to nnn.n dB below the highest value which had occurred during the measurement. In the example above, this would be 025.0 dB. Press **EXIT** to accept the value, which will then appear as YY.Y in the text field “AutoTH Max–: XX.X   YY.Y” on the right of the screen.

The end of the portion of the decay curve used for the curvefit is defined by the lower threshold. Select the lower threshold by pressing **lthresh [I]**, which will produce the message “ENTER THRESHOLD nnn.n” on the upper right of the screen. Use the numeric keypad to enter a value which will define the end of the curvefit time interval as the instant when the sound has decayed to nnn.n dB below the highest value which had occurred during the measurement. In the example above, this would be 020.0 dB. Press **EXIT** to accept the value, which will then appear as XX.X in the text field “AutoTH Max–:XX.X YY.Y” on the right of the screen.

To perform automatic curvefitting, access the RT60 Menu from the vsTime Menu by pressing **RT60 [B]**. If this is to be the first curvefit made with this set of decay records, press the softkey sequence **RESET [D]**, **YES [A]** to reset the RT60 register.

The automatic curvefitting process may be applied to a limited number of frequency bands if desired. This is done by positioning the solid and dotted cursors along the frequency axis such that they enclose just those frequency bands for which the curvefit is to be performed. Use the softkeys **SOLID [N]**, **DOTTED [O]** and **BOTH [P]** along with the horizontal arrow keys to define this region; the **CURSOR** hardkey will not function from the RT60 Menu. The frequency range between the cursors is indicated digitally on the right of the screen, 5th line down, by the message "Δf = XX – YY" where XX is the lower frequency limit and YY is the upper frequency limit. If it is desired to apply the curvefit to the entire set of frequency bands, place the cursors at the two extreme limits of the frequency axis.

After positioning the cursors, pressing **Auto.RE [E]** will initiate the following sequence for each filter band between them:

Step 1 In the time domain, the upper and lower threshold values, relative to the maximum level during the measurement time, are used to define the time interval over which the curvefit is to be performed. In some cases where the background noise level is higher than the level corresponding to the initial noise level minus the decay of the lower threshold, the decay curve will not cross the lower threshold. In such a case, the curvefit for that frequency is aborted and a value of zero is assigned as the RT60 time.

Step 2 A least–squares best–fit is performed on the portion of the decay curve within the time interval defined by the two thresholds to determine the slope of the decay.

Step 3 From the slope of the decay curve, the data is extrapolated to produce the RT60 value, in seconds, to represent the time which would have been required for a decay of 60 seconds at the same rate of decay.

Step 4 The RT60 value for that frequency band is stored in the RT60 register, replacing whatever value had been there previously. The .RE in the **Auto.RE [E]** implies that the new value REPLACES the previous value.

At the conclusion of the automatic curvefit procedure, the RT60 values for all the frequency bands between the cursors will be displayed. The message "THRESHOLD NOT MET–

NO RT60" means that for at least one frequency band, the condition described in (1) above has occurred and, therefore, in at least one band an RT60 value of zero will occur in the display. Even when this message appears and the zero value occurs for one or more frequency bands, for all the frequency bands where the curvefit has been successfully applied RT60 values will be stored in the RT60 register and they will appear in the RT60 display. As the cursor is moved across the RT60 display, the RT60 value for the indicated frequency will be displayed digitally on the right of the screen, 7th line down, along with the number of averages used to calculate the RT60 value. At this instant, since the RT60 was reset before beginning, the message will read "#AVE = 01" for all bands having a non–zero value of RT60.

In those cases where the automatic curvefit has failed to provide a non–zero value of RT60, it is recommended that the user examine the decay curve to determine why this has happened. In extreme conditions it may be necessary to utilize a manual curvefit between cursors for each troublesome frequency band to determine a meaningful RT60 value and to store it in the RT60 register.

For statistical accuracy, it is common to make multiple decay measurements at each microphone position and to average the RT60 values determined for each decay together in the RT60 register. When the RT60 register already contains values, pressing the key **Auto.AV [F]** instead of **Auto.RE [E]** will cause the RT60 values automatically measured at each frequency to be averaged with the value (or values) already contained in the RT60 register rather than to replace them. Using the vsTime autostore synchronized with the noise generator, it is easy to rapidly measure multiple decays, each of which will be stored in a separate vsTime record. Note that the softkeys **Auto.RE [E]** and **Auto.A [F]** are available in the vsTime Menu as well as the RT60 Menu. When the data are well behaved, it is more convenient and rapid to do the curvefitting from the RT60 Menu because the **A.Prev [K]** and **A.Next [L]** keys used for moving between different vsTime autostored blocks are in that Menu. Using the automatic curve fit technique, the user would recall the first vsTime record and determine a set of RT60 values using the key **Auto.RE [E]**. Then, upon pressing **A.Next [L]**, the next vsTime record will be recalled. Pressing **Auto.Av [F]** will then produce a second set of RT60 values based on the second vsTime decay record and these would be averaged with the first set of RT60 values already in the RT60 register. Continuing in this manner each of the vsTime records would be sequentially recalled and automatic curve fit performed to produce RT60 values based on the average of a number of separate decay measurements.

At any time during the recall and curvefitting process the user can access the RT60 Menu to display the RT60 values. Using the cursor, both the average value of RT60 and the number of averages upon which the value is based are displayed for the frequency corresponding to the cursor position.

Automatic Determination of RT60 Using Fixed Thresholds

This procedure is exactly the same as that described above, with the exception that the upper and lower threshold values used to define the portion of the time decay curve over which the curvefit is to be performed are actual levels of the decay curve rather than the difference between the decay curve level and maximum noise level previous to the decay process. For example, one could select to perform the curve fit over the time interval during which the level decays from 90 to 70 dB. To select the Fixed Threshold mode, from the RT60 Menu press **Auto.TH [K]** until the message on the right of the display, 9th line down, reads “THRESHOLDS: XX.X YY.Y” and press **EXIT**.

Select the upper threshold by pressing **hthresh [J]**, which will produce the message “ENTER THRESHOLD nnn.n”. Use the numeric keypad to enter a value which will define the beginning of the curvefit time interval as the instant when the sound level equals nnn.n dB. In the example above this would be 090.0 dB. Press **EXIT** to accept the value, which will then appear as YY.Y in the text field “THRESHOLDS: XX.X YY.Y” on the right of the screen.

Select the lower threshold in the same manner by pressing **lthresh [I]**, which will produce the message “ENTER THRESHOLD nnn.n”. Use the numeric keypad to enter a value which will define the end of the curvefit time interval as the instant the sound level has decayed to nnn.n dB. In the example above this would be 070.0 dB. Press **EXIT** to accept the value, which will then appear as XX.X in the text field “THRESHOLDS: XX.X   YY.Y” on the right of the screen.

The curve fit are performed as described for the Max–based thresholds. The key **Auto.RE [E]** will initiate an automatic curvefit based on the fixed thresholds and the RT60 values will replace the values previously stored in the RT60 register. The key **Auto.AV [F]** will initiate an automatic curvefit based on the fixed thresholds and the RT60 values will be averaged with those already stored in the RT60 register. There is one aspect of using the fixed thresholds to bear in mind, however. When using fixed thresholds, it is a required condition that the decay curve at each frequency begin above the upper threshold and decay to a level below the lower threshold. For any frequency band for which both conditions are not satisfied, the curvefit procedure is aborted and the value RT60 is assigned to be zero. This produces the message “THRESHOLD NOT MET–NO RT60” to warn that for at least one frequency band a zero value of RT60 will appear in the RT60 register. When using Max–based thresholds, the upper threshold will always be below the initial noise level, so the only instance where the calculation of the RT60 at a particular frequency would be aborted will be when the decay curve reaches the background noise level before the lower threshold condition is met.

Averaging of Autostored Time Decay Records

Rather than calculating RT60 values for each time decay curve and averaging these together, one might prefer to average together the original time decay curves measured for a number of tests in order to obtain a single averaged decay curve and then use this curve to calculate RT60 values.

A general description of the averaging of autostore records is presented at the end of Chapter 15. It must be remembered that the data in the different autostore records are averaged together bin-by-bin. This means first of all that the spectral storage rate (DELTA) used during the acquisition must be the same for all records. Also, the beginning of the decay for each record should occur at approximately the same number of time intervals from the beginning of the record. This can be done by using the noise generator in the “OFF/RUN” mode and using the same value of Delay Time, since this determines the instant of the shut-off of the noise generator relative to the beginning of each autostore sequence.

If a large number of decays are to be measured over a single testing period, typically a number of decays at each of a number of microphone positions, there may not be sufficient memory in the 3000+ to store an autostore record for each. It may be desirable from a memory utilization standpoint to measure multiple decays at each microphone position, average these together, then delete the original decay records and save only the averaged decay curve for later calculation of RT60 values for each microphone position. This block averaging of decay curves, followed by record deletion, can be done much more rapidly in the field than evaluating and storing RT60 values for each decay before deleting the autostore records, which is another alternative for the efficient use of memory.

Storage and Recall of RT60 Data

When a set of RT60 values have been determined and saved in the RT60 register using one of the methods described previously, store the register to memory from the RT60 Menu by pressing **STORE**. This will produce the message “STORE -RT60 N” on the upper right of the screen indicating that it has been stored into the active file as the Nth record of type RT60.

Do not forget that a note field may be created and stored with each RT60 record, which may be very useful later for differentiating between different stored records.

Recall of RT60

To recall an RT60 data block from the active memory file, the 3000+ must be in the RT60 Menu. Pressing **RECALL** will produce the message “Overwrite ALL RT60 data?” indicating that if the RT60 recall operation is continued, this newly recalled data will take the place of the data previously in the RT60 register. To continue, press **YES [A]**. To abort the recall and save the data which is presently in the RT60 register, press **NO [C]**.

Upon continuing the recall, the message “RECALL - RT60 N” on the upper right of the screen will indicate that the Nth RT60 record has been recalled from the active memory and placed in the RT60 register, which is now being displayed. The message “*recall data” on the lower right of the screen indicates that the horizontal arrow keys are assigned to recall RT60 records. Press the horizontal arrow keys to recall and display the particular RT60 record which is desired. Press **CURSOR** to reassign the horizontal arrow keys so they will no longer control the record recall function.

Room Acoustics Measurements

Airborne sound transmission loss is a measure of the acoustical isolation provided between adjacent rooms or spaces by walls or partition elements such as floor-ceiling assemblies, doors, windows or roofs. It is used to estimate the level of noise which will exist in a room containing no sources of noise due to the presence of a noise source within an adjacent room.

Impact isolation is a measure of the impact sound insulation of a floor-ceiling assembly and associated supporting structures. It is used to estimate the level of noise which will exist in a room containing no sources of noise due to the presence of impact excitation on the upper side of the floor-ceiling assembly, such as footsteps on the floor of the space above the room.

There exist clearly established standards for the measurement and calculation of the parameters associated with both airborne sound transmission loss and impact sound insulation. It is essential that the user be familiar with these standards before undertaking such tests, since there are many fine details which must be addressed precisely during the measurement phase of the project. These details are beyond the scope of this manual.

The firmware within the Model 3000+ is designed to perform the calculations as specified in these standards, but the accuracy of the final results is highly dependent upon the proper attention to measurement details.

There are two different standards organizations whose standards are followed by the majority of the acoustic professionals. In the United States, the standards from the American Society of Testing Materials (ASTM) are usually appropriate, while in the remainder of the world the standards from the International Standards Organization (ISO) are most commonly followed. Many countries have also instituted their own national standards, but in most cases these are identical in procedure to the ISO standards. As you will see in the following sections, the calculations of the acoustic parameters done within the Model 3000+ can be performed according to either ASTM or ISO standards.

Airborne Sound Transmission Loss Measurements

The determination of the airborne sound transmission loss, either in a laboratory or the field, requires a noise source, typically a loudspeaker, within one room which is designated the Source Room. The adjacent room, designated the Receiving Room, will be excited acoustically by energy transmitted through the wall or partition element between the two rooms. In many cases the loudspeaker will be excited to produce broadband noise, typically pink noise. With the speaker operating, the space-averaged 1/3 octave sound pressure level spectra are measured within both the Source and the Receiving Room. The space-averaged spectrum may be determined by making a number of separate measurements at different locations within the room and performing a block average operation. Another technique is to mount the microphone and preamplifier on a rotating microphone boom and perform a linear single average over a time interval which represents several complete rotations of the boom. If using a Model 3000+, it is possible to make measurements in the Source and Receiving Rooms simultaneously by utilizing two microphones.

In addition to the Source Room and Receiving Room spectra, a measurement of the background noise in the Receiving

Room is performed without excitation of the Source Room. In cases where the spectrum levels measured in the Receiving Room during excitation from the Source Room are close in magnitude to those of the background spectrum levels, the standards call for a correction factor to be applied to the measured Receiving Room levels to account for the effect of the background noise on the measurement.

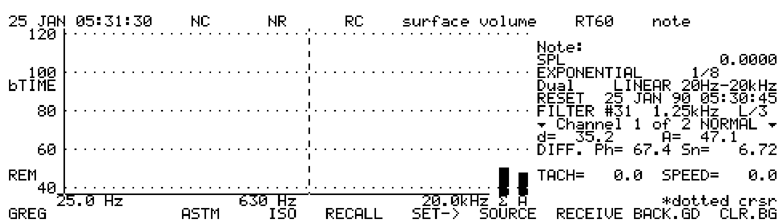
The fourth parameter which must be measured is the sound decay time, RT60, within the Receiving Room. This is typically done by moving the speaker from the Source Room to the Receiving Room and following the procedures described earlier in this Chapter of the manual.

At the conclusion of the measurement phase of the project, the following data will have been measured and stored in the analyzer.

- Source Room spectrum (space-averaged)
- Receiving Room spectrum (space-averaged)
- Receiving Room background spectrum
- Sound decay time (RT60)

To calculate and display the airborne sound transmission parameters, access the Rooms Menu, as shown in Figure 21-3, from the Main Menu by pressing **ROOMS [I]**.

Figure 21- 3 ***Rooms Menu***



Input values of the Test Partition Surface Area (m^2) and the Receiving Room Volume (m^3) by pressing **surface [D]** and **volume [E]**, respectively, typing in the value via the numeric keypad, and pressing **EXIT**.

Recall the space-averaged Source Room spectrum and press the key sequence **SET-> [I]**, **SOURCE [M]** to define it as such for the calculation. Similarly, recall the space-averaged Receiving Room spectrum, press **SET-> [I]**, **RECEIVE [N]**, and recall the Receiving Room background spectrum and press **SET-> [I]**, **BACK.GD [O]** to define them for the calculation. Once they have been defined, these data blocks can be displayed by pressing **SOURCE [M], RECEIVE [N]**, or **BACK.GD [O]** to examine the Source Room, Receiving Room or Receiving Room background spectra, respectively. Only the data in the frequency range 100 Hz-4 kHz are saved in these files since the analysis is limited to that frequency range.

In some cases a user may be confident that the Receiving Room spectrum is sufficiently above the background spectrum that no correction will be necessary and not wish to measure the background spectrum. In this case, simply press **CLR.BG [P]** to reset the levels of the background spectrum to be used for the calculation to zero dB in all frequency bands.

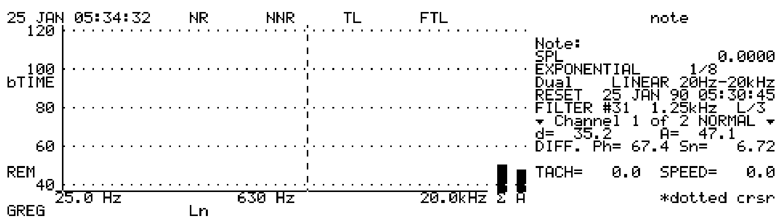
To define the RT60 record to be used for the calculation, press **RT60 [F]** to access the RT60 Menu. If the desired record is already in that buffer, simply press **EXIT** to return to the Rooms Menu. Otherwise, press **RECALL**, **YES [A]**, use the horizontal arrow keys to recall the desired record, then press **EXIT**.

Select the standards organization whose standards you wish to follow for the determination of the airborne sound transmission parameters by pressing either **ASTM [I]** or **ISO [J]**.

ASTM Airborne Sound Transmission Parameters

Pressing **ASTM [I]** will produce the ASTM Rooms Menu as shown in Figure 21-4.

Figure 21- 4***ASTM Rooms Menu***



The following lists a set of parameters which are defined by the ASTM standards.

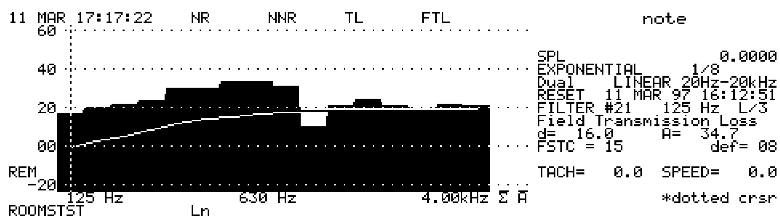
Parameter	Standard
Noise Reduction (NR)	ASTM E90-90 and E336-90
Normalized Noise Reduction (NNR)	ASTM 336-90
Transmission Loss (TL)	ASTM E90-90
Field Transmission Loss (FTL)	ASTM E336-90

To perform the calculation and display the result as a function of frequency on the analyzer screen, simply press the softkey whose label corresponds to that parameter. (The parameter Ln is discussed in a later section on Impact Noise Isolation.)

If a message “WARNING: High Background” appears on the upper right of the screen, this means that the difference between the Receiving Room spectrum and the Receiving Room background spectrum levels at one frequency or more is less than 5 dB. In such a case, the corrected spectrum can only be used as an estimate of the upper limit of the impact noise level and this should be noted in the report.

As an example, Figure 21-5 shows a display of Field Transmission Loss.

Figure 21-5 ***Field Transmission Loss Display***



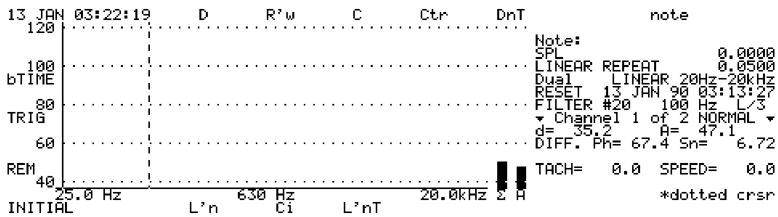
ASTM Standard E413-87 defines single number rating indices corresponding to each of the above parameters which are determined by a curvefitting procedure described in the standard. When each of the above parameters are displayed as a function of frequency, the results of the curvefitting procedure are overlaid on the screen as well. The calculated single number rating index (or indices) are displayed on the lower right of the screen. The message “def = XX” which appears to the right of the index value represents the sum of the deficiencies above the curvefit line as described in the standard.

Parameter	**Index**
Noise Reduction (NR)	Noise Isolation Class (NIC)
Normalized Noise Reduction (NNR)	Normalized Noise Isolation Class (NNIC)
Transmission Loss (TL)	Sound Transmission Class (STC)
Field Transmission Loss (FTL)	Field Sound Transmission Class (FSTC)

ISO Airborne Sound Transmission Parameter

Pressing **ISO [J]** will produce the ISO Rooms Menu as shown in Figure 21-6.

Figure 21- 6 ***ISO Rooms Menu***



The following lists a set of parameters which are defined by the standard ISO 140/4 1978 Field measurements of air-borne sound insulation between rooms. Another standard, ISO 140/3 1978 Laboratory measurements of airborne sound insulation of building elements defines sound reduction index (R), but in most cases this will be the same as the apparent sound reduction index (R') defined in 140/4. Thus, we use the symbol R' to represent both for these calculations.

Parameter	Standard
Apparent Sound Reduction Index (R')	ISO 140/3-1978 (E) and ISO 140/4-1978 (E)
Level Difference (D)	ISO 140/4-1978 (E)
Standardized Level Difference (D_{nT})	ISO 140/4-1978 (E)

The standard ISO 717/1 1982 defines single number rating indices corresponding to each of these parameters which are determined by a curvefitting procedure described in the standard. When each of the above parameters are displayed as a function of frequency, the results of the curvefitting procedure are overlaid as well. The calculated single number rating index (or indices) are displayed on the lower right of the screen. The message “def = XX” which appears to the right of the index value represents the sum of the deficiencies above the curvefit line as described in the standard.

To perform the calculation and display the result as a function of frequency on the analyzer screen, simply press the softkey whose label corresponds to that parameter.

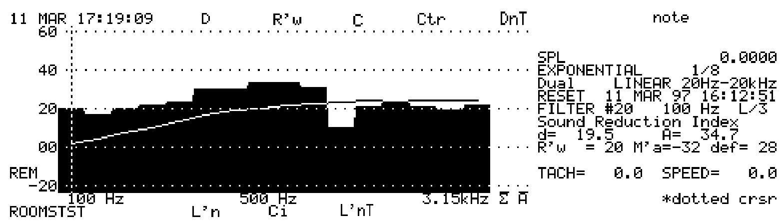
The parameters L_n, $L_{nT,}$ and C_i are discussed in a later section on Impact Sound Insulation.

As an example, Figure 21-7 shows a display of Sound Reduction Index.

If a message “WARNING: High Background” appears on the upper right of the screen, this means that the difference between the Receiving Room spectrum and the Receiving Room background spectrum levels at one frequency or more is less than 3 dB. In such a case, precise values for the Receiving Room spectrum levels cannot be determined and the results should not be considered valid.

If a message “Deviation > 8 dB” appears on the upper right of the screen, it means that a maximum unfavorable deviation greater than 8 dB has occurred in at least one frequency band when determining the values of the Indices.

