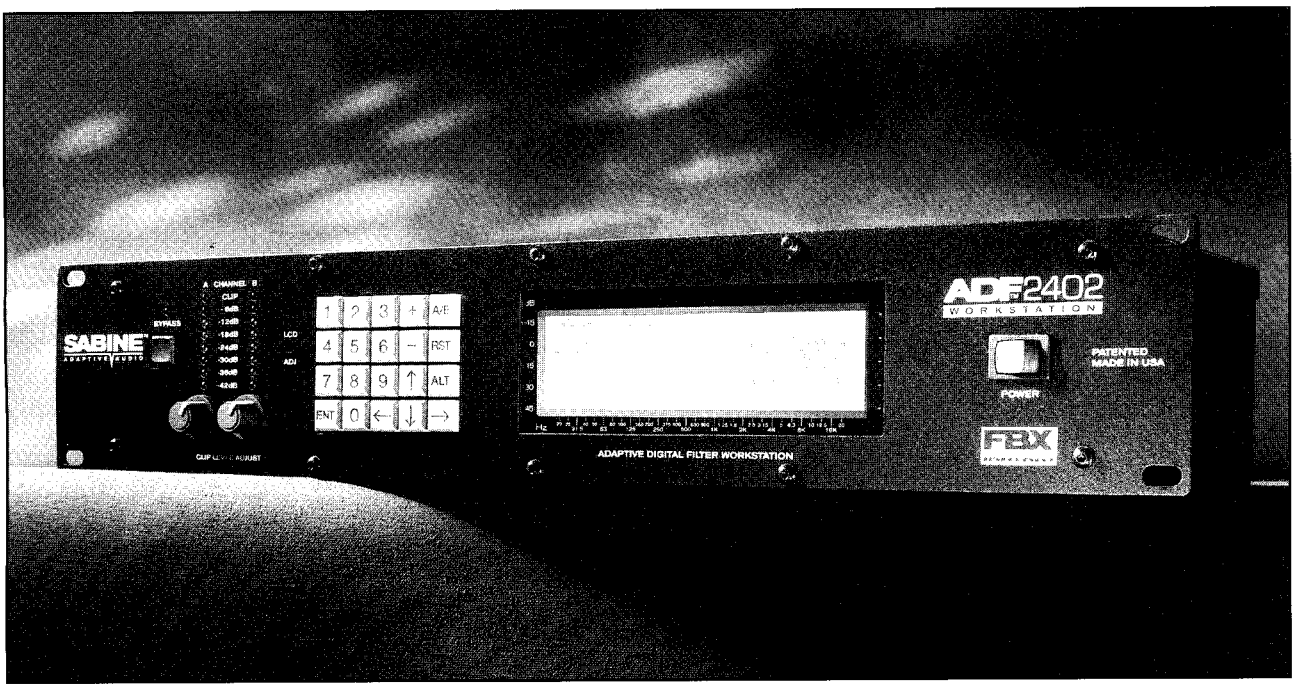
ADF™1201 WORKSTATION

ADF™2402 WORKSTATION

OPERATING GUIDE

Last 1201 built: 7/29/97

Ser.# 1201146



SABINE™
ADAPTIVE AUDIO

Quick Start

The ADF is pre-configured at the factory so you can automatically control feedback without a thorough understanding of all its powerful features and benefits. To add an ADF to your existing sound system, follow this step by step procedure:

1. Place the microphones and speakers in the physical locations where they will be during the program.

2. Patch the ADF just before the power amp or electronic crossover in the audio chain. Use low impedance (XLR) connectors (pin 2 high).

3. Plug the power supply into the ADF and then into the power outlet.

4. Pull the microphone sliders down on the mixer so the system will not emit feedback.

5. Power up the sound system, the ADF, and finally the power amp.

6. Adjust the LCD ADJ until your screen reaches the appropriate contrast.

7. Play a CD or other program material through the system.

8. Press the red BYPASS button. (The BYPASS LED will light to indicate BYPASS mode.) If you use a graphic equalizer to contour the sound color and eliminate feedback, remove all anti-feedback filters from the EQ. Set the EQ for sound color contour only. The ADF will provide the proper notching for feedback removal and prevention.