Figure 21-7 ***Sound Reduction Index***



Impact Sound Insulation Measurements

Parameter	Index
Level Difference (D)	Weighted Level Difference (D_W)
Apparent Sound Reduction Index (R’)	Apparent Weighted Sound Reduction Index (R'_W) and Airborne Sound Insulation Margin (M'_a)
Standardized Level Difference (D_{nT})	Weighted Apparent Standardized Sound Reduction Index ($D_{nT,W}$)

The determination of Impact Sound Insulation, whether in a laboratory or in the field, requires the use of a standardized tapping machine to deliver impacts on the floor of the upper space, which acts as a source of sound generation in the room below. A space-averaged 1/3 octave sound pressure spectrum of the sound in the test room below is measured while the tapping machine is in operation. It is very important to follow the standard carefully, as it may be necessary to measure not only at different points within the test room, but also with the tapping machine placed at several different position on the floor of the upper room. The use of the block averaging function will serve to combine the spectra from these different measurements into a single space-averaged spectrum as required.

A background spectrum is also measured in the test room while the tapping machine is not being operated.

As is done for the measurement of airborne sound transmission, it is necessary to measure the sound decay time (RT60) in the test room as described earlier in this Chapter. Actually, the procedure for the determination of Impact Sound Insulation is similar to that for airborne sound transmission loss, with the exception being that the tapping machine produces the acoustic excitation in the test room so there is no Source Room as such, just a Receiving Room.

At the conclusion of the measurement phase of the project, the following data will have been measured and stored in the analyzer.

- Test Room (Receiving Room) spectrum (space-averaged, possibly source position averaged as well)
- Test Room background spectrum
- Sound decay time (RT60)

To determine the impact sound insulation parameters, access the Rooms Menu, as shown in Figure 21-3, from the Main Menu by pressing **ROOMS [I]**.

Input the value of the Test Room volume (m^3) by pressing **volume [E],** typing the value via the numeric keypad, and press **EXIT**.

Recall the space-averaged Test Room spectrum and press **RECEIVE [N]** to define it as such for the calculation. Then, recall the Test Room background spectrum and press **BACK.GD [O]** to define that for the calculation. In some cases a user may be confident that the Test Room spectrum is sufficiently above the background spectrum that no correction will be necessary and they do not wish to measure the background spectrum. In this case, simply press **CLR.BG [P]** to reset the levels of the background spectrum to be used for the calculation to zero dB in all frequency bands.

To define the RT60 record to be used for the calculation, press **RT60 [F]** to access the RT60 Menu. If the desired record is already in that buffer, simply press **EXIT** to return to the Rooms Menu. Otherwise, press **RECALL**, **YES [A]**, use the horizontal arrow keys to recall the desired record, then press **EXIT**.

Select the standards organization whose standards you wish to follow for the determination of the impact sound insulation parameters by pressing either **ASTM [I]** or **ISO [J]**.

ASTM Impact Sound Transmission

Pressing **ASTM [I]** will produce the ASTM Rooms Menu as shown in Figure 21-4.

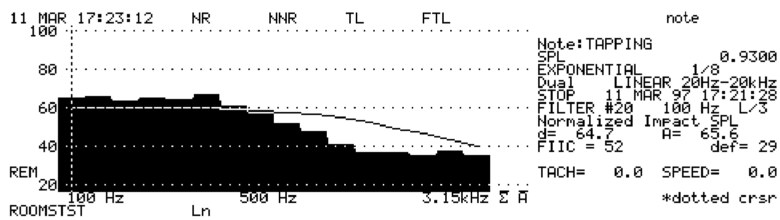
The standard ASTM E1007-90 Field Measurement of Tapping Machine Impact Sound Transmission Through Floor-Ceiling Assemblies and Associated Support Structures describes the procedure for determining the Normalized Impact Sound Pressure Level (Ln) from the measured data. Another standard, ASTM E492-90 Impact Noise Isolation, Laboratory is appropriate for laboratory measurements, but the actual calculation of the Normalized Impact Sound Pressure Level is the same as used in the field.

The standard ASTM 989-89 Impact Isolation Class describes the determination of Field Impact Insulation Class (FIIC)_from the Normalized Impact Sound Pressure Level using a curvefitting technique. The same calculation is applied to the laboratory measurements to obtain the Impact Insulation Class (IIC).

To determine these ASTM parameters, simply press **Ln [I]** to obtain a display similar to Figure 21-8.

If a message “WARNING: High Background” appears on the upper right of the screen, this means that the difference between the Test Room spectrum and the Test Room background spectrum levels at one frequency or more is less than 5 dB. In such a case, the corrected spectrum can only be used as an estimate of the upper limit of the impact noise level and this should be noted in the report.

Figure 21- 8***Normalized Impact Sound Pressure Level Display***



ISO Impact Isolation

Pressing **ISO [J]** will produce the ISO Rooms Menu as shown in Figure 21-6.

The following parameters are defined by the standard ISO 140/7 1978 <u>Field measurements of impact sound insulation of floors</u>. Another standard, ISO 140/6 1978 <u>Laboratory measurement of insulation of floors</u> is appropriate for laboratory testing, but the calculation procedure is the same as used for the field measurements. Thus, we use the parameter symbols with a prime, denoting field measurements, in the analyzer firmware.

Parameter	**Standard**
Normalized Impact Sound Pressure Level (L'_n)	ISO 140/7-1978 (E) and ISO 140/8-1978 (E)
Standardized Impact Sound Pressure Level (L'_{nT})	ISO 140/8-1978 (E)

The standard ISO 717/2 1982 defines single number rating indices corresponding to each of these parameters which are determined by a curvefitting procedure described in the standard.

Parameter	Index
Normalized Impact Sound Pressure Level (L'_n)	Weighted Normalized Impact Sound Pressure Level ($L'_{n,W}$) and Impact Sound Protection Margin (M'_i)
Standardized Impact Sound Pressure Level (L'_{nT})	Weighted Standardized Impact Sound Pressure Level ($L'_{nT,W}$)

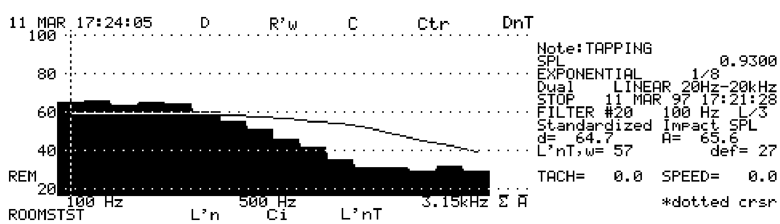
To determine and display these ISO parameters, simply press the softkey whose label corresponds to the desired parameter.

If a message “WARNING: High Background” appears on the upper right of the screen, this means that the difference between the Test Room spectrum and the Test Room background spectrum levels at one frequency or more is less than 3 dB. In such a case, precise values for the Receiving Room spectrum levels cannot be determined and the results should not be considered valid.

If a message “Deviation > 8 dB” appears on the upper right of the screen, it means that a maximum unfavorable deviation greater than 8 dB has occurred in at least one frequency band when determining the values of the Indices.

As an example, Figure 21-9 shows an example of a Standardized Impact Sound Pressure Level display.

Figure 21-9 ***Standardized Impact Sound Pressure Level***

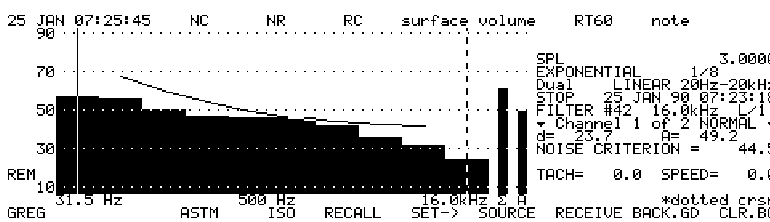


Noise Criteria Curves

A single number technique for representing the character of steady indoor background noise is based on the 1957 Noise Criteria Curves. These are a set of similar octave band reference curves which are overlaid graphically upon an octave frequency spectrum measured in the room to represent the background noise. Each curve is designated by a number, which represents the value of sound pressure level corresponding to the 1 kHz band. The user seeks that reference curve which is not exceeded by any of the background noise octave bandwidth levels but which is as close to touching one of them as possible.

In the 3000+ this function can be performed automatically using a displayed 1/1 octave spectrum which has either just been measured, or has been recalled from memory. If the spectrum has been measured using 1/3 octaves, it must first be converted to the 1/1 octave bandwidth format from the Display Menu by pressing **1/1 [A]**. Access the Rooms Menu from the Analysis Menu by pressing **ROOMS [I]**, and then press **NC [A]** to obtain the display shown in Figure 21-10. The best-fit NC curve is displayed as an overlay with the spectrum, and the NC value corresponding to that curve is displayed digitally on the right side of the screen in the format “NOISE CRITERION = XX.X”. The NC display cannot be stored to internal memory, but it can be printed out in the usual manner.

Figure 21-1 0 ***NC Display***



Noise Rating Curves

The Noise Rating (NR) Curves are used in the same manner as the Noise Criteria (NC) Curves to produce a single number rating of steady background noise according to the ISO Recommendation ISO/R–1996–1971, Acoustics–Assessment of Noise with Respect to Community Response. With the spectrum of the background noise displayed in a 1/1 octave bandwidth format, from the Rooms Menu press **NR [B]**.

The RC Noise Rating Procedure

The RC Noise Rating Procedure is presented in the ANSI Standard ANSI S12.2-1995 Criteria for Evaluating Room Noise. In addition to producing a single number rating of the background steady background noise, the quality of the spectra is described in terms of one or more of the following:

Neutral Spectrum (N)

Rumbly Spectrum (R)

Hissy Spectrum (H)

Acoustically Induced Perceptible Vibration (Va and/or Vb)

The spectrum measurement must use either the 1Hz-10kHz or the 1 Hz-20 kHz frequency range to have the required low frequency bands. As with the NC and NR rating procedures, the steady background noise spectrum is displayed in the 1⁄1 octave bandwidths format. Then, from the Rooms Menu press **RC [C]**.

CHAPTER

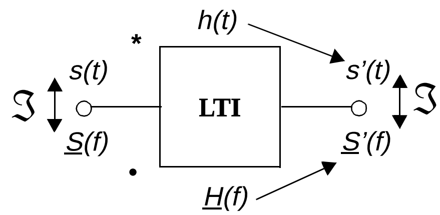
22 Measurements Using Maximum Length Sequences (MLS)

Most of the measurements 3000+ users are familiar with are made in the frequency domain; spectrum analysis and system measurements such as transfer functions, coherence, etc. The MLS technique uses a periodic sequence of pulses having a particular pattern to excite a system. Measurements are made in the time domain making use of the deterministic nature of the sequence to reduce the signal-to-noise ratio by performing time synchronous averaging on the measured signal. As will be seen in this chapter, the primary application of this technique is to perform accurate measurements in the presence of high levels of random background noise.

A Little Theory

Most of the measurement and analysis techniques we utilize assume that the system is linear and time-invariant. In the case of acoustic and electroacoustic systems this is usually a valid assumption. For a linear, time-invariant system one can perform analysis in either the time or the frequency domain, and move between them via the Fourier Transform as indicated in the following diagram.

Figure 22-1 ***Time Frequency Transforms***



Where s(t) is the input signal in the time domain

S(f) is the frequency spectrum of the input signal

h(t) is the impulse response of the system

H(f) is the complex frequency response of the system

s'(t) is the output signal in the time domain

S'(f) is the frequency spectrum of the output signal

and the underbar indicates that the quantity is complex.

In terms of describing the system, both the impulse response h(t) and the complex frequency response H(f) provide complete system information and either can be calculated from the other using a Fourier Transform or an inverse Fourier Transform as follows:

$$\underline{H(f)} = F\,[h(t)]$$

$$h(t) = F^{-1}\,[\underline{H(f)}]$$

Frequency Domain Measurements

Frequency domain measurement techniques seek to determine the system frequency response through measurements of the input and output frequency spectra.

The output frequency spectrum is calculated by a multiplication, as follows;

$S'(f) = S(f) \bullet H(f)$

If one is concerned only with magnitudes and not phase, one can multiply both frequency spectra by their complex conjugates to obtain

$|S'^2(f)| = |S^2(f)| \Sigma |H^2(f)|$

If the goal of the measurement is to characterize the system by measuring the input and output frequency spectra, the result is

$|H^2(f)| = |S'^2(f)| / |S2(f)|$

and if we are working with log magnitudes of these quantities,

$\log|H^2(f)| = \log|S'^2(f)| / |S^2(f)| = \log|S'^2(f)| - \log |S^2(f)|$ (eq. 1)

For frequency domain measurements, the Larson Davis Model 3000+ provides dual channel analysis with cross-channel measurement capabilities using either FFT analysis for narrowband constant bandwidths or digital filters for 1/1 or 1/3 octave bandwidths.

Time Domain

Time domain measurement techniques seek to determine the system impulse response through measurements of the input and output signals. From this, the complex frequency response can be calculated as the inverse Fourier Transform of the impulse response.

In the time domain, the input-output relationship is expressed as

$$s'(t) = s(t) \bullet h(t) = \int h(t')s(t-t')dt'$$

For certain types of input signals, this can be simplified.

Impulse Excitation

A theoretically ideal impact, characterized as having a finite energy, with an amplitude approaching infinity and a duration approaching zero, can be represented by the Dirac delta function

$$s'(t) = \int h(t')\delta(t-t')dt' \approx h(t)$$

This means that a measurement of the time domain response of the system to excitation by an impact is an approximation to the system impulse response. The closer the impact resembles an ideal Dirac pulse, the better the approximation will be. The problem is that the requirement of short duration means that a very high amplitude is necessary in order to inject a significant amount of impulse energy into the system. In practice, this means that when using impact excitation it is difficult to achieve levels of response which are much greater than the ambient level. Another associated problem is that excessive impact amplitudes can exceed the amplitude linearity range of the system and produce distortion.

Stationary Noise Excitation

The main parameter for the description of stationary random noise is the autocorrelation function. In a system with statistical signals, the convolution of the autocorrelation function of the input with the system impulse response is referred to as the crosscorrelation function.

$$\Phi ss'(\tau) = \Phi ss(\tau) \bullet h(\tau) = \int_{-\infty}^{\infty} h(t')\Phi ss(\tau - t)dt'$$

If the input signal is such that its autocorrelation function approximates a Dirac delta function, then

$$\Phi ss' = \int h(t')\delta(t-t')dt' \approx h(\tau)$$

Using such an excitation signal, a measurement of the cross-correlation function will provide a direct approximation to the system impulse response. However, the restriction is now that the autocorrelation function of the input resemble a Dirac delta function, not the signal itself. The requirement that the signal have a short duration no longer applies.

There are a variety of signals whose autocorrelation resemble that of a Dirac delta function. Among these signals are a type called maximum length sequences (MLS) for which there exists a fast correlation algorithm called the Hadamard Transformation which is rather efficient for the computation of the impulse response function from the measurement of the system output based on the MLS input excitation.

The MLS signal is a periodic sequence with each sequence consisting of 2^N-1 impulses equally spaced in the time domain. The pulses have equal amplitudes, but may be either positive or negative. In practice the sequence is generated using a shift register with feedback such that the signs of the resulting pulses satisfy certain mathematical requirements for it to be a maximum length sequence. The power spectrum of an MLS signal looks like white noise and when used to drive a loudspeaker, it sounds like a periodically repeated white noise signal due to the periodicity of the MLS sequence.

Room Acoustics Applications

The traditional method for measuring the airborne sound transmission loss between rooms is to generate a stable broadband or bandlimited acoustic signal in one room (source room) and measure the sound pressure spectra in both the source room and the adjacent room (receiving room). Additional tests are performed in the receiving room to determine the background noise spectrum and the reverberation time.

According to equation eq.1, the difference between the sound pressure spectra measured in the source and receiving rooms, called the Noise Reduction, should define the performance of the wall structure separating the rooms.

Eq. 1 $\log |H^2(f)| = \log |S'^2(f)| - \log |S^2(f)| = L2 - L1$

where L1 and L2 are the 1/3 octave sound pressure spectra measured in the source and receiving rooms, respectively.

The standards governing the measurement of airborne sound transmission loss are formulated in terms of the Transmission Loss, which is proportional to the inverse of the term $\log|H^2(f)|$ in the above equation, and an additional term is required to deal with the steady loss of acoustic energy in the receiving room due to absorption at the room surfaces. The actual equation is

$$TL = L_1 - L_2 + 10\log\left(\frac{S}{A}\right)$$

where

S = the surface area of the test partition, in m^2

A = sound absorption of the receiving room, in metric Sabines

Thus, the TL is determined by the difference between two measured sound spectra, the Noise Reduction, plus the term representing the room absorption. In the majority of cases, this method is totally adequate to provide accurate results. However, one situation which can occur in practice is that the sound pressure level obtained in the receiving room is too close to the background noise level to obtain an accurate measurement. One example of this would be when testing walls or partitions having extremely high transmission losses where it is difficult to generate a strong enough signal in the source room to achieve a satisfactorily high level in the receiving room. Another would be the case where there is a relatively high background noise such as might occur when measuring the transmission loss between adjacent apartment units, offices or schoolrooms while there is noise being generated in the corridor or other adjoining rooms. Yet another is where there is a source of background noise within the receiving room itself, as would occur when there were people talking inside the rooms while the testing was taking place. The standards address this issue, providing correction factors to be used when the receiving room spectrum levels are within 10 dB or less of the background spectrum levels, but even this approach becomes invalid as the difference approaches 6 dB. The difficulty rests in part with the fact that frequency domain spectrum averaging of a noise signal can improve the accuracy of a measurement by converging towards the theoretical mean value, but it cannot remove the contribution of the noise to the measured signal.

As long as the background noise is random in character, the use of MLS excitation with measurement in the time domain can provide a solution to this problem. Because the MLS signal is deterministic in nature, synchronous averaging in the time domain can reduce the effect of random background noise by 3 dB each time the averaging time is doubled. In other words, by sampling at a rate synchronous with the period of the sequence, the components related to the test signal add in phase while those of the random noise, which are uncorrelated, will average towards zero. The resulting gain in the signal-to-noise ratio is equal to

$$\Delta_{ave} = 10 \log N \mathrm{dB}$$

where N = number of averages

The advantage of the MLS technique in room acoustics is to provide an accurate measurement of the impulse response in the presence of high background noise levels. In practice, the source room is excited by the MLS sequence and separate measurements are made in the source and receiving rooms in order to determine the impulse response of each. Having these, a straightforward spectrum analysis of these impulse responses will provide the source and receiving room sound power spectra required for the calculation of the Transmission Loss. Since the impulse response of the receiving room is essentially the response of the room to an actual acoustic impulse, such as the firing of a blank pistol or the breaking of a balloon, the traditional methods of reverberation time measurement can be applied to the impulse response function.

Performing MLS Measurements

The measurement of impulse response functions using the MLS technique requires a Model 3000+ equipped with the 2800-OPT 11 Digital Signal Generator and the 2900-OPT 81 MLS Module. The measurement is done from the MLS Menu using an MLS sequence of 65,535 points (2^{16} - 1). At the conclusion of the measurement, the impulse response function, consisting of 65,535 digital samples, is displayed and stored in a buffer. If desired, it can be stored to the active memory file if there is sufficient available memory (250 kilobytes required per channel per file), Upon exiting from the MLS Menu, by selecting to leave the MLS mode active, the data in this buffer is kept to be used as an input for subsequent frequency analysis. Thus, the impulse response can be "played" back through the analyzer like a digital tape loop.

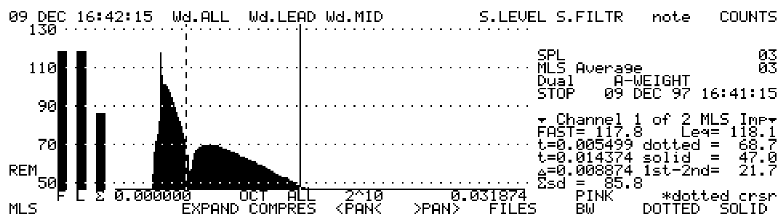
The MLS signal will be generated from the Signal Generator Output connector on the top of the instrument as denoted on the rear panel label. For room acoustic applications, this should be connected to the input of the power amplifier and speaker used to excite the source room or the space to be tested. The Larson Davis CBL061 cable with a mini phone plug and a BNC male connector may be useful.

As mentioned above, the usual application of the MLS technique involves frequency analyzing the measured impulse response. In order that the sample rate obtained when using the impulse response function as an input signal match the analysis, the user must establish what type of frequency analysis is to be used before the MLS measurement is performed; either 1/1, 1/3 octave or FFT. For the Model 3000+, it is assumed that whichever of these is active when accessing the MLS Menu is the one which will be performed following the measurement. Thus, the user should set the analyzer to either digital filters (1/1 or 1/3 octave) or FFT prior to accessing the MLS Menu. If upon exiting the MLS Menu the user changes to the other type of analysis, the MLS Mode will be de-activated and a corresponding message displayed on the upper right of the screen.

Note that the MLS Mode may be operated in either single or dual channel mode, determined by the number of channels active at the time the MLS Menu was accessed.

The MLS Menu, shown in Figure 22-2, is accessed from the Main Menu by pressing **SYSTEM, MLS [N]**.

Figure 22- 2 ***MLS Menu***



Selection of Coloring of the MLS sequence

When using 1/1 or 1/3 octave bandwidths, where the filter bandwidth decreases with frequency, the standard MLS sequence may produce insufficient energy at low frequencies. The user can select to “color” the MLS signal in such a manner that equal energy is provided in each octave or 1/3 octave band. To do this, the user presses **S.FILTR [F]** followed by one of the following keys:

Softkeys_	Softkey Functions
WHITE [A]	No coloration
PINK [B]	Pink Noise coloration

This selection will produce the display of either “WHITE” or “PINK” above the label of the hardkey **[N]**.

Selection of Frequency Bandwidth

The length of the MLS sequence in time is inversely proportional to the upper frequency of the analysis which is to be performed following the measurement. It also depends upon which type of frequency analysis is active. The upper frequency of the analysis is selected by pressing **BW [N]** and one of the following:

Softkeys	Softkey Functions	Length of Sequence, seconds
20 kHz [A]	20 kHz upper frequency	1.02 (1/1 or 1/3 octave) 1.275 (FFT)
10 kHz [B]	10 kHz upper frequency	2.04 (1/1 or 1/3 octave) 2.55 (FFT)
5 kHz [C]	5 kHz upper frequency	4.08 (1/1 or 1/3 octave) 5.1 (FFT)
2.5 kHz [D]	2.5 kHz upper frequency	8.16 (1/1 or 1/3 octave) 10.3 (FFT)

Selection of Number of Averages

The synchronous time averaging process reduces the effective background noise level by 3 dB each time the number of averages is doubled. Thus, the improvement in signal-to-noise ration is given by

$\Delta ave = 10 \bullet \log N$,

where N = number of averages

To select the number of averages, press **COUNTS [H]**, which produces the message

"Enter # of AVERAGES XXXX" on the upper right of the screen, with a flashing cursor to indicate that the keypad, and the horizontal arrow keys, are to be used to enter a value. Press **EXIT** after entering the desired number. This number will then be displayed on the right of the screen, 2nd line down, in the form “MLS Averages XX”. Length of measurement time is the price to be paid for increasing the signal-to-noise ratio of the measurement. The total measurement time equals the number of averages multiplied by the length of the sequence, given above. It is not unusual to use measurement times on the order of ten minutes or more.

Output Level Control

The relative output level is controlled by pressing **S.LEVEL [E]** and using the horizontal arrow keys to adjust the level between 0 and -40 as indicated by the message “*level - XX” on the lower right of the display. After adjusting the level, press **CURSOR** so that the horizontal arrow keys no longer control the output level adjustment.

Performing a Measurement using the MLS Technique

To begin a measurement, press **RUN/STOP** and note that the state of the analyzer indicated on the right of the screen, 4th line down, changes to RUN. When the number of averages completed, displayed on the 1st line, reaches the desired number of averages, 2nd line, the analyzer will stop and the state will change to STOP.

As the measurement proceeds, the level of the vertical bar above the symbol “F” indicates the measured level corresponding to a Fast RMS detector (1/8 second exponential averaging time). This level should be as high as possible on the screen without an overload indication. If necessary, use the vertical arrow keys to change the gain in order to obtain a good signal level. The average will be reset and the measurement started again each time the gain setting is changed.

Display of the Impulse Response

At the conclusion of the measurement period, the envelope of the impulse response is displayed on the screen using a log amplitude scale. The default display scales the x-axis such that the x-axis represents the total time of the MLS sequence, as shown in Figure 22-2. The time domain display of the impulse response cannot be presented, since a log scale is used which does not permit the display of a curve passing through zero. Due to the limited resolution of the screen, only 256 data points are presented along the x-axis. Thus, data corresponding to 256 time points are represented by each data point on the screen, which is taken to be equal to the maximum level found among the 256 time points.

It may not be possible to observe the impulse response curve if it is of very short duration relative to the sequence length, since it might be compressed into a single vertical line or two near the origin of the axis. Repeated presses of the key **EXPAND [I]** will expand the x-axis scale by multiples of two. The key **COMPRES [J]** can be similarly used to compress the scale after it has been expanded. While in an expanded state, the keys **<PAN< [K]** and **>PAN> [L]** can be used to pan the display left or right, respectively, since only a portion of the total record can be seen on the screen at one time.

As the scale is expanded, the number of time points represented by each data point on the screen is decreased accordingly; expanded once, each data point represents 128 time points; expanded twice, each data point represents 64 time points, etc. Regardless of the number of time points represented, the level of each data point is the maximum found among the time points represented.

On left of the screen, the level of the vertical bar with the “L” below it represents the total energy contained in the complete sequence of points. The adjacent vertical bar with the “Σ” below it represents the energy contained in the portion of the sequence between the two cursors, dotted and solid. This permits the user to determine the relative percent of energy in different portions of the impulse response. For example, when there is a reflection of sound energy from a nearby surface, this will appear as a lobe of higher energy later in the time sequence. Using the cursors, the distribution of energy between the direct sound field and the reflected sound field can be determined. Further in this chapter, it is explained how the impulse response curve can be edited in order to utilize only portions of the impulse response for playback during analysis.

Exiting to the MLS Mode ON

To exit from the MLS Menu in a state which permits the impulse response to be used as a time domain input for frequency analysis, press **EXIT, MLS ON [A]**. As long as the analysis type (digital filters or FFT) are not changed, the MLS Mode will remain on and the time series will be repeatedly played as an input signal much as a tape loop. Changing the analysis type will de-activate the MLS ON mode as indicated by a brief message on the upper right of the screen.

Exiting to the MLS Mode OFF

To exit from the MLS Menu and have the analysis function operate in the standard manner, using input signals taken from the channel 1 and 2 input connectors, press **EXIT, MLS OFF [B]**.

Windowing of the Impulse Response Signal

There are situations where the user may wish to edit the impulse response function before using it as an input signal in the MLS ON mode of frequency analysis. The Model 3000+ provides two windowing functions for this purpose. The windowing is done using the two cursors, dotted and solid. Pressing **Wd.LEAD [B]** implements a tapered leading edge window shaped like the first quarter cycle of a cosine function between the cursors. Pressing **Wd.MID [C]** implements a mid-band window resembling a squared cosine between the two cursors. With a window active, the cursors may be moved left or right to modify the position and width of the window relative to the impulse response function. Figures 3, 4 and 5 illustrate the effect of these windows on the impulse response signal. Pressing **Wd.ALL [A]** removes the effect of the two windows and restores the originally measured function.

Figure 22-3 ***Unmodified Impulse Response Function***

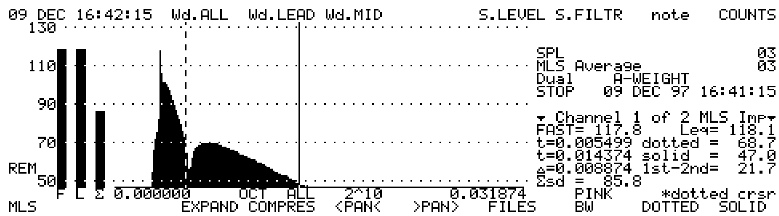


Figure 22-4 ***Effect of Leading Edge Window on Impulse Response***

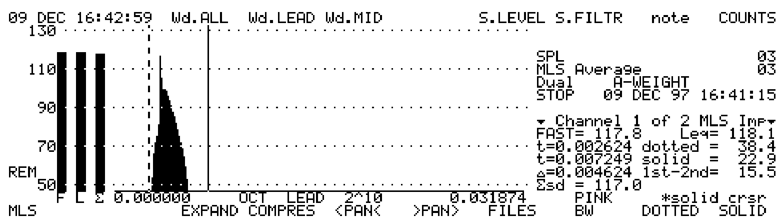
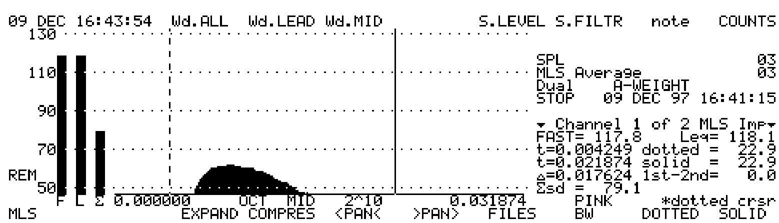


Figure 22-5 ***Effect of Mid-band Window on Impulse Response***



The vertical bar above the symbol "L" indicates the relative energy contained in the complete impulse response while the level above the symbol "Σ" indicates the relative energy contained in the portion of the signal between the two cursors.

Evaluation of Airborne Sound Transmission Loss from the Impulse Response Function

It is recommended that the user become familiar with the standard procedure for the measurement of airborne sound transmission loss as described in Chapter 21, Room Acoustics Measurements, before attempting to utilize the MLS procedure for performing these measurements. In the standard procedure, the measured parameters which must be input using the Rooms Menu are the Source Room spectrum, the Receiving Room spectrum, the Receiving Room background spectrum and the reverberation time in 1/3 octave bands (RT60). We will see in the following that the impulse responses measured in the source and receiving rooms can be used to determine the source and receiving room spectra, and the RT60 spectrum can be determined from the impulse response of the receiving room. The receiving room background spectrum must be measured in the standard manner, performing a 1/3 octave spectrum analysis of the signal in the receiving room without acoustic excitation of the source room.

Determining Source and Receiving Room spectra: In order to determine the source room and receiving room spectra, we utilize the MLS measurement to obtain the impulse response of both the source room and the receiving room. By using two microphones, one in the source room and one in the receiving room, the impulse responses of the two rooms can be measured simultaneously using the instrument in the dual channel mode. If that is not convenient, perhaps due to a problem running cables, each of these measurements can be made separately.

After exiting from the MLS Menu and leaving the MLS function ON, the impulse response signal can be frequency analyzed in 1/3 octaves in order to determine the corresponding spectra for the source and receiver room. This is done by performing a spectrum analysis using a linear average over a time greater than the duration of the impulse response. It is important that the same averaging time is used for the spectrum analysis of both the source room and receiving room spectra. For this reason, it is preferable that the source and receiving room impulse responses be measured simultaneously using both channels, since these measurements will be performed at the same time using the same averaging time selection. Otherwise, just be certain that the same parameters are used for the two spectrum measurements. Since the impulse response is a function of time, performing the spectrum analysis using different values of averaging time will produce different spectrum shapes. However, in the routine which calculates the airborne sound transmission loss the receiving room spectrum is subtracted from the source room spectra, so these effects are cancelled out as long as the two spectra have been integrated over the same time periods. Also, the use of the difference between the spectra cancels the effects of the sound reproduction system (amplifier and speaker) on the impulse responses measured individually in the two rooms.

Determining the RT60 decay time values: The impulse response measured in the receiving room is the time response which would be obtained in the receiving room due to excitation by a theoretically ideal acoustic impulse. Determining the reverberation time from this is procedurally identical to the determination of the decay time based on the response of the room to an impulsive excitation such as a starter pistol or a bursting balloon, which can be done by the methods described in Chapter 21, Room Acoustics Measurements. One selects to perform 1/3 octave spectrum analysis using Linear Repeat Averaging, with an averaging time much less than the anticipated reverberation time, and then performs a byTime autostore measurement using a delta time for the storage interval which is equal to the averaging time of the analysis. Following this, the vsTime display format is used as the basis for the determination of the decay time as described in Chapter 21. The main difference is that the impulse response is played back repeatedly, so there could be several response and decay patterns within the same vsTime record from which the decay parameters can be obtained.

Determination of the Airborne Sound Transmission Loss: Once the Source and Receiving Room Spectra and the RT60 data have been determined from the impulse responses as described above, the calculation of the airborne sound transmission loss is performed from the Rooms Menu exactly as described in Chapter 21.

CHAPTER

23 Classification (Class) Lines (Optional)

General Explanation of the Concept

The class lines function of the analyzer is a graphical technique for classifying a spectrum (or spectra) in terms of its interaction with a family (or families) of user–defined curves.

In this section we describe the use of class lines as applied to spectral data displayed on the analyzer in the amplitude versus frequency format. The class lines function can also be employed with spectra/order data displayed in the multi-line vsRPM/Speed format. This is described in a later section of this chapter.

A simple example of the use of class lines applied to frequency spectra is a quality control application where a sound or vibration spectrum measured for a machine under test is compared to a curve in the frequency domain to determine whether or not the machine is acceptable or not (e.g. Pass/Fail decision).

In Figure 23-1, the spectrum is below the curve at all frequencies, indicating an acceptable unit (Pass), while in Figure 23-2 the spectrum level at 1.6 kHz exceeds the curve, indicating an unacceptable unit (Fail) based on that criterion, even though the spectrum levels are below those of the spectrum shown in Figure 23-1 at most frequencies.

Figure 23-1 ***Pass Example***

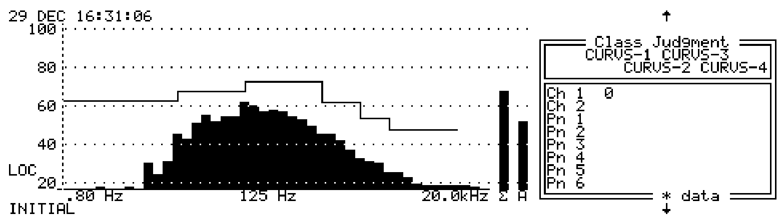
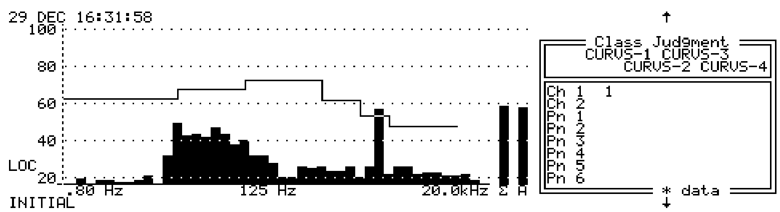
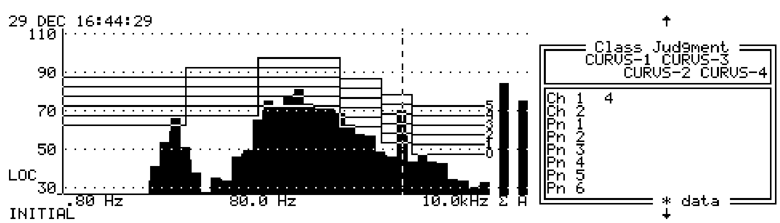


Figure 23-2 ***Fail Example***



A more advanced approach is indicated in Figure 23-3, where a family of parallel curves are used for the comparison with the test spectrum.

Figure 23-3 ***Classification Example***



The classification of the spectrum in this case could be based on the frequency for which the level has exceeded the maximum number of curves of this family. In this case there are six curves, labeled 0-5. Although the level has exceeded one curve at 10 Hz and at 80 Hz, and two curves at 125, 160 and 400 Hz, the maximum incursion of the spectrum into the family of curves is at 1.6 kHz where the spectrum level crosses four curves. The classification table on the right of the screen indicates that for the spectrum measured in channel 1 of the analyzer, the classification is based on a family of classification curves named “CURV–1”. The classification value of “4” reported in the table indicates that this spectrum has, at one frequency, crossed four of the lines of that family of classification curves, and that at no other frequency did the spectrum cross a larger number of curves (five or more). Looking back at Figure 23-1 and Figure 23-2, we see that the classification table also indicates Pass or Fail as 0 (Pass, no crossing of curve) or 1 (Fail, curve has been crossed).

Further examination of the classification table indicates the possibility of comparing the test spectrum with four different families of curves, denoted in this example as CURV–1, CURV–2, CURV–3 AND CURV–4, since there are four columns available. In addition, the existence of two rows corresponding to Ch1 and Ch 2 implies that the spectra measured for channels 1 and 2 can be compared independently against these four families of curves. Of course one can only display one of the two spectra at one time, along with the appropriate family of classification curves.

When working with data in the vsRPM/Speed format the user can define as many as 32 pens, each representing a specific channel number and frequency band or order number, as described in Chapter 17. When performing the class lines function the additional rows appearing in the first column below the Ch 1 and Ch 2 rows, denoted “Pn N” (N = 1,2,3,...), permit the class lines to be applied to each of the curves corresponding to each pen number as well as to the two channels of data displayed on the analyzer itself.

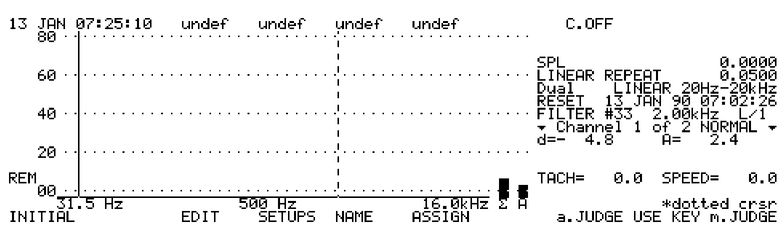
The ability to classify a test item according to different classification criteria in a single operation is of great practical importance. Suppose, for example, the test object is a motor which is used in a variety of different machines and the major concern of the machine manufacturer is that the radiated noise be within acceptable limits. In many cases the noise radiation by the machine associated with the motor will depend predominantly upon the vibration spectrum of the motor, the transfer function of the vibration between the motor mount and the sound radiating surfaces of the machine, and the radiation efficiency of the machine surface in converting surface vibration into acoustically radiated energy. Thus the actual noise radiated by a particular machine will depend on both the parameters of the motor and those of the machine itself.

The design of one machine may make it particularly susceptible to motor vibrations at a particular frequency, yet rather insensitive to those at other frequencies. Stated another way, when the goal of the test program is to minimize the noise radiated by a machine utilizing that motor, then the classification of a motor in terms of its vibration spectrum must in some way reflect the vibration transmission/noise radiation characteristics of the machine itself. Simply said, with this classification technique using four different classification curve families, one for each of four different machine designs, the motors could be sorted or classified in terms of acceptability for use in any of these four different machine designs by a single test. The fact that two channels can be measured and classified at once permit the user to classify the sound or vibration of one motor at two positions, or to test two motors on different test lines at the same time.

Accessing the Class Lines

Class lines are only applicable to spectra measured in the Standard Analysis Mode (STAND 1 or STAND 2), so the analyzer must first be configured to one of these. Both digital filters and FFT analysis may be used. Access the Class Lines Menu from the System Menu by pressing **CLASS [H]**. If the Class Lines function is already ON, the Class Lines Menu shown in Figure 23-4 will be displayed. If the Class Lines function is switched OFF, press **C.ON [F]** to turn it ON. The default bootup has the Class Lines function switched OFF.

Figure 23-4 ***Class Lines Menu***



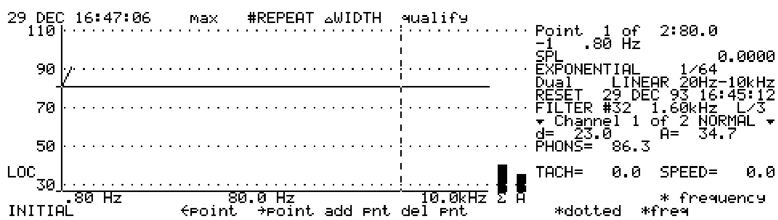
Labeling the Class Lines

To identify a set or family of class lines, the user can assign a label having up to seven characters to each of the softkeys **[A]**, **[B]**, **[C]** and **[D]**. When delivered, each of these softkeys will be labeled “undef”. Create a label by pressing **NAME [K]** which will produce the message “Push Class to Name” on the upper right of the screen. Press the softkey for which label is to be defined. The message “Enter class name:” with a flashing cursor beneath the first character of the field prompts the user to type in a label name using the alphanumeric keypad and press **EXIT**.

Creating a Single Class Line

Press **EDIT [I]** and reply to the message “Select Class to edit” on the upper right of the screen by pressing one of the softkeys **[A]**, **[B]**, **[C]** or **[D]**. The screen will look like Figure 23-5, unless a line or family of lines have been created previously.

Figure 23- 5 ***Default Editing Menu***



Since we are describing the creation of a single class line, if there are a family of curves displayed upon accessing the Edit Menu, press **#REPEAT [B]**, use the numeric keypad to type “00” into the field on the upper right of the screen and press **EXIT**, which will collapse the family of lines to a single base line.

The generation of a single class line is essentially the creation of a connect–the–dots sequence on the screen using up to a maximum of twenty points. There are always at least two points active. If a class line has not already been created, the display will be of the default setup, as shown in Figure 23-5.

The coordinates of a selected point are indicated graphically on the screen by the intersection of a horizontal and a vertical line and numerically on the upper right of the screen by the message

“point X of Y: nn.n

(ANSI filter number) (center frequency)

when using digital filters, or the message

“point X of Y: nn.n

(center frequency)

when using FFT filtering. X represents the number of the selected point, Y represents the total number of points presently defined for the line, nn.n is the amplitude coordinate of point X and the center frequency is the frequency coordinate of point X.

The two default points will be located horizontally at the two lowest frequency band center frequencies, with amplitudes of 80.0 and 90.0 dB for points 1 and 2, respectively.

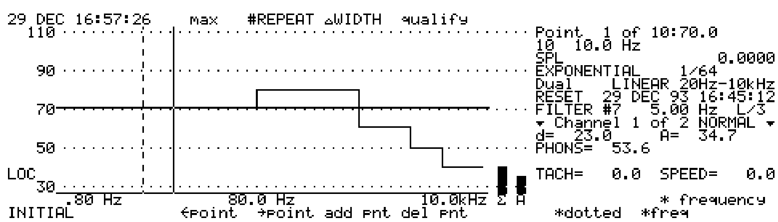
The key ←**point [I]** and →**point [J]** are used to move along the line from point to point. Press these two keys and notice how the horizontal and vertical lines move to “cross–hair” the selected point coordinates on the screen and how the amplitude and frequency coordinates are displayed digitally on the upper right of the screen. In this exercise we want to show how to create a class line from the default state. To return the status of the class lines to the default state, press the following key sequence from the Class Lines Menu: **SETUP [J]**, **DEFAULT [H]**, **YES [A]**, **EXIT** which will reset all the labels for **[A]**, **[B]**, **[C]** and **[D]** to “default” and replace all the pre–existing curves to two point curves having the coordinates described above.

For the purpose of this exercise, we will create a class line named “TEST”. Begin by naming this class line by pressing the key sequence **NAME [K]**, **[A]**, **SHIFT**, **CLEAR**, **T**, **E**, **S**, **T**, **EXIT**.

To create the class line, press the key sequence **EDIT [I]**, **TEST [A]**. Use the keys **←point [I]** and **→point [J]** to move the horizontal and vertical cross–hair lines between points one and two, noting their coordinates as displayed on the upper right of the screen. We will begin the creation by accessing point 1. Press the up and down vertical arrow hardkeys on the lower right of the front panel and notice that point 1 will be moved up or down in steps of 0.1 dB. If the **SHIFT** hardkey is pressed at the same time, point 1 will move up or down in steps of 1.0 dB. Pressing the left or right horizontal arrow keys will shift the location of point 1 left or right along the frequency axis in steps of one filter bandwidth. Pressing the **SHIFT** hardkey at the same time will result in a much larger step size.

In this exercise, we are going to create the line shown in Figure 23-6.

Figure 23- 6 ***Single Class LIne Example***



Begin by accessing point 1 and using the vertical and horizontal arrow keys to move it to the coordinates (70 dB, 10 Hz). Note that as this point is moved horizontally to the right, its designation is changed to point 2 because it is now to the right of the other point, now designated as point 1. Next, access the other point, now point 1, and move it to the coordinates (70 dB, 63 Hz). Fix the location of this point by pressing **add pnt [K]**, and notice that the coordinates listed on the upper right are for point 3. Use the cursors to move point 3 to the coordinates (80 dB, 63 Hz) and press **add pnt [K]**. Continue the sequence as follows:

move point 4 to (80 dB, 630 Hz), **add pn t[K]**

move point 5 to (60 dB, 630 Hz), **add pn t[K]**

move point 6 to (60 dB, 2 kHz), **add pnt [K]**

move point 7 to (50 dB, 2 kHz), **add pnt [K]**

move point 8 to (50 dB, 4 kHz), **add pnt [K]**

move point 9 to (40 dB, 4 kHz), **add pnt [K]**

move point 10 to (40 dB, 10 kHz), **add p n t[K]**

The class line is now completed. Use the ←**point [I]** and →**point [J]** keys to move back and forth through the sequence of points. If you press **EXIT**, the class line will disappear because we are out of the editing mode and have not turned them ON. If you do exit, you can return to editing this line by pressing **EDIT [I]**, **TEST [A]**.

As points are added, deleted and moved, they are always numbered sequentially across the screen from left to right. In the editing mode, when an existing point is accessed, it can still be moved vertically and horizontally. For small movements, you will see that is simply deforms the shape of the line, as would be expected. However, if it is moved horizontally sufficiently far that it passes one of the other existing points, either to the left or the right, its point number is shifted accordingly and the point which had been passed now assumes the point number previously associated with the point being moved and the point which was moved will have a point number one lower or higher depending upon whether it has moved to the left or right of the existing point. When an existing point is deleted, all the numbers of the points in sequence to the right of that point are decreased by one, and when a point is added within a sequence of existing points all the numbers of the points in sequence to the right will be increased by one.

The best way to become familiar with the creation of a single class line is to experiment. Until one becomes quite familiar with the method, the recommended technique is to sketch the desired pattern on paper, with coordinates, and generate the line using a simple series of points created sequentially from left to right.

Assigning Max or Min Mode

In most noise and vibration applications of this technique, the desirable result of a test is that noise or vibration spectrum levels be as low as possible. In terms of the class lines, this might correspond to the desire that the spectrum levels remain below a single classification line, or that they cross as few lines in the upward direction as possible. We refer to this as the Max mode of operation, and when a family of class lines is created, they are numbered sequentially in the vertically upward direction beginning with 0 (see section below). The higher the number associated with the classification of a spectrum using the Max mode, the more “severe” the rating because it is associated with increasingly higher spectrum levels (or at least at one frequency).

The class lines also support a Min mode. In terms of a single class line, a classification of 0 indicates that the spectral levels are all above the line and a classification of 1 denotes that in at least one frequency band the level is below the class line. A family of lines created from the Min mode are numbered sequentially in the downward vertical direction, starting with 0. The classification of a spectrum indicates the degree to which the spectrum levels have penetrated downwards across the family of lines, and how many lines have been crossed in that direction.

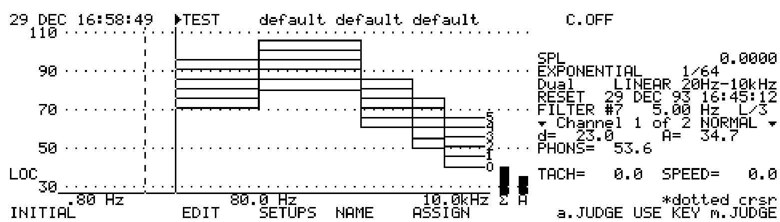
In the editing mode, repeated presses of the softkey **[A]** will toggle the mode between Max and Min, as indicated by the label displayed above that key.

Creating Multiple Class Lines

A family of parallel class lines can be generated from a single class line by pressing **#REPEAT [B]**, and in response to the message “Number of repeats nn” on the upper right of the screen use the numeric keypad to type in a value and press **EXIT**. To set the spacing between the lines, press ∆**WIDTH [C]** and in response to the message “Repeat Delta nn dB”, use the numeric keypad to type in the number of dB spacing to used between lines and press **EXIT**.

When in the Max mode, the class lines will be numbered sequentially in the upward vertical direction as shown in Figure 23-7.

Figure 23-7 ***Multiple Class Lines Example***



In the Min mode, they will be numbered sequentially in the downward vertical direction.

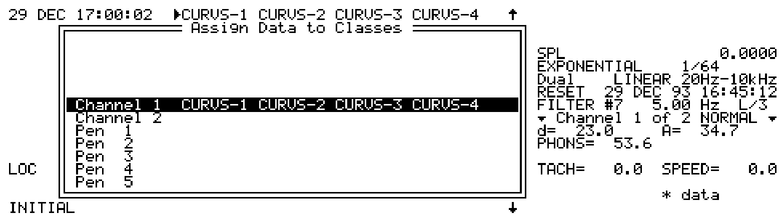
Turning On a Class Line Family

Repeatedly pressing one of the class line softkeys **[A]**, **[B]**, **[C]** or **[D]** will toggle the status of that line between ON and OFF. ON status is indicated by an arrowhead symbol to the left of the softkey label. Each may be set to ON or OFF independently. In the ON state, class lines previously defined for each family will be displayed when not in the editing mode. When more than one family is in the ON state, the class lines for all the ON families will be displayed simultaneously. A family of class lines cannot be assigned (see below) or used for comparison to a spectrum unless it is in the ON state.

Assigning Class Lines to an Input Channel

At any given time there could be as many as four families of class lines defined and in the ON state. The assignment operation establishes which family, or families, of class lines are to be used for comparison with a spectrum measured for a particular channel. Access the Assignment Menu by pressing **ASSIGN [L]**, producing a display such as shown in Figure 23-8.

Figure 23- 8 ***Assignment Menu***



Utilize the softkeys ↑ **[E]** and ↓ **[M]** to align the highlighted line with either Channel 1 or Channel 2, this being the input channel to which the class line families are to be assigned. Repeatedly pressing any one of the softkeys named for a particular family of class lines (in Figure 23-8 these are named **CURVS–1 [A]**, **CURVS–2[B]**, **CURVS–3 [C]** and **CURVS–4 [D]**) will cause the name to alternately appear and disappear along the row associated with that channel number. All the names appearing along the row at the time of exiting from the Assignment Menu are assigned to that channel. Once these are assigned, whenever a judgement (or comparison) is made between a spectrum measured for that channel and the class lines, the comparison will be made simultaneously using all the class line families whose names appear in the row associated with that channel. In this manner different combinations of the four possible class line families may be assigned to the two input channels.

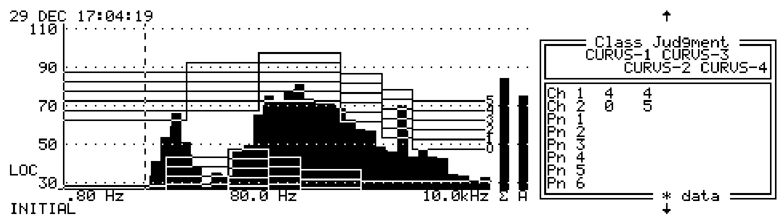
Automatic Judgement of Spectra (all channels) Using a Softkey

In order to perform a judgement of spectra based on class lines using a softkey, it is first necessary to press the softkey corresponding to the hardkey **[O]** until its label is **USE KEY [O]**.

Since measurements cannot be made while in the Class Lines Menu, exit to either the System Menu or the Main Menu using the **EXIT** hardkey. From there, either perform a new measurement or recall and keep a previously stored measurement, and access the Class Lines Menu by pressing

CLASS [H] (if in the System Menu) or the sequence **SYSTEM**, **CLASS [H]** (if in the Main Menu). Perform the judgement by pressing **a. JUDGE [N]**, which will produce a display similar to that shown in Figure 23-9.

Figure 23-9 ***Automatic Judgement Example***



In the table displayed on the right of the screen, for each channel there will be a number corresponding to each of the assigned class line families indicating the classification of that spectrum with respect to that particular family of class lines. In this example note that the family CURVS-1 is in the Max mode while the family CURVS-2 is in the Min mode. An example of this combination of modes might be where the most desireable result is that the spectrum lie completely between the two families, and the larger the variation, in either the upward or downward direction, the less acceptable the result.

Even though only one spectrum can be displayed at a time, channel 1 or channel 2, the judgement is made for both channels at the same time. If, just previous to the judgement, the highlighted line in the Assignment Menu had been aligned with channel 2, it is possible that the line for channel 1 will not be seen on the table. In that case use the ↑ **[G]** softkey to bring that line back down into the table.

Manual Judgement of a Displayed Spectrum using a Softkey

As above, press the softkey corresponding to the hardkey **[O]** until its label is **USE KEY [O]**. Return to the System Menu or the Main Menu and either make a new measurement or recall and keep a previously stored measurement. Display the spectrum which is to be judged (channel 1 or channel 2), then access the Class Lines Menu and press **m.**

JUDGE [P] which will produce the message "Select class to judge" on the upper right of the screen.

Upon pressing a softkey representing one of the named class line families, a selection of softkeys will be presented at the top of the screen, each one representing one line of the family selected. There will only be as many lines represented as were defined for that family originally. Press any one of these, noting that only that particular line of that family is displayed, permitting visual comparison of the spectrum with that line. However, at this point, the set of softkeys representing different lines of that family remains along the top of the screen, enabling the user to continue to select any particular line of that family for visual comparison against the displayed spectrum.

NOTE: At this particular point in the sequence we are describing, the entire set of lines of the families not yet selected do appear, along with the single line of the selected family.

Upon exiting from this Menu the message "Select class to judge" will again appear on the upper right of the screen. However, if, previous to exiting, a single line of one family had been displayed as a result of a previous selection, then only this single line of that family will continue to be displayed. At this point the user can select another family and proceed, as described above, to display only one line of that family. The result will be that one line only, for each of these two different families, are displayed along with the spectrum, in addition to the entire set of lines for the families not yet selected. Continuing until one line from each of the families has been selected, the user can visually compare the spectrum with any single line from any or all of the four possible families.

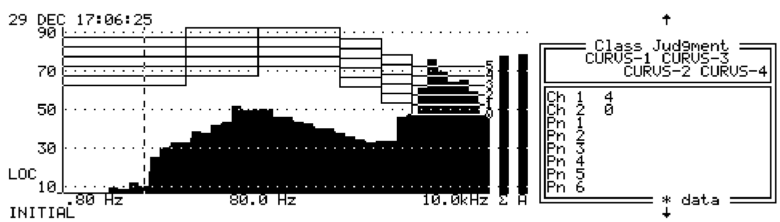
Automatic Judgement Based on Stop State of Analyzer

For this particular mode of operation, press the softkey corresponding to the hardkey **[O]** until its label is **AT STOP [O]**. In this mode it is not necessary to access the Class Line Menu to perform a judgement, which makes it ideal for on–line applications. Exit to either the System Menu or the Main Menu and press **RUN/STOP** to begin a measurement. As soon as the measurement is stopped, either by pressing the **RUN/STOP** key a second time or waiting until a linear single average or a count single average is completed, the judgement is performed and the judgement table automatically displayed on the right of the screen. Press **EXIT** to return to the System or Main Menu, and **RUN/STOP** to initiate a new measurement preparatory to performing another judgement as described above.

Classifications Requiring Line Crossings at Multiple Frequencies

In the preceding descriptions, the classification of a spectrum was based upon the farthest penetration of any one spectrum level into a family of class lines, either in the upward direction for Max mode or in the downward direction for Min mode. The numerical classification was simply the number of lines crossed at the frequency representing the furthest penetration. Another classification scheme could involve the requirement that the line of maximum penetration be crossed at multiple frequencies. This is done from the Editing Menu by pressing **qualify [D]** and in response to the message "Points to qualify   nn" on the upper right of the screen, typing in a number using the numeric keypad and pressing **EXIT**. In the example shown in Figure 23-10, a value of 03 has been entered as the qualification value for the family CURV–1.

Figure 23-10 ***Judgement Example using qualification = 3***

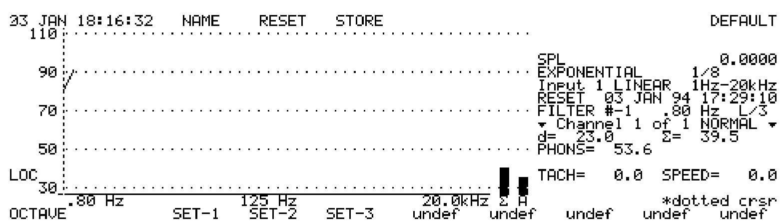


Upon performing a judgement of a spectrum, the numerical classification of the spectrum will represent the maximum number of lines in that family which have been crossed at least three times. As can be seen in the table, the classification in this example is 4 because four lines have been crossed at a minimum of three frequencies and the fifth line has only been crossed at two frequencies. Had the qualification value been zero this spectrum would have been classified as 6 because all six lines have been crossed at one frequency.

Storage of Class Lines to Setup Menu Softkeys

Each set of four user–defined and named class lines can be stored to non–volatile memory from the Class Lines Setup Menu, shown in Figure 23-11, which is accessed from the Class Lines Menu by pressing **SETUPS [J]**.

Figure 23-1 1 ***Class Lines Setup Menu***



A different set of class lines can be stored under each of the eight softkeys at the bottom of the screen. When first delivered, each of these softkeys will have the label “undef” to indicate that no user-defined label has yet been defined for them. Prior to storing a set of class lines to a particular softkey it is best to assign a meaningful name to its label in order to be able to remember in the future which set of class lines has been stored there. This is done by pressing **NAME [A]**, responding to the message “Push Setup to Name” on the upper right of the screen by pressing the softkey whose label is to be defined, responding to the subsequent message “Enter setup name” by typing in a name using the alphanumeric keypad, and pressing **EXIT**. This name will now appear as the softkey label above the hardkey it represents. This procedure simply labels a softkey so the user may label as many of these eight softkeys as desired, either all at once or as convenient.

To store the presently active set of class lines to any of the eight softkeys along the bottom of the screen press **STORE [C]**, and in response to the message “Store in this setup” on the upper right of the screen simply press the softkey to which that set of class lines is to be stored.

Recalling a Set of Class Lines from Setup Menu Softkeys

To recall a set of class lines stored under a softkey in the Class Lines Setup Menu simply press that key. The message “Overwrite current setup?”, on the upper right of the screen, warns that this procedure will cause the four class lines presently active to be replaced by the set which is being recalled and, unless the presently active set has already been stored, it will be lost. Press **YES [A]** to continue with the recall or **NO [C]** to abort the recall operation. After completing the recall operation, and exiting from the Class Line Setup Menu to the Class Lines Menu, the names of the four class line families which have been recalled will appear as softkey labels at the top of the screen.

Storing Class Lines Stored under Setup Menu Softkeys to Non–volatile Memory

As explained above, a set of four class line families can be stored to each of the eight softkeys in the Class Lines Setup Menu, which represents a total of thirty–two class line families. All of these can, in turn, be stored to the non–volatile memory as a Class setup record and, if desired, stored to floppy disk as well. Thus the user can develop a library of different class lines, each stored in either non–volatile memory or on disk. Upon recalling a single Class setup record, all thirty–two families are recalled in groups of four, one per softkey. From the Class Lines Setup Menu press **STORE** and note the message on the upper right of the screen “STORE – Class setup n” indicating that all class lines stored under all eight of the softkeys have been stored to the nth record of type Class setup”.

Recalling Class Lines from Non–Volatile Memory to the Class Lines Setup Softkeys

To perform this operation press **RECALL**. The message "Overwrite all setup?" on the upper right of the screen warns that the setups being recalled will replace those presently active under all softkeys and that, unless they have already been stored as a Class setup, they will be lost. Press **YES [A]** to continue with the recall or **NO [C]** to abort the recall. When **YES [A]** has been pressed the message "RECALL – Class setup n" will appear on the upper right of the screen to indicate that the nth record of type Class setup has been recalled. At the same time the labels of the eight softkeys along the bottom of the screen will indicate the names used to store each set of four class line families. At this point, before exiting, the left and right horizontal arrow keys can be used to page backwards and forwards through all the stored records of type Class setups. As this is done the value of the record number in the message on the upper right and the softkey labels will change accordingly. When the desired Class setup record has been recalled press **EXIT** to cease the recall process and return to the Class Lines Setup Menu.

Turning Off the Class Lines Function

To turn off the Class Line function, from the Class Lines Menu, press **C.OFF [F]**.

CHAPTER

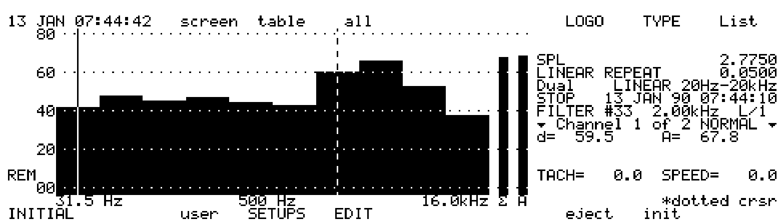
24 Printing Data Screen Displays and Tables

The Model 3000+ can print any displayed data directly to a Hewlett Packard compatible laser printer or an Epson Compatible printer equipped with Graphics capability via the Centronics parallel interface. A tabular output of the data displayed on the screen can also be obtained.

Accessing the Print Menu

The Print Menu, shown in Figure 24-1, is accessed by pressing the hardkey **PRINT**.

Figure 24- 1 *Print Menu*



Initializing the Printer

If the printer is connected and turned on when the 3000+ is booted up, the printer initialization is performed as part of the boot up procedure if RS232 communications were selected before the 3000+ was turned off. If the parallel port was selected before the instrument was turned off it will boot up with this configuration. In cases where the parallel port was selected the initialization will not be performed at bootup. Press **init [O]** to initialize the printer.

Creating the Logo

A user-defined logo or heading is printed at the top of each printout from the Model 3000+. To create the logo, press **LOGO [F]** and in response to the message “Logo:” on the upper right of the screen type in the desired logo using the alphanumeric keypad in the same manner as used to create notes for annotating data records, and press **EXIT**.

Selecting Printer Type

Press **TYPE [G]** to select the printer type which is to be used for the printout. Use the keys ↑**[B]** and ↓ **[J]** to scroll the printer listing vertically until the desired printer is highlighted. Many printers use an interrupt communication with the printer which permits the analyzer to continue drawing while the printer is printing, which we refer to as **FAST** mode. Pressing the key **[L]** will toggle the softkey label between **FAST** and **COMPAT.** Set this to **FAST** when using a printer which supports the interrupt communication and to **COMPAT.** for those which do not. When in doubt, try **FAST** and if this does not work select **COMPAT.** When the printer has been selected as described above, press **EXIT** to return to the Print Menu.

Printing the LCD Screen Display

To obtain a printout of the display presently on the LCD screen, press **screen [A]**. Because three screen display printouts will fit onto a single sheet, the actual printout will not occur until the third of a sequence of display printouts is initiated. To obtain a screen display printout on a single sheet press **eject [N]** following **screen [A]**.

Printing a Data Table

To obtain a printout in tabular form of the data which is being displayed on the LCD screen, press **table [B]**. In the tabular printout, the measured data values will be printed under the column labeled “RMS-dB”. When Digital Display Weighting has been selected to be other than No Weighting (NO WGT), the displayed values will be different from the measured values by the amount of the selected weighting function. The display weighted values are printed under the column labeled “DISP-dB”.

Printing LCD Screen Display and Data Table

To obtain a printout of both the display presently on the LCD screen and a data table representing the data being displayed, press **all [C]**.

Print to Screen (List) Function

To print a Data Table to the screen instead of a printer, press **List [H]**, which will provide the user with softkeys to select the desired format of the listing: **Left [A]**, **Right [B]**, and **Wrapped [C]**.

Aborting a Printout

To abort a printout in progress of LCD screen data, press **abort [P]**.

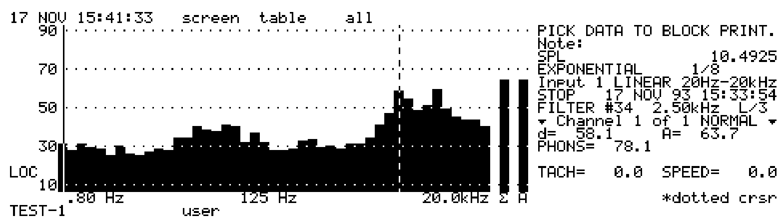
Ejecting a Sheet of Paper, or Making a Form Feed

To eject a single sheet of paper from the printer or to create a form feed, press **eject [N]**.

Block Printing of Stored Data Records

The block print function permits the Model 3000+ to recall and print a sequence of stored data records of the same data type. To do this, first recall a record of the type which is to be printed. The softkey **BLOCK [D]** will not appear until after a stored record has been recalled. Then, while still in the Recall Menu, press **PRINT**, which will display the complete Print Menu as shown in Figure on page -1, including the softkey **BLOCK [D]**. Pressing **BLOCK [D]** will then display the Block Print Menu shown in Figure 24-2, and the message “Pick data to block print” on the upper right of the screen.

Figure 24-2 ***Block Print Menu***



Select among the options by pressing one of the following:

screen [A], table [B] or **all [C]**.

The message “RECORD # XXXX - # YYYY” on the upper right of the screen will indicate the range of record numbers of that type which are presently stored in the active file. Use the keypad to edit these numbers such that they represent the range of sequential records which are to be printed, and press **EXIT**. To abort a printout in progress, press **abort [P]**.

Custom Printouts

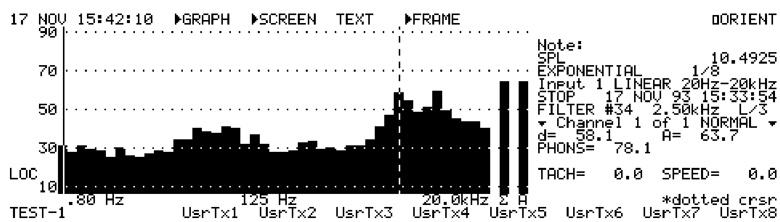
When using either laser or Epson compatible printers, you can generate and store up to eight custom printout formats capable of presenting the displayed data block using a scalable graph size, in either portrait or landscape orientation.

Along with the measured data, the custom printout can include most of the measurement setup parameters as text strings and eight user-defined text strings whose characters, location, size and location (vertical/horizontal) are under your control.

Accessing the Custom Printout Module

From the Print Menu, press **EDIT [K]** to access the Edit Menu, shown in Figure 24-3.

Figure 24- 3 ***Edit Menu (custom printouts)***



Scaling of the Custom Printout

The work space available for the custom printout depends upon the orientation selected for the graphic. In the portrait orientation the available height is 260 mm and the available width is 200 mm. In the landscape orientation, the available height is 200 mm and the available width is 260 mm. Figure 24-4 and Figure 24-5 present worksheets to assist you in the layout of the custom printouts for portrait and landscape orientations, respectively. As part of the definition of a custom printout, the user establishes the portion of the available height and width of the printout which is to be used for the graphic presentation of the data by defining the coordinates of the origin (lower left corner) and the height and width of the graphic. Similarly, the user defines the origin of each printed text string, the character size and the orientation (vertical or horizontal). It is recommended to begin by using a copy of one of these worksheets to sketch an approximation of the desired custom printout.

Figure 24-4 ***Portrait Worksheet***

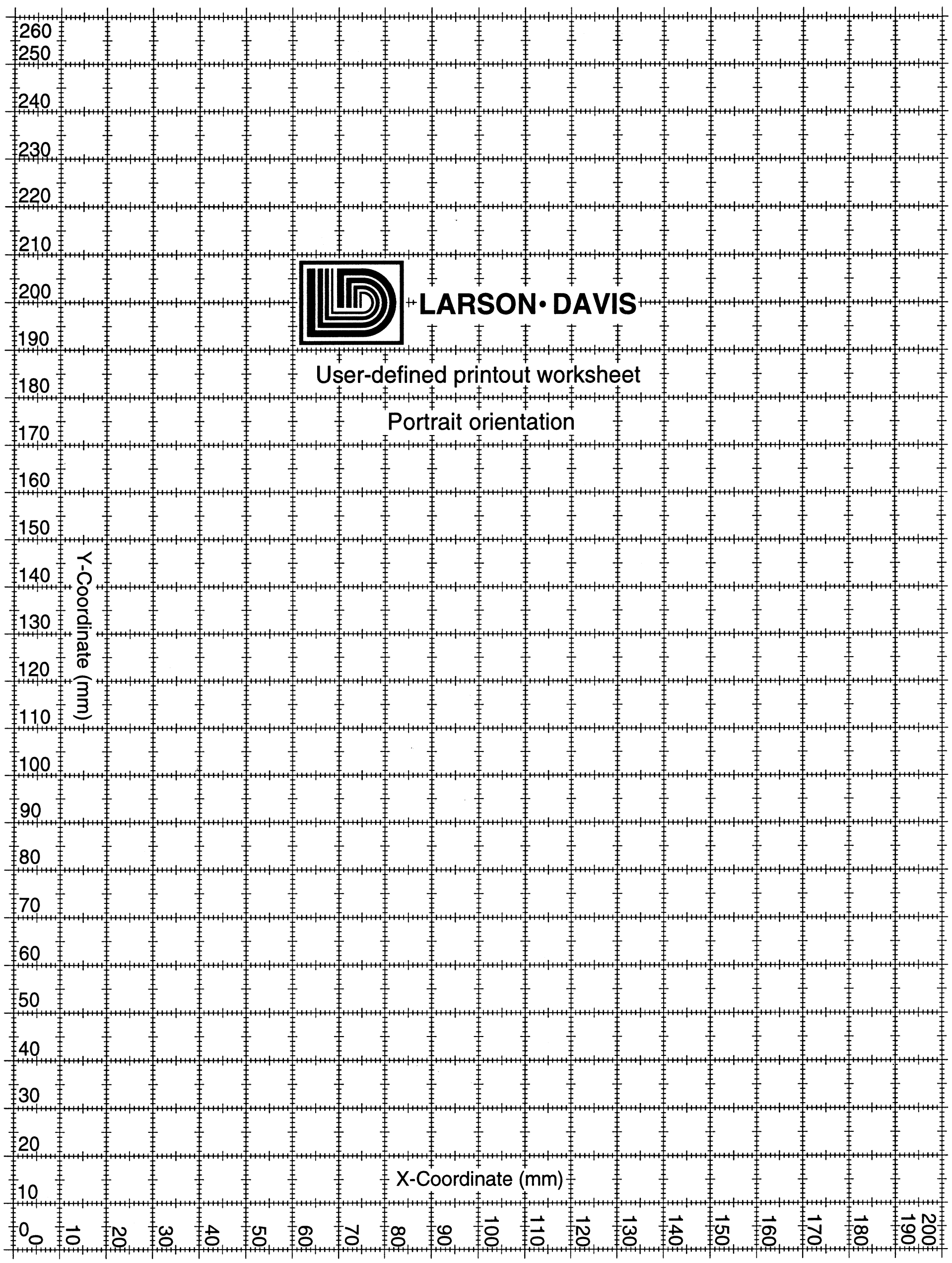
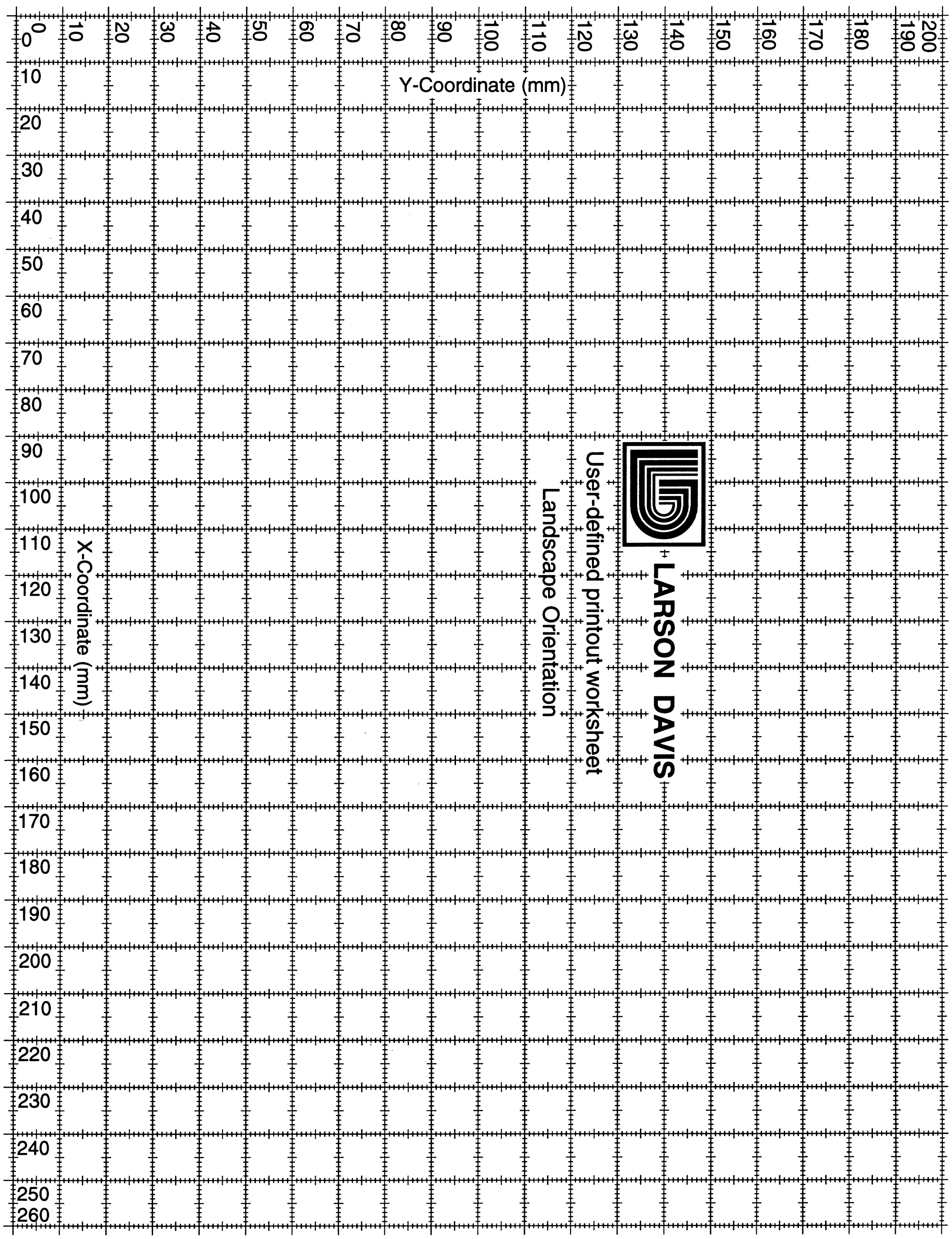
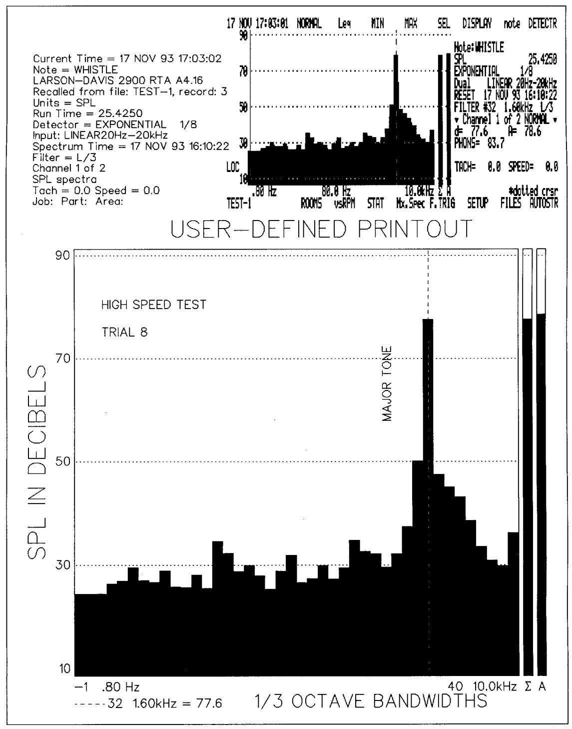


Figure 24- 5 ***Landscape Worksheet***



As an example, consider the custom printout illustrated in Figure 24-6.

Figure 24-6 ***Example of Custom Printout***



This particular example includes two separate graphic areas: a direct reproduction of the LCD display from the analyzer itself and a scaled custom graphic at the bottom of the page. The text strings printed on the upper portion of the printout describe parameters associated with the measurement of the data, all of which are stored as part of the data block in the analyzer and available for printout under user control. The various other text lines such as “USER-DEFINED PRINTOUT”, “SPL IN DECIBELS”, “1/3 OCTAVE BANDWIDTHS”, “HIGH SPEED TEST”, “TRAIL 8” and “MAJOR TONE” are user-defined text strings.

The location, size and orientation of each of the measurement setup text strings and the user-defined text strings are under user control, so the general form of a custom printout could be very different from this example. However, certain of these items fall into groups and are either set ON or OFF when configuring a custom printout. Referring to Figure 23-3, the softkey **GRAPH [A]** is pressed to initiate the configuration of the custom graphic portion of the printout. If this has been selected to be an active part of the present configuration (turned ON), there will be a small symbol ▼ displayed to the left of the softkey label. If this symbol is not displayed, it indicates that the custom graphic portion of the printout is turned OFF and it will not be printed.

Press the softkey **SCREEN [B]** to configure the position, size and orientation of the printout reproducing the LCD display on the analyzer screen. You may set this to ON or OFF.

Press the softkey **TEXT [C]** to configure the location, size and orientation of the measurement setup text strings stored with the data block in the instrument. There are fifteen of these, and each may be turned ON or OFF independent of the others.

The softkey **FRAME [D]** turns ON or OFF the printing of a frame outlining the page of the printout.

The softkey **ORIENT [H]** selects either portrait or landscape orientation of the printout.

Custom Graphic Configuration

The eight softkeys explained below are used to define the separate user-defined text lines, including location, size, and orientation. Each may be set ON or OFF individually.

To configure the custom graphic portion of the printout, press **GRAPHIC [A]**. In response to the message “Print SCALABLE GRAPHIC ?” on the upper right of the screen, press **YES [A]** if a graphic is desired in the printout or **NO [C]** if it is not. If NO, the display will return to the Edit Menu. If YES, the configuration procedure will continue with the display of the message

"Enter Position:"
"X = nn.n Y = nn.n mm"

Enter numerical values defining the origin (lower left corner) of the graphic.

Following this, the message

"Enter Size:"
"W = nn.n H = nn.n"

will prompt you to enter values for the width and height of the portion to be used for the custom graphic in the same manner as the coordinates of the origin were input above.

Next, the message "Print FRAME, AXES & GRID?" prompts you to select either **YES [A]** or **N [C]** to turn ON or OFF the printing of these parameters of the custom graphic printout.

The following message "Print DOTTED CURSOR?" permits you to select whether or not the dotted cursor, with a digital printout of the frequency and amplitude corresponding to the cursor position, is to appear in the custom graphic printout. Select **YES [A]** or **NO [C]**. If YES, the message will prompt you for input of the origin of the Trace Legend block on the custom printout. The Trace Legend is applicable only with the vsRPM/Speed display format of the analyzer where multiple curves are displayed simultaneously. The trace legend is a table indicating the line type used for each of the different curves as well as the level at the cursor position for each curve. If turned ON for other display formats, nothing different will be seen on the Custom Printout.

The next message "Print CLASS LINES?" prompts you to decide whether or not the class lines are to be included on the custom graphic printout. Select **YES [A]** or **NO [C]**.

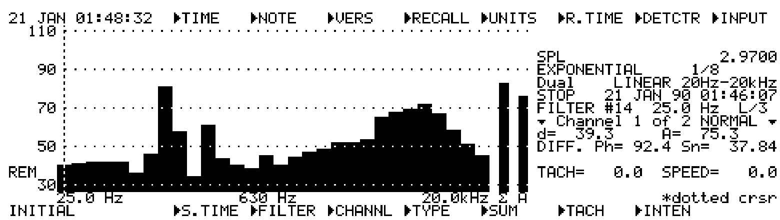
LCD Graphic Printout

A reproduction of the LCD display on the analyzer as part of the custom printout can be configured by pressing **SCREEN [B]**. If YES, messages on the upper right of the screen will prompt for input of the origin, width and height of the portion of the printout to be used for the reproduction of the LCD display, in the same manner as the parameters for the custom graphic were input. If NO, this function will be turned OFF for the custom printout.

Measurement Parameter Text Strings

Press **TEXT [C]** to access the Text Menu shown in Figure 24-7.

Figure 24- 7 ***Text Menu***



Each of the fourteen labeled softkeys refers to a particular parameter associated with the measurement and stored as part of the data block whose status or value may be presented as a text string on the custom printout. The specific parameters are as follows:

<u>Softkeys</u>	<u>Softkey Functions</u>
TIME [A]	Current Time
NOTE [B]	Note (stored with data block)
VERS [C]	Version Number of Analyzer Firmware
RECALL [D]	If data has been recalled from memory, identifies the filename, data type and record number
UNITS [E]	Units of displayed parameter
R.TIME [F]	Run Time of the measurement
DETCTR [G]	Detector and Averaging Time
INPUT [H]	Input setting (analog filters)
S.TIME [I]	Spectrum Time (of measurement)
FILTER [J]	Filter Type
CHANNL [K]	Channel Number
TYPE [L]	Data Type

Softkeys	Softkey Functions
SUM [M]	Sum value
TACH [N]	Tach and Speed values
INTEN [O]	Job, Part and Area Names

Upon pressing any of these keys, and responding to the prompt by pressing **YES [A]**, subsequent messages will prompt you to input the origin, character height and orientation of the printout of that text string. The text string can be oriented to print horizontally from left to right by selecting **LEFT [A]** or vertically from lower to upper by selecting **UP [C]**.

To turn OFF the printing of that particular text string in the custom printout, select **NO [C]** in response to the original prompt message “Print <string parameter> ?”. You can configure all of these text strings as desired, yet select individually whether each is to appear on the final printout.

User Text Strings

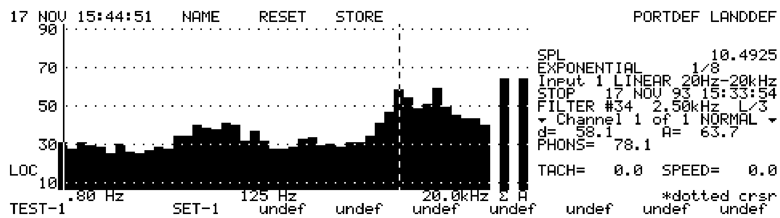
The eight softkeys below the screen on the Edit Menu are used to define user-defined text strings, which may be up to twenty-five characters in length. These are configured much the same as the measurement setup text strings described above. Upon pressing any one of these, the message “Print USER TEXT # n?” indicates that you are addressing the nth user-defined text string. Press **YES [A]** to print that text. Subsequent messages will prompt you to input the origin, size and orientation of the printed string. The final message “Enter text” prompts you to input the actual text string using the alphanumeric keypad, the press **EXIT**. The symbol will appear to the left of the label for each user-defined text string which has been set ON and which will be printed.

Selecting **NO [C]** in response to the original message “Print USER TEXT # n?” will turn OFF the printing of that particular string. When the text string has been set to OFF, the symbol will not appear to the left of its softkey label.

Storing a Custom Printout Setup to a Softkey

From the Print Menu, press **SETUPS [J]** to access the Custom Print Setup Menu shown in Figure 24-8.

Figure 24- 8 ***Setups Menu***



When delivered, all eight of the softkey labels the screen will read “undef” to indicate they are as yet undefined. Before storing a custom print setup, assign a label identifying the custom printout setup to one of the softkeys as follows: Press **NAME [A]**. The message “Push Setup To Name” will prompt you to press the softkey to which the label is to be assigned. The message “Enter setup name:” will then prompt you to input the label name using the alphanumeric keypad and press **EXIT**. The newly entered name will now appear as the label of that softkey.

To store the custom print setup presently active, press **STORE [C]** and respond to the message “Store in this setup:” by pressing the softkey to which the setup name has been assigned. Up to eight different custom print setups can be stored by name, one for each softkey. To delete all the setups previously stored, and to return all the softkey labels to “undef”, **press RESET [B]**. The message ““ARE YOU SURE?” prompt for confirmation of the reset operation. Press **YES [A]** or **NO [C]** as appropriate.

Recalling a Custom Print Setup from a Softkey

To recall, or make active, a custom print setup which has been stored to a softkey, simply press the labeled softkey to which the desired setup has been previously stored. The message “Overwrite current setup?” warns that the presently active setup will be lost when the one being recalled is made active. Press **YES [A]** to proceed with the recall or press **NO [C]** to abort the recall operation.

Storing Print Setups to Memory

The storage operation described above is to store a single custom print setup to one of the eight softkeys in the Custom Print Setup Menu. These setups will remain active in non-volatile memory when the instrument is shut off. However, it is possible to store the entire set of eight setups, with their softkey labels, to non-volatile memory as well. Thus, while you can have access of up to eight different setups from the Custom Print Setup Menu, any number of these sets of eight setups can be recalled from memory as well, replacing the eight which were previously available. Press **STORE** to store the present set of Custom Print Setups to memory. The Message “STORE-Print Setup N” on the upper right of the screen indicates that this set of Custom Print Setups have been stored as the Nth record of the type Print Setup in memory.

Recalling Print Setups from Memory

To recall a Print Setup from memory, press **RECALL**. The message “Overwrite ALL SETUPS?” on the upper right of the screen warns that upon recall all the custom print setups presently stored in the eight labeled softkeys will be lost. If these are of importance, they should be stored as described in the proceeding paragraph prior to recalling another set from memory. Press **YES [A]** to continue with the recall operation or press **NO [C]** to abort the recall operation.

Upon pressing **YES [A]**, the message “RECALL-Print Setup N” on the upper right of the screen will indicate that the Nth Print Setup (consisting of eight setup softkeys) has been recalled. If this is not the record number corresponding to the Print Setup desired, use the left and right arrow keys to page through the sequence of record numbers until the desired record number is displayed, then press **EXIT**.

NOTE: While paging through the sequence of Print Setup records the softkey labels stored with that record number will appear on the screen, making it easy to determine when the desired Print Setup record has been accessed.

Default Custom Printout Setups

Two pre-defined user setups, one in portrait format and one in landscape format, are included to assist you in the creation of custom printouts. These are selected by pressing **PORTDEF [G]** or **LANDEF [H]**, respectively, and responding to the message “overwrite current setup?” on the upper right of the screen by pressing **YES [A]**. Examples of these default custom printouts are shown in Figure 24-9 and Figure 24-10.

Figure 24- 9 ***Portrait Default***

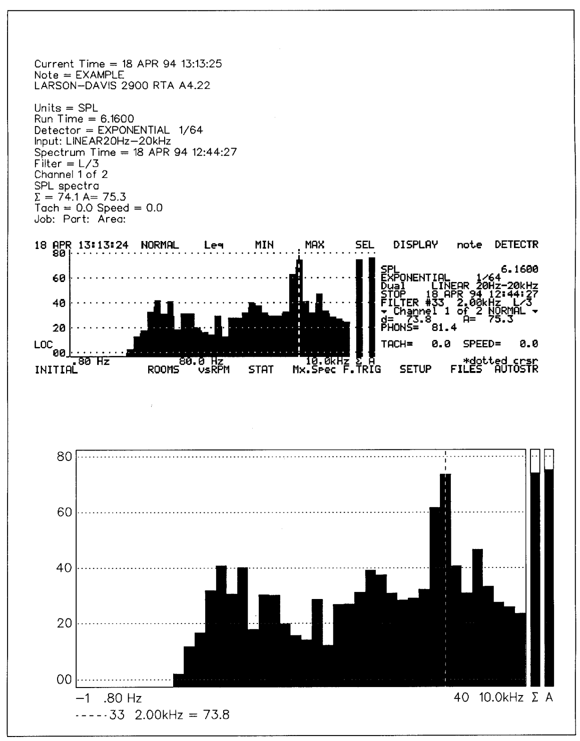
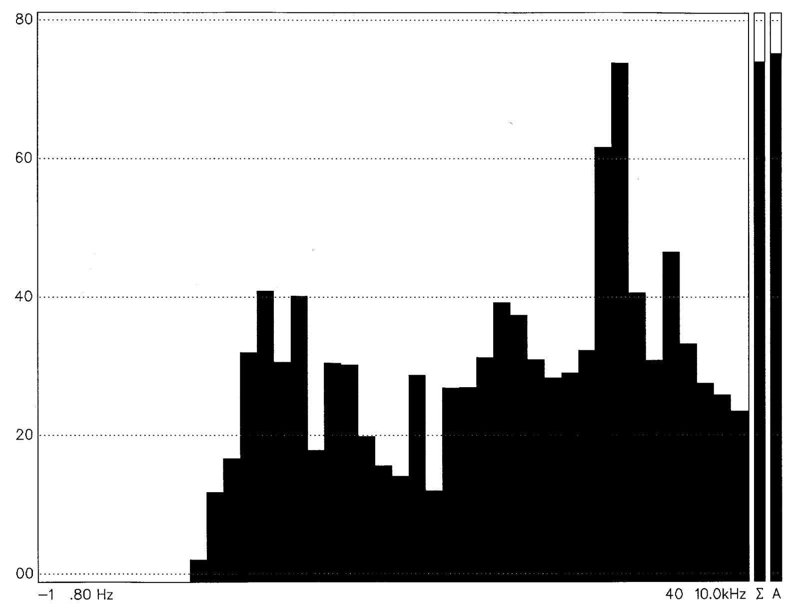


Figure 24-10 ***Landscape Default***



Initiating Printing of a Custom Printout

With the desired custom printout active and the desired data block displayed, simply press **user [I]** to initiate the printout. The message “Print Error” on the upper right of the screen indicates a problem with the communication between the analyzer and the printer. To abort a printout, simply press **abort [P]**. In cases where the printer has a buffer memory, the printing cannot be stopped until the buffer has been printed and the analyzer is again in communication with the printer.

CHAPTER

25 Frequency Domain Synchronous Averaging (FDSA)

The 3000+ has the capability to perform a patent pending method of synchronous averaging. Previously, synchronous averaging has always been done in the time domain and triggered from a third trigger signal or a voltage slope state in the original signal. There are occasions when this is not possible because of poor signal conditions or inaccessible triggers. To solve these problems and to provide a simple means to synchronously average intensity signals, **FDSA** can be used.

Benefits of FDSA

- Does not require an external trigger (or 3rd channel).
- Works even if the trigger signal is “weak”.
- Can be used for synchronous averaging of sound intensity data.

How FDSA Works

The trigger signal is generated by the spectrum itself:

- A reference frequency is selected where the phase is set to zero.

- Time and phase are related in the frequency domain by the following relationship.

$$\text{Phase} = e^{jwT}$$

- A phase transform is applied to the spectrum which relocates (in time) the spectrum.
- After the transform spectra may be summed to perform the analysis.

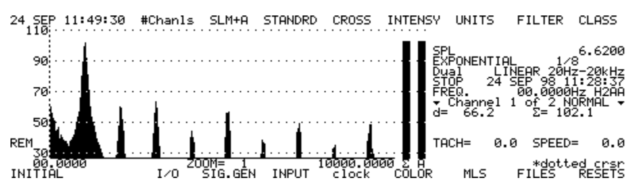
It is also possible to add multiple conditions. A second reference can be chosen so that its phase is also coherent. With two trigger conditions, various signals can be used as triggers to improve signal to noise ratios. Two triggers are usually required when ZOOM is active and not based at zero.

Selecting FDSA

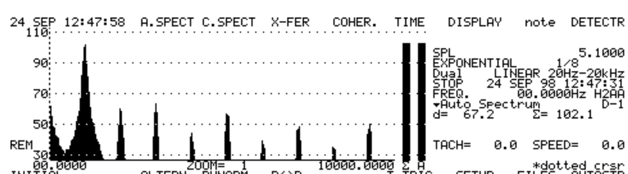
The FDSA feature is accessed through the Cross Analysis menu or the Intensity Analysis menu. To access the FDSA function:

Step 1 Put the 3000+ into FFT analysis mode.

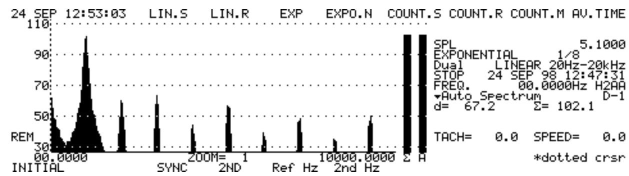
Step 2 From the system menu select **CROSS [D]** or **INTENSITY [E**].



Step 3 Press the **EXIT** hardkey and the CROSS channel analysis menu will appear.

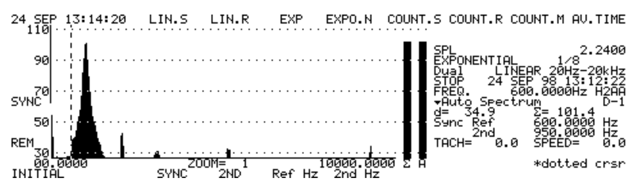


Step 4 Press the **DETECTOR [H]** key to access the FDSA functions.



The Detector menu provides the following options for FDSA:

Key Name	Key Function
SYNC [I]	Toggling this key turns the averaging (FDSA) on and off.
2ND [J]	Turns the second reference on or off.
Ref Hz [K]	Sets the frequency of the reference to the current cursor position.
2ND Hz [L]	Sets the frequency of the second reference to the current cursor position.



While in the DETECTOR menu the current status of the averaging is shown. The averaging runs to a maximum of 32,767 counts from the time of reset.

After it reaches the count of 32,767 the data sum will not change. A reset should be performed prior to each measurement.

CHAPTER

26 Softkey Menus

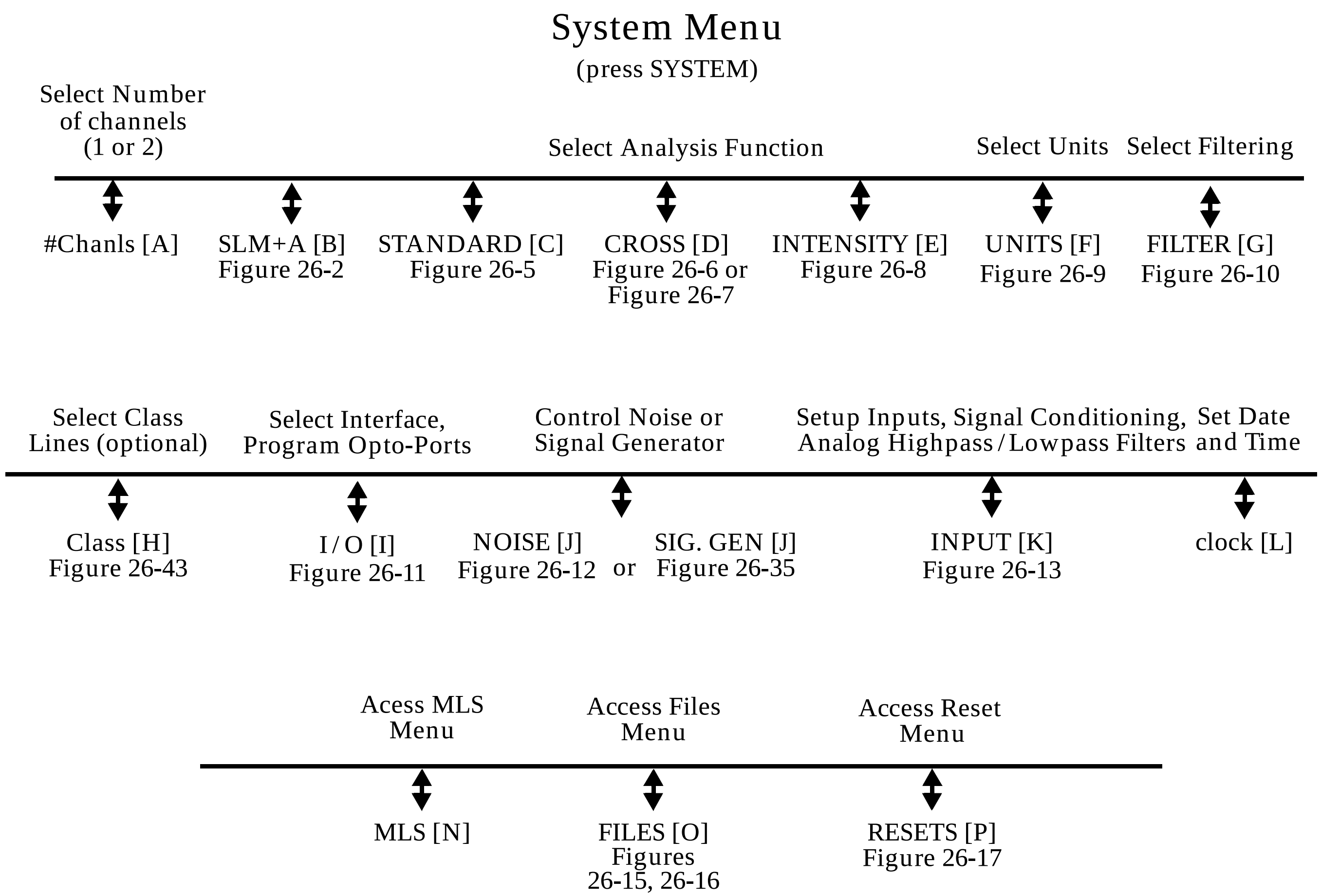

System Menu
(press SYSTEM)
Select Number of channels (1 or 2)
Select Analysis Function
Select Units
Select Filtering
#Chanls [A]
SLM+A [B] Figure 26-2
STANDARD [C] Figure 26-5
CROSS [D] Figure 26-6 or Figure 26-7
INTENSITY [E] Figure 26-8
UNITS [F] Figure 26-9
FILTER [G] Figure 26-10
Select Class Lines (optional)
Select Interface, Program Opto-Ports
Control Noise or Signal Generator
Setup Inputs, Signal Conditioning, Analog Highpass/Lowpass Filters
Set Date and Time
Class [H] Figure 26-43
I/O [I] Figure 26-11
NOISE [J] Figure 26-12
or
SIG. GEN [J] Figure 26-35
INPUT [K] Figure 26-13
clock [L]
Acess MLS Menu
Access Files Menu
Access Reset Menu
MLS [N]
FILES [O] Figures 26-15, 26-16
RESETS [P] Figure 26-17

Figure 26-1

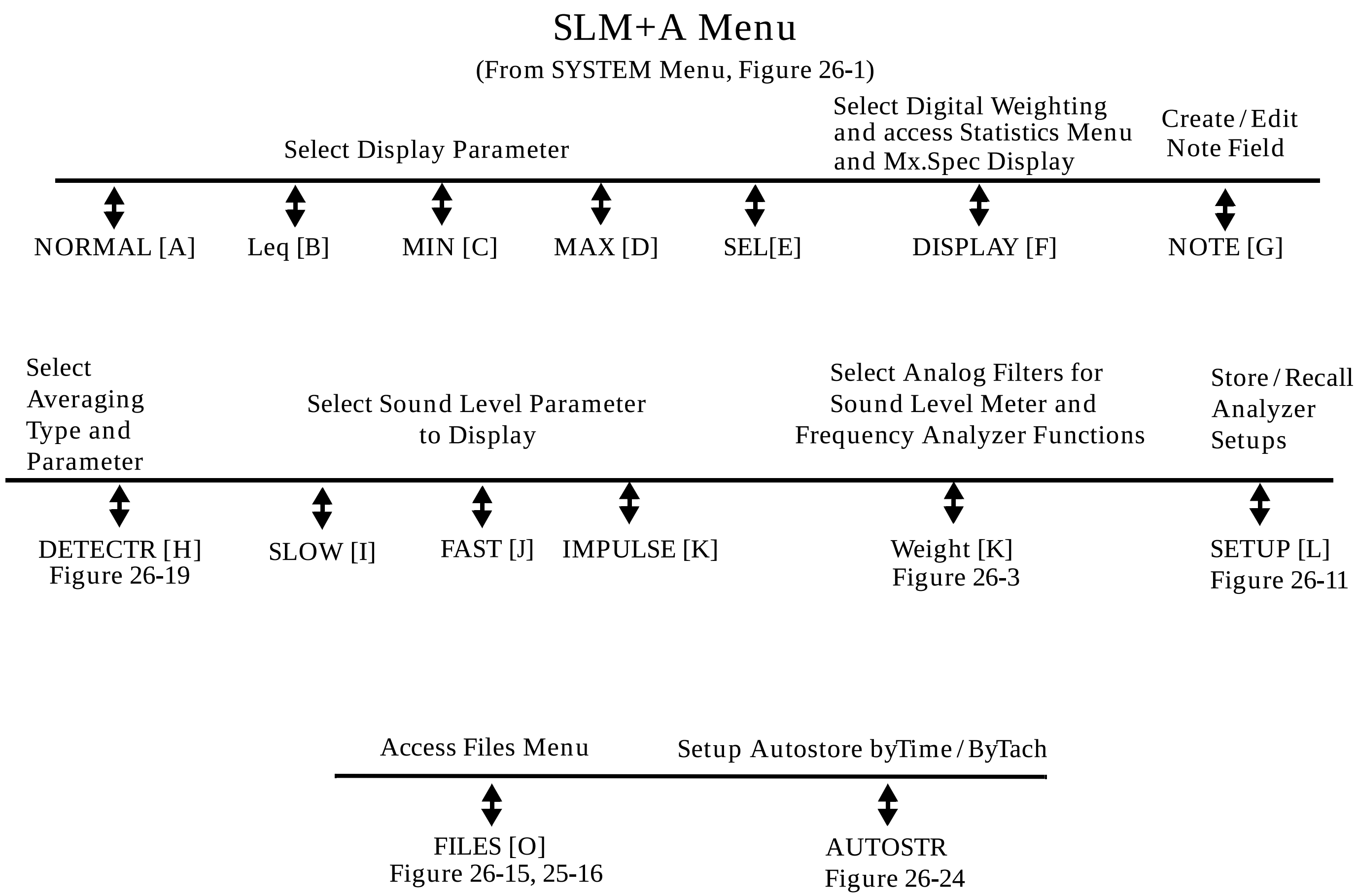

SLM+A Menu
(From SYSTEM Menu, Figure 26-1)
Select Display Parameter
Select Digital Weighting and access Statistics Menu and Mx.Spec Display
Create / Edit Note Field
NORMAL [A]
Leq [B]
MIN [C]
MAX [D]
SEL[E]
DISPLAY [F]
NOTE [G]
Select Averaging Type and Parameter
Select Sound Level Parameter to Display
Select Analog Filters for Sound Level Meter and Frequency Analyzer Functions
Store / Recall Analyzer Setups
DETECTR [H] Figure 26-19
SLOW [I]
FAST [J]
IMPULSE [K]
Weight [K] Figure 26-3
SETUP [L] Figure 26-11
Access Files Menu
Setup Autostore byTime / ByTach
FILES [O] Figure 26-15, 25-16
AUTOSTR Figure 26-24

Figure 26-2

SLM Weight Menu

(from SLM Menu, Figure 26-2)

Select Weighting for Analysis Function

A-WGT [A]	C-WGT [B]	1-20k [C]	20-20k [D]	1-10k [E]	20-10k [F]	WIDE [H]

Select Weighting for SLM Function

A-WGT [I]	C-WGT [J]	1-20k [K]	20-20k [L]	1-10k [M]	20-10k [N]

Figure 26-3

SLM Display Menu

(from SLM Menu, Figure 26-2)

Access vsRPM Graphics Menu	Access Statistics Menu	Display Max Spectrum	Create / Edit Note Field
vsRPM [A] Figure 26-21C	STAT [B] Figure 26-22	Mx.Spec [C]	note [G]

Access Digital Display Weighting Menu	Toggle Bandwidth Compensation ON / OFF
Dig.WGT [I] Figure 26-19A	BWNORM [P]

Figure 26-4

Standard Analysis Menu (1 and 2 Channels)

(Exit from System Menu, Figure 26-1, after selecting STAND 1 or STAND 2)

Select Display Parameter

- NORMAL [A]
- Leq [B]
- MIN [C]
- MAX [D]
- SEL [E]

Select Digital Weighting and Broadband Readout

- DISPLAY [F] Figure 26-18

Create / Edit Note Field

- note [G]

Select Averaging Type and Parameter

- DETECTR [H] Figure 26-19B

Access Room Acoustics Menu

- ROOMS [I] (octave filters only) Figure 26-20

Access vsRPM Graphics Menu

- vsRPM [J] Figure 26-21C

Access Statistics Menu

- STAT [K] Figure 26-22

Display Spectrum corresponding to the maximum broadband level

- Mx.Spec [L]

Frequency Trigger

- F.TRIG [M] Figure 26-23

Store / Recall Analyzer Setups

- SETUP [N] Figure 26-14

Access Files Menu

- FILES [O] Figure 26-15, 26-16

Setup Autostore byTime / byTach

- AUTOSTR [P] Figure 26-24

Figure 26-5

Cross Analysis Menu with Octave Filtering

(Exit from System Menu, Figure 26-1, after selecting CROSS and Octave Filters)

Select Display Parameter

A.SPEC [A]

C.SPEC [B]

X-FER [C]

COHER [D]

INVERSE [E]

Create / Edit
Note Field

note [G]

Select Averaging
Type and Parameter

DETECTR [H]
Figure 26-19B

Select Complex
Parameter
Magnitude / Phase

ALTERN [I]

Toggle Bandwidth
Compensation
On / Off

BWNORM [J]

Store / Recall
Analyzer Setups

SETUP [N]
Figure 26-14

Access
Files Menu

FILES [O]
Figure 26-15, 26-16

Setup Autostore
byTime / byTach

AUTOSTR [P]
Figure 26-24

Figure 26-6

Cross Analysis Menu with FFT Filtering

(Exit from System Menu, Figure 26-1, after selecting CROSS and FFT filtering)

Select Display Parameter (Toggle Between Upper and Lower Rows with Display [F])

A.SPECT [A]	C.SPECT [B]	X-FER [C]	COHER [D]	TIME [E]	DISPLAY [F]
A.CORRE [A]	C.CORRE [B]	IMPULSE [C]	INVERSE [D]	CEPSTRM[E]	

Create / Edit Note Field	Select Averaging Type and Parameter	Select Complex Parameter Real / Imag or Magnitude / Phase	Toggle Bandwidth Compensation On / Off	Select Coordinate System Polar / Rectangular
note [G]	DETECTR [H] Figure 26-19B	ALTERN [I]	BWNORM [J]	P < > R [K]

Display Liftered Spectrum	Setup Time Trigger	Store / Recall Analyzer Setups	Access Files Menu	Setup Autostore byTime / byTach
LIFTER [L]	T.TRIG [M] Figure 26-25	SETUP [N] Figure 26-14	FILES [O] Figure 26-15, 26-16	AUTOSTR [P] Figure 26-24

Figure 26-7

Intensity Analysis Menu

(Exit from System Menu, Figure 26-1, after selecting Intensity)

Access Power Summation Menu	Select Display Parameter			
POWER [A]	INTNSTY [B]	QUALITY [C]	SPL [D]	P.VELOC [E]

Select Digital Weighting and Broadband Readout	Create / Edit Note Field	Select Averaging Type and Parameter	Input Label Names for Job, Part and Area		
DISPLAY [F] Figure 26-19A	note [G]	DETECTR [H] Figure 26-19B	job [I]	part [J]	area [K]

Input Values for Pressure, Temperature and Spacer	Input Value for Surface Area	Store / Recall Analyzer Setup	Access Files Menu	Setup Autostore byTime / by Tach
pr / t / sp [L]	meter2 [M]	SETUP [N] Figure 26-14	FILES [O] Figure 26-15, 26-16	AUTOSTR [P] Figure 26-24

Figure 26-8

Units Menu

(from System Menu, Figure 26-1)

Select Differentiation or Integration	Assign Name to Units Key	Make all Channels have same Units	Reset User-defined Units	Input Noise Floor Levels for SLM Mode
SLOPE [A]	name[B]	SAME [C]	R.UNITS [D]	Noise.F [E]

Input a mV / Unit Calibration Value	Input a dB / Volt Calibration Value	Input a numerical value to calibrate level of indicated frequency band	Select dBμV Units	Select SPL Units	Select User-defined Units
mV cal [P]	V cal [G]	level [H]	dBμV [I]	SPL [J]	[K] - [P]

Figure 26-9

Filter Menu

(from System Menu, Figure 26-1)

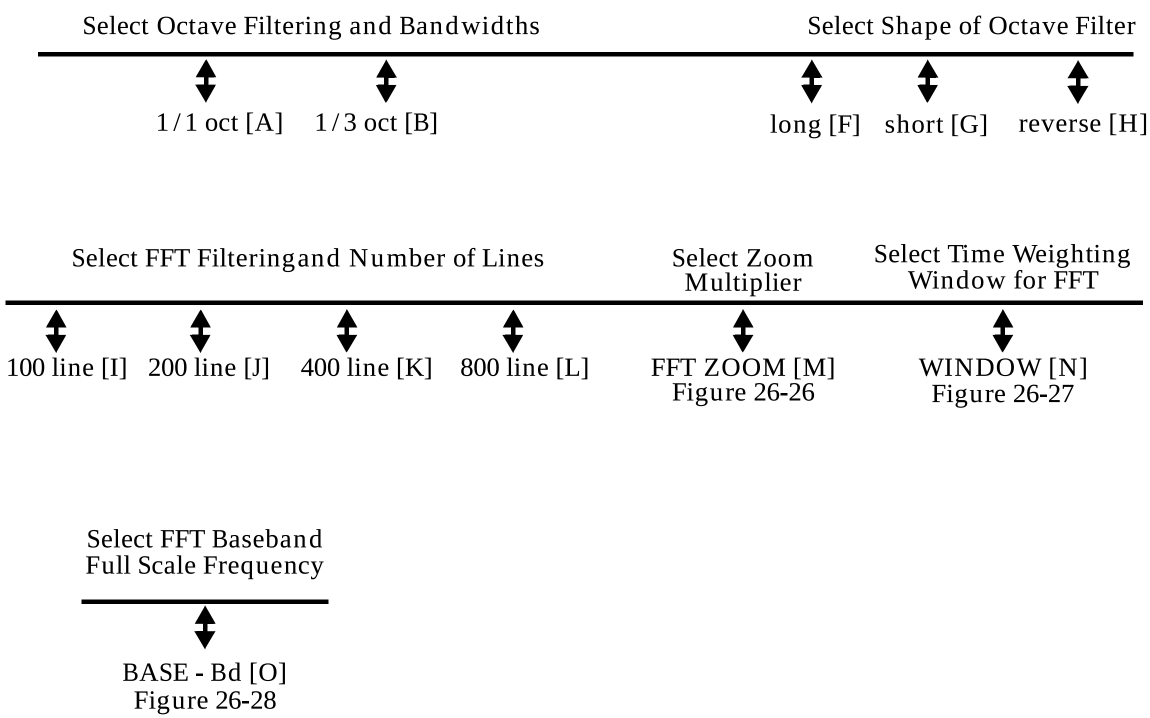


Figure 26-10

I/O Menu

(from System Menu, Figure 26-1)

Enable or disable remote control	Select Parameter to be represented by DC voltage output	Configure I/O Port	Program Opto-Ports (rear panel)		Set Beeper Operation	Switch opto-isolated inputs between remote switch mode and intensity probe mode
3200RC [A]	DC out [B]	IO port [C] Figure 26-31	KEY A [D]	KEY B [E]	Beeper [F] Figure 26-32	RS1/INT [G]

Select RS-232 Interface	Select RS-232 Baud Rate					Select Parallel Port
RS-232 [I] or T. Link [I] (Passby option)	1200 [J]	2400 [K]	4800 [L]	9600 [M]	19200[N]	P. Port [P]

Figure 26-11

Noise Menu

(from System Menu, Figure 26-1)

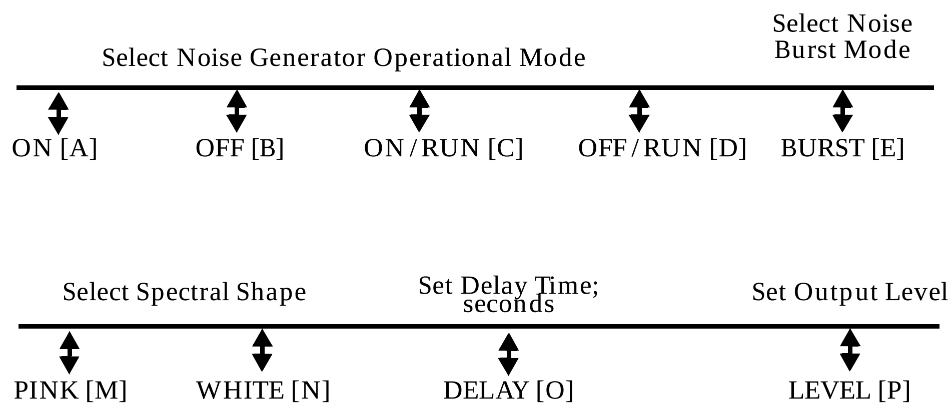


Figure 26-12

Input Menu

(from System Menu, Figure 26-1)

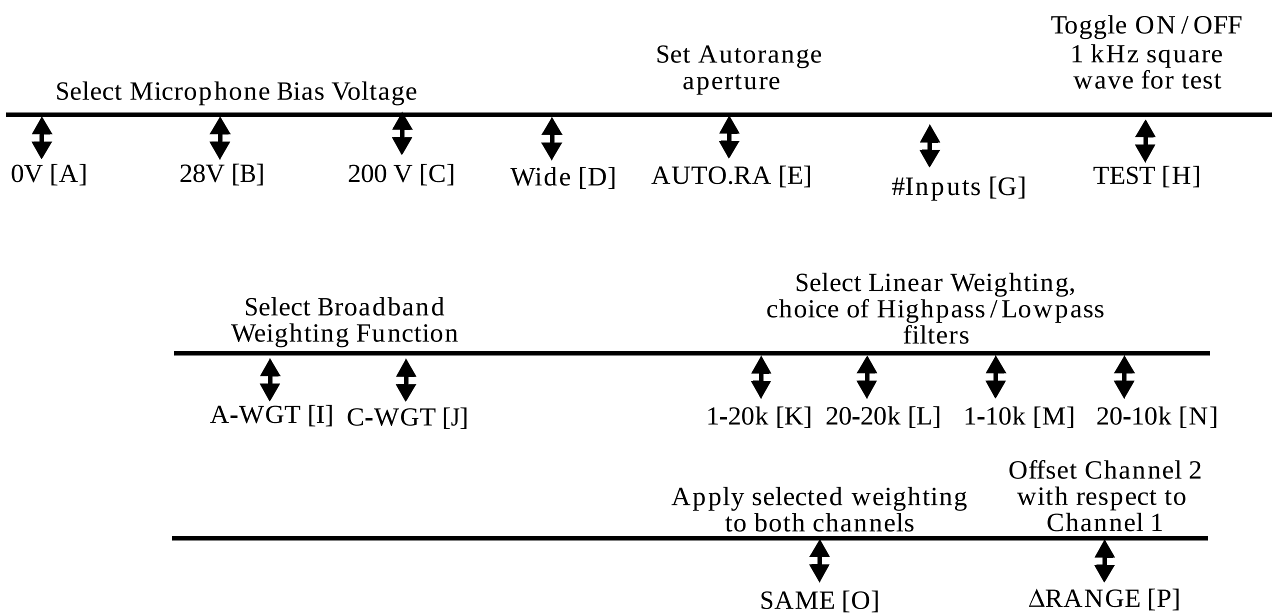


Figure 26-13

Setup Menu

(from System Menu, Figure 26-1 or one of the Analysis Menus, Figures 26-5, 6,7, and 8)

Assign Name to Setup Key	Reset User-defined Setups	Store present Setup to User-defined Setup	Assign Analyzer to Boot-up to a User-defined Setup
name [B]	R. SETUP [D]	STORE [E]	->BOOT [F]

Create/Edit Note Field	Select Default Setup	Select User-defined Setup
note [G]	DEFAULT [I]	[J] - [P]

Figure 26-14

Files Menu (Memory)

(from System Menu, Figure 26-1 or one of the Analysis Menus, Figures 26-5, 6, 7, and 8)

Create and Name a Memory File	List Records in Highlighted Memory File	Move Left Side Highlight Up	Store Highlighted Memory File to Disk	Move Right Side Highlight Up
create [A]	RECORDS [B] Figure 26-16	[C]	disk [D]	[E]

Format Disk	Delete Highlighted Memory File	Rename Highlighted File	Move Left Side Highlight Down
format [F]	delete [I]	rename [J]	[K]

Recall Highlighted Disk File to Memory	Move Right Side Highlight Down	Delete Highlighted Disk File	Rename Highlighted Disk File
mem [M]	[M]	delete [N]	rename [O]

Figure 26-15

Files Menu (Records)

(from Files Menu (Memory), Figure 26-15)

Create and Name a Memory File	Move Left Side Highlighted Up	Move Right Side Highlighted Up	Edit Note Field of Hightlighted Record
create [A]	↑ [C]	↑ [E]	note [G]

Exit File Menu and setup analyzer to be same as Highlighted Record	Delete Highlighted Memory File	Rename Highlighted Memory File	Move Left Side Highlight Down
KEEP [H]	delete [J]	rename [J]	↓ [K]

Move Right Side Highlight Down	Delete Highlighted Record (Right Side) from Highlighted Memory File (Left Side)	
↓ [M]	delete [N]	→ ByTach [O]

Figure 26-16

Resets Menu

(from System Menu, Figure 26-1)

Reset analyzer memory	Reset units	Reset all user-defined setups	Reset the active RT60 buffer
R.MEMRY [A]	R.UNITS [B]	R.SETUP [C]	R.RT60 [D]

Reset all user-defined digital weighting	Reset all user-defined key macros	Reset the statistics table	Resets Normalization Function
R.USER [E]	R.MACRO [F]	R.STATS [G]	R. NORM

Figure 26-17

Display Menu

(from one of the Analysis Menus, Figure 26-5, 6, 7, and 8)

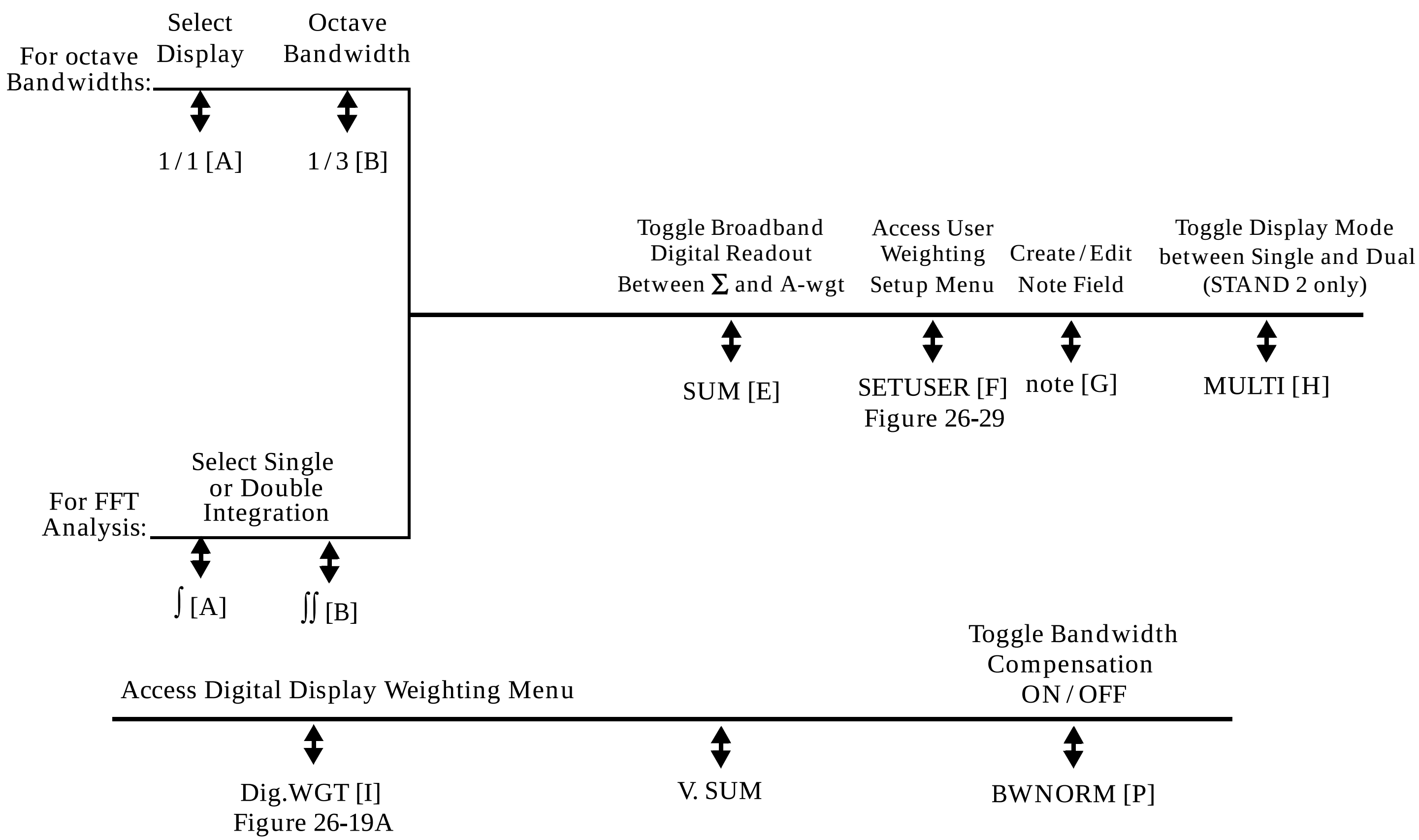


Figure 26-18

Digital Display Menu

(from Display Menu, Figure 26-19A)

Select form of
Zwicker Loudness (octave filters only)

Zw.FREE [A]   Zw.DIFF [B]

Select Digital Display Weighting

NO WGT [I]   A [J]   C [K]   USER [L]   -A [M]   -C [N]   -USER [O]

Figure 26-19A

Detector Menu, Octave Bandwidths

(from one of the Analysis Menus, Figure 26-5, 6, 7, and 8)

Select Averaging Type

LIN.S [A] LIN.R [B] EXP [C] BT/EXP [D] BT/LIN [E]

Select Averaging Time

AV.TYPE [H]

Detector Menu, FFT Analysis

Select Averaging Type

LIN.S [A] LIN.R [B] EXP [C] EXPO.N [D] COUNT.S [E] COUNT.R [F] COUNT.M [G]

Select Averaging Time

AV.TYPE [H]

Figure 26-19B

Rooms Menu

(from the Standard Analysis Menu, Figure 26-5)

Evaluate NC Rating of 1/1 Octave Spectrum	Evaluate NR Rating of 1/1 Octave Spectrum	Evaluate RC Rating of 1/1 Octave Spectrum	Enter Surface Area of Partition, m2	Enter Volume of Receiving Room, m3	Access RT60 Menu
NC [A]	NR [B]	RC [C]	surface [D]	volume [E]	RT60 [F]

	Select Standard Group to follow for Calculations and Access Menu			Use with Source, Receive, and Background keys to define data block to be used	Display Source Room Spectrum
note [G]	ASTM [I] Figure 26-21A	ISO [J] Figure 26-21B	RECALL [K]	SET-> [L]	SOURCE [M]

Display Receiving Room Spectrum	Display Receiving Room background spectrum	Reset Receiving Room background spectrum to a very small value
RECEIVE [N]	BACK.GD [F]	CLR.BG [P]

Figure 26-20

ASTM Menu

(from the Rooms Menu, Figure 26-20)

Evaluate Noise Reduction and Noise Isolation
NR [A]

Evaluate Normalized Noise Reduction and Normalized Noise Isolation Class
NNR [B]

Evaluate Transmission Loss and Sound Transmission Class
TL [C]

Evaluate Field Transmission and Field Sound Transmission Class
FTL [D]

Evaluate Normalized Impact Sound Pressure Level and Field Impact Isolation Cross

note [G]

Ln [I]

Figure 26-21A

ISO Menu

(from the Rooms Menu, Figure 26-20)

Evaluate Level Differnce and Weighted Level Difference

Evaluate Souind Reduction Index, Apparent Weighted Sound Reduction Index, and Airborne Sound Insulation Margin

D [A] R’ w [B] C [C] Ctr [D]

Evaluate Standardized Level Difference and Weighted Apparent Standarized Sound Reduction Index

DnT [E] note [G]

Evaluate Normalized Impact Sound Pressure Level, Weighted Normalized Impact Sound Pressure Level, and Impact Sound Protection Margin

Evaluate Standardized Impact Sound Pressure Level and Weighted Standardized Impact Sound Pressure Level

L’n [I] Ci [J] L’nT [K]

Figure 26-21B

vsRPM Menu

(from theStandard Analysis Menu, Figure 26-5)

Set slope +, -, ±	Amplitude sensitivity of real-time tracking display	Toggle Tagbars ON / OFF	Set limits for horizontal axis, RPM / Speed	
SLOPE [A]	Δlevel [B]	TAGBARS [C]	X-Span [D]	
Toggle horizontal axis between RPM / Speed	**Create / Edit Note Field**	**Select Peak Hunt Parameter for Post-process Order Tracking**	**Set Status of selected trace**	**Hide or Unhide selected trace**
X-AXIS [F]	note [G]	WIDTH [H]	STATUS [I]	HIDE [J]
Access Tach / Speed Scaling Menu	**Select parameter for vsRPM graph**		**Select trace to set parameters**	
TACHSET [K] Figure 25-30	B-ORDERS [L] B-FREQ [L]		CAL [M]	PEN [N]
Select Channel for indicated trace	**Select parameter for indicated trace**			
CHANNEL [O]	FREQ [P] ORDER [P]			

Figure 26-21C

Statistics Menu

(from SLM Display Menu, Figure 26-4 or Standard Analysis Menu, Figure 26-5)

ON / OFF Toggle Statistics	Set time interval for update of statistics matrix	Set Amplitude Range of statistics matrix	Reset statistics matrix	Hide or Unhide selected trace	Clear traces	Redraw using latest data from statistics table
ON / OFF [A]	S.TIME [B]	dB-SPAN [C]	R.STAT [D]	HIDE [F]	CLEAR [G]	REDRAW [H]

Store statistics table	Recall statistics table	Merge recalled statistics table with active statistics table
STO-TBL [I]	RCL-TBL [J]	MRG-TBL [K]

Select N for Ln value of indicated trace	Switch between channels 1 and 2
N [N]	CHANNEL [O]

Figure 26-22

Frequency Trigger Menu

(from Standard Analysis Menu, Figure 26-5)

Select Trigger Criteria
for cursor frequency

>= [I] +SLOPE [J] -SLOPE [K] <= [L]

Select Manual Trigger

MANUAL [M]

Select Automatic Rearm
following an Autostore

RE-ARM [N]

Turn off
Frequency Trigger

OFF [O]

Input level corresponding
to trigger criteria

level [P]

Figure 26-23

Autostore Menu

(from SLM Display Menu, Figure 26-2 or Standard Analysis Menu, Figure 26-5)

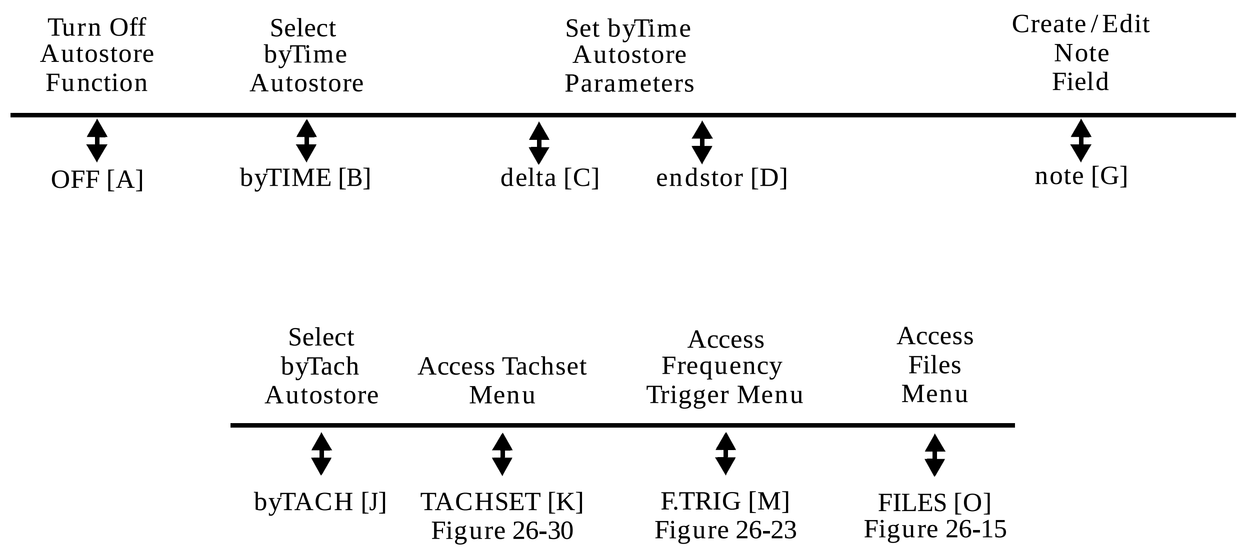


Figure 26-24

Time Trigger Menu

(from Cross Analysis with FFT Filtering Menu, Figure 26-7)

Select Spectrum Display	Select Time Domain Display	Create / Edit Note Field
A.SPECT [A]	TIME [E]	note [G]

Select Complex Parameter Real / Imag or Mag / Phase	Toggle Bandwidth Compensation ON / OFF	Select Polar or Rectangular Coordinates
ALTERN [I]	BWNORM [J]	P<>R [K]

Toggle Slope Criteria ≠ or Ø	Adjust Trigger Level	Adjust Trigger Delay + or -	Adjust Ch 2 delay w / r to Ch 1	Turn off Time Trigger
SLOPE [L]	LEVEL [M]	DELAY [N]	2-DELAY [O]	OFF [P]

Figure 26-25

FFT Zoom Menu

(from Filter Menu, Figure 26-10)

Select Zoom Multiplier

Use horizontal arrow keys to select Center Frequency of Zoom Analysis

1 [A]	2 [B]	4[C]	8[D]	16 [E]	32 [F]	64 [G]	128 [I]	256 [J]	512 [K]	BASE-Hz [O]

Figure 26-26

FFT Time Weighting Menu

(from Filter Menu, Figure 26-10)

Select Time Weighting Function

RECT. [A] HANNING [B] FLAT [C] ZEROPAD [D] IMPACT [E] EXP-2 [F] EXP-4 [G] EXP-6 [H]

Figure 26-27

FFT Baseband Menu

(from Filter Menu, Figure 26-10)

20 kHz[A] 10 kHz [B] 5 kHz [C] 2.5 kHz [C]

Figure 26-28

Set User Menu

(from Display Menu, Figure 26-18)

Clear Working Data Buffer	Store Displayed User Weighting	Recall Stored User Weighting
CLEAR [C]	STO.USR [I]	RCL.USR [J]

Add Recalled User Weighting to Displayed Curve	Subtract Recalled User Weighting from Displayed Curve	Create a User Weighting Curve from Displayed Spectrum
ADD [K]	SUBTRCT [L]	MAKE [M]

Toggle Interpolation Function ON / OFF	Input Level Value for Cursor Position
INTERP [N]	edit [P]

Figure 26-29

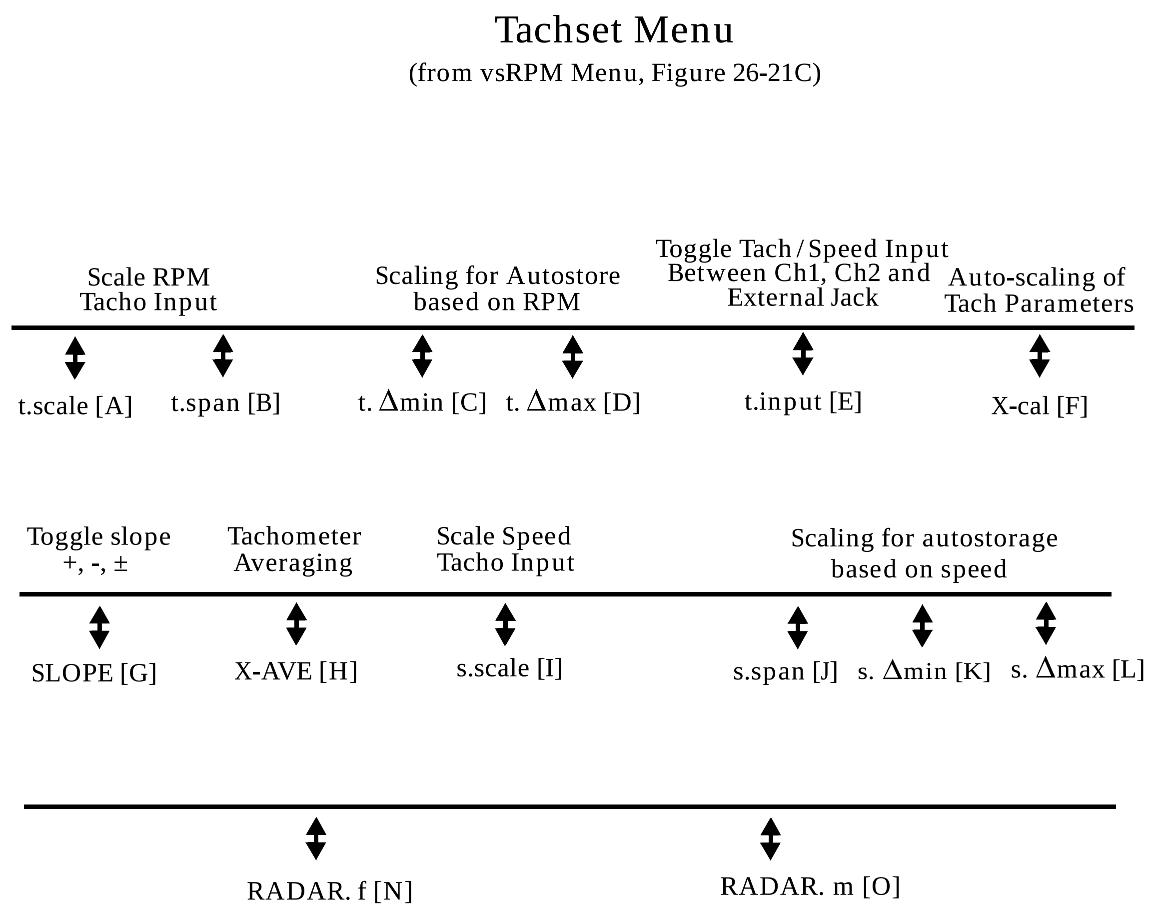

Tachset Menu
(from vsRPM Menu, Figure 26-21C)
Scale RPM Tacho Input
Scaling for Autostore based on RPM
Toggle Tach / Speed Input Between Ch1, Ch2 and External Jack
Auto-scaling of Tach Parameters
t.scale [A]
t.span [B]
t. Δmin [C]
t. Δmax [D]
t.input [E]
X-cal [F]
Toggle slope +, -, ±
Tachometer Averaging
Scale Speed Tacho Input
Scaling for autostorage based on speed
SLOPE [G]
X-AVE [H]
s.scale [I]
s.span [J]
s. Δmin [K]
s. Δmax [L]
RADAR. f [N]
RADAR. m [O]

Figure 26-30

IO Port Menu

(from I/O Menu, Figure 26-11)

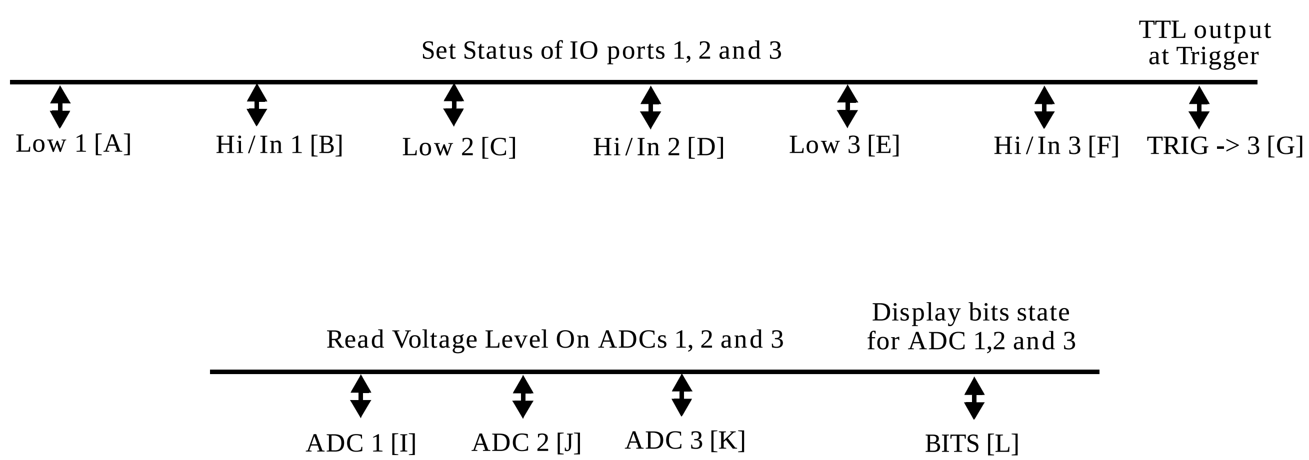


Figure 26-31

Beeper Menu

(from I/O Menu, Figure 26-11)

No Beeper	Beeper Active for all conditions	Beeper for key presses	Beeper for overload	Beeper for error conditions
NONE [A]	ALL [B]	KEYS [C]	OVER [D]	ERROR [E]

Figure 26-32

Standard Recall Menu

(press **RECALL**)

Select Waterfall Display	Perform Block Averaging	Perform Block Maximum	Perform Block Average	Exit from Recall Menu; Keep 3000+ Setup the same as the last recalled datablock
W.Fall [A]	B.AVE [B]	B.MAX [C]	B.SUM [D]	KEEP [H]

Delete Record presently displayed

DELETE [I]

*recall [P] Reassigns horizontal arrow keys to recall role (after controlling cursor)

Figure 26-33

Autostore Recall Menu

(press **RECALL**, with autostore activated)

Select Waterfall Display	Perform Block Averaging	Perform Block Maximum	Perform Block Summation	Present Data in versus Time Format, selected frequency	Exit from Recall Menu; Keep 3000+ Setup the same as the last recalled datablock
W.Fall [A]	B.AVE [B]	B.MAX [C]	B.SUM [D]	vsTIME [E]	KEEP [H]

Delete Record presently displayed	Recall Record Previous in Storage Sequence	Recall Record next in Storage Sequence
DELETE [I]	A.Prev [N]	A.Next [M]

*recall [P] Reassigns horizontal arrow keys to recall role (after controlling cursor)

Figure 26-34

Signal Generator Menu

(from System Menu, Figure 26-1)

Select Signal Generator Operational Mode

- ON [A]
- OFF [B]
- ON / RUN [C]
- OFF / RUN [D]

Control Attenuation

- <-ATT [G]
- ->ATT [H]

Access Menu for Selected Signal Type

- SINE [I] Figure 26-36
- PINK [J] Figure 26-38
- WHITE [K] Figure 26-40
- PULSE [L] Figure 26-42

Set Delay Time, seconds

- DELAY [O]

Figure 26-35

Sine Generator Menu

(from Signal Generator Menu, Figure 26-36)

Set Frequency Sweep Limits
- Fstart [A]
- Fend [B]

Set Sweep Times
- T.se [C] or Nse [C] or R.se [C]
- T.es [D] or Nes [D] or R.es [D]

Set Sweep Levels
- Lse [E]
- Les [F]

Control Attenuation
- <-ATT [G]
- ->ATT [H]

Sweep Control
- @START [I]
- @END [J]
- SWEEP [K]
- LIN [L] or LOG [L]
- TIME [M] or COUNT [M]

Access 2-tone Menu
- 2-tone [N] Figure 26-37

Set Auto Level
- Auto-L [O]

Adjust Response
- FILTER [P]

Figure 26-36

Dual Tone Menu

(from Sine Generator Menu, Figure 26-36)

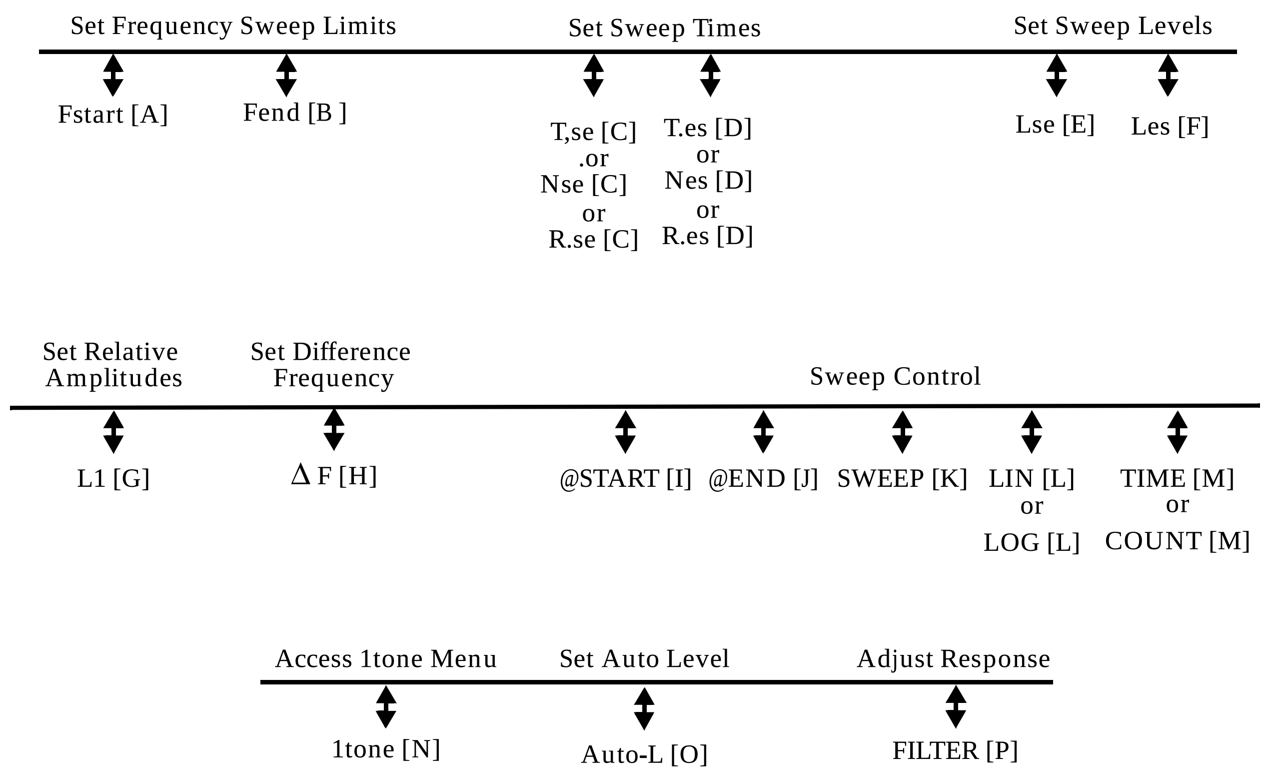


Figure 26-37

Wideband Pink Noise Menu

(from Signal Generator Menu, Figure 26-35)

Access Bandlimited Pink Noise Menu	Set Amplitude	Control Attenuation	
1/3 [A] Figure 26-39	LEVEL [E]	<-ATT [G]	->ATT [H]

Figure 26-38

Bandlimited Pink Noise Menu

(from Wideband Pink Noise Menu, Figure 26-38)

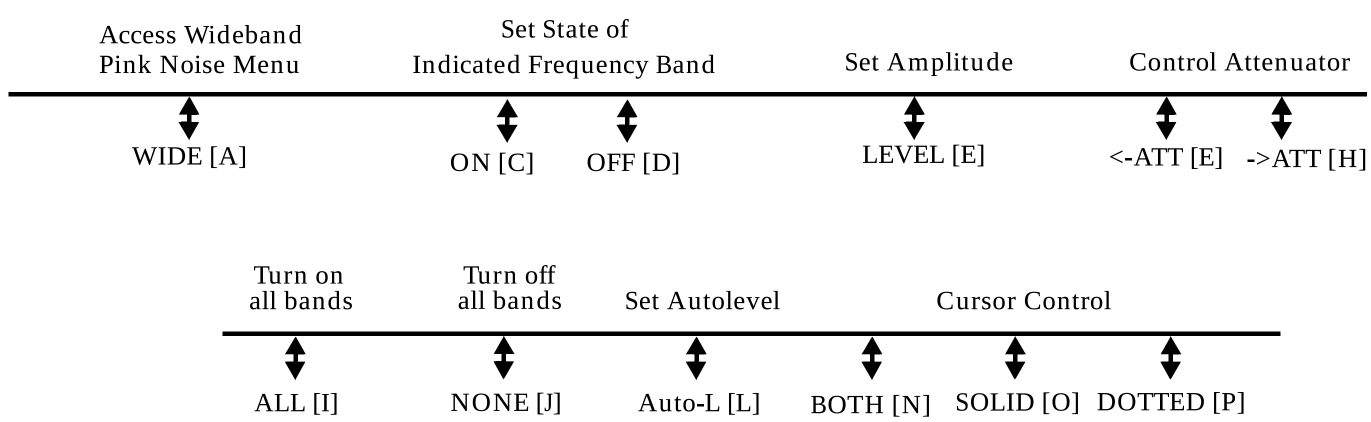


Figure 26-39

Wideband White Noise Menu

(from Signal Generator Menu, Figure 26-35)

Access Psuedo-White Noise Menu	Set Amplitude	Control Attenuation	
PSUEDO [A] Figure 25-41	LEVEL [E]	<-ATT [G]	->ATT [H]

Figure 26-40

Psuedo-White Noise Menu

(from Wideband White Noise Menu, Figure 26-40)

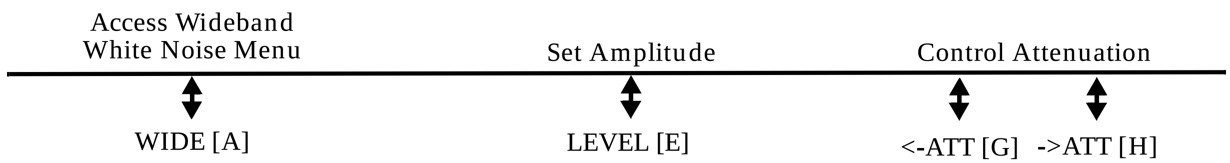


Figure 26-41

Pulse Noise Menu

(from Signal Generator Menu, Figure 26-35)

Select Direction of Pulses	Select Time On	Select Time Off	Set Amplitude
POS / NEG [B]	T.on [C]	T.off [D]	LEVEL [E]

Figure 26-42

Class Lines Menu

(from System Menu, Figure 26-1)

User-defineable names for Class Line

Undef [A] Undef [B] Undef [C] Undef [D]

Set Class Lines ON / OFF

C.ON [F]
or
C.OFF [F]

Access Class Lines
Edit Menu

EDIT [I]
Figure 26-44

Access Class Lines
Setup Menu

SETUPS [J]

Name Class Lines

NAME [K]

Assign Class Lines

ASSIGN [L]

Select Judgement Type

a.JUDGE [N] USE KEY [O] m.JUDGE [P]

Figure 26-43

Class Lines Edit Menu

(from Class Lines Menu, Figure 26-43)

Select max / min	Select number of line repeats	Select Width between lines	Set qualifier parameter
max [A] or min [A]	# REPEAT [B]	Δ WIDTH [C]	qualify [D]

Shift to point at left	Shift to point at right	Add new point	Delete point	Horizontal arrows Control Cursor	Horizontal Arrows Move Frequency Location of Point
<-point [J]	->point [J}	add pnt [K]	del pnt [L]	*dotted [N]	*freq [O]

Figure 26-44

Class Lines Setup Menu

(from Class Lines Menu, Figure 26-43)

Assign Name to Setup Key	Reset Setups	Store Setups	Select default Setup Format
NAME [A]	RESET [B]	STORE [C]	DEFAULT [H]

Select Setup (labeled by defined Names)
[I] - [P]

Figure 26-45

SHIFT Menu

(Press hardkey SHIFT)

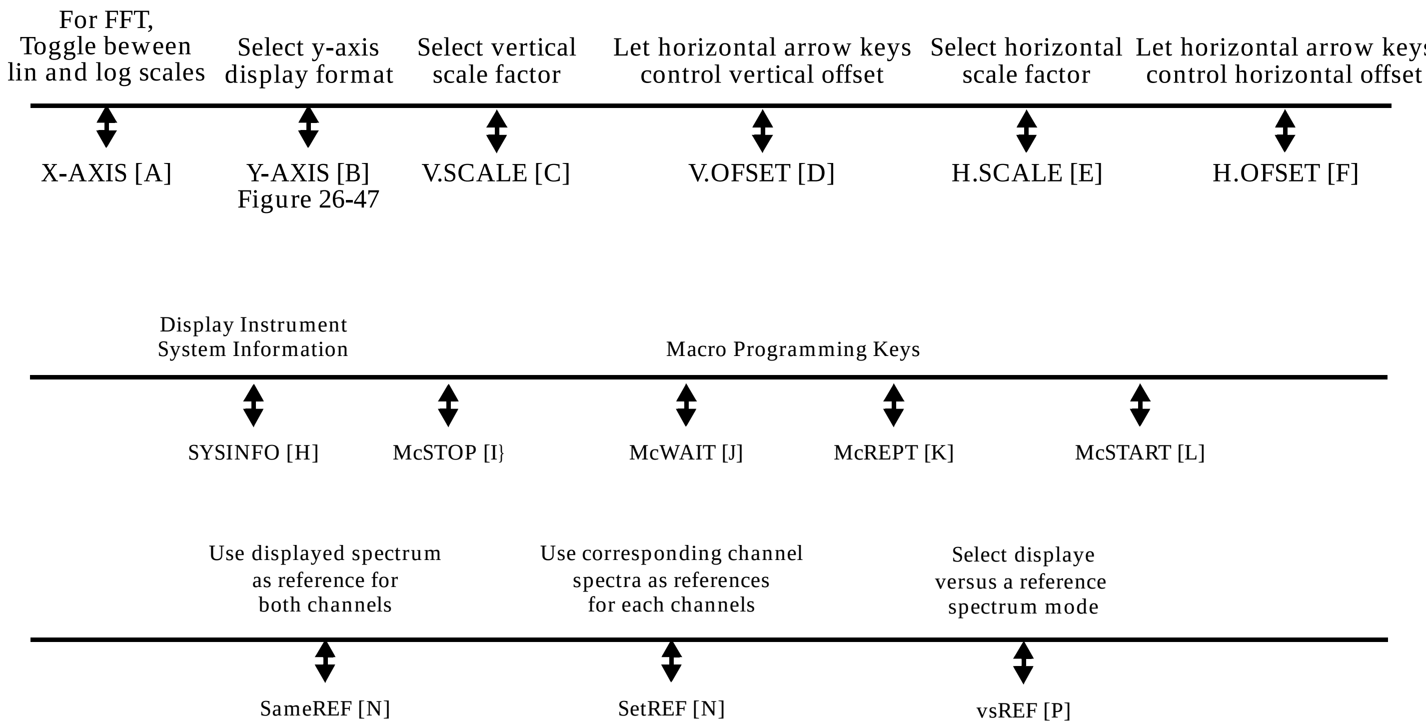


Figure 26-46

Y-AXIS Menu

(from SHIFT Menu, Figure 26-46)

Linear Units Linear Scaling	Log Units Log Scale	Linear Units Log Scale	Squared Linear Units Log Scale
LIN / LIN [A]	LOG / LOG [B]	LIN / LOG [C]	SQU / LOG [D]

Figure 26-47

Index

B

C

D

E

F

M

Q

R

S

X

Z