9. Press the BYPASS button again to return the ADF to ACTIVE mode. (The BYPASS LED will go out.) Adjust the front panel CLIP LEVEL so that the ADF clips at the same volume level as the power amp.

10. Turn off the CD program material. Lower the mixer's master volume. Raise the microphone sliders to the level needed in the performance.

11. Set up only one channel at a time. Turn down the mixer volume and the clip level knob for ADF channel B while setting channel A, and vice versa.

12. Select ADF's MAIN MENU 1 (Filter Control) and press ENTer. Now you can see the frequency, width and depth of the ADF's filters.

13. Press the front panel RST button to reset the filters. Hold until the filters are reset to 0 frequency and 0 depth (about three seconds). Channel A and channel B must be reset individually on the ADF-2402 in mono mode.

14. Now you must teach the ADF which frequencies feed back. SLOWLY raise the master volume slider. Inevitably, feedback will occur at a specific frequency. The ADF will automatically determine this frequency and place a 1/10 octave filter that cancels the feedback. You will notice that the filter's frequency, width and depth are displayed on the LCD.

15. Repeat step 13 until the seventh filter is set. Now lower the master volume a little so that the system is not on the verge of another feedback point. Make a mark on the master volume slider. This is the highest level you can reach and still control feedback without changing the system's configuration.

16. The sound system is now ready to use. Any new, transient feedback caused by moving or getting too close to the microphones will be automatically filtered by the three FBX DYNAMIC filters.

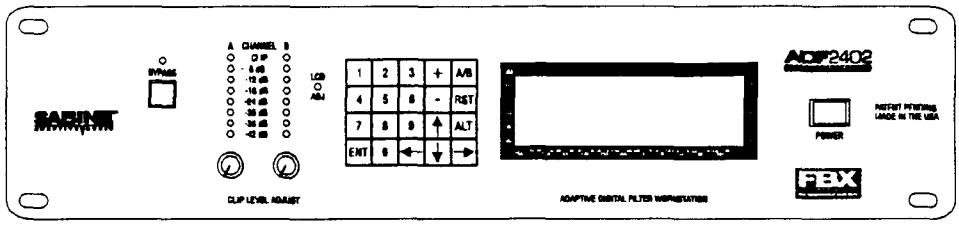
17. Read and follow the cautions at the end of this manual.

About Your ADF

Read the following detailed discussions and cautions to obtain the best performance from your ADF.

HARDWARE SET-UP:

ADF Front Panel



Mounting: Mount the ADF in a standard 19", 2- unit rack space.

Power On/Off: Power to the ADF is toggled on and off with the front panel power switch labeled POWER, found at the right end of the front panel.

CAUTION:

When the power to the ADF is OFF, the ADF is in BYPASS mode. In BYPASS mode, the input signal is routed directly to the output, fully bypassing and defeating any feedback protection which may have been in effect when the unit was turned on and functioning. Please read the CAUTIONs following the paragraph on BYPASS below regarding possible damage, and at the end of this manual regarding exposure to high sound levels.

Bypass: The ADF provides a hard-wired bypass: i.e., when in bypass mode, the input signal is routed directly to the output jacks without modification. The ADF automatically switches to bypass mode when the power is turned off. The bypass LED (red) illuminates when the system is in bypass mode.

CAUTION:

Remember that placing the ADF in BYPASS mode completely defeats (bypasses) the ADF's feedback extermination and filtering ability. Before turning off the power to the ADF, and/or before placing the ADF in BYPASS mode, make sure that your microphones are off or at low level so that feedback will not occur. Uncontrolled feedback may cause permanent damage to your sound system and building structures, as well as cause hearing loss to those people exposed to high sound levels. Please read the CAUTION regarding exposure to high sound levels, found at the end of this manual.

LCD Display: The user interacts with the ADF through the keypad and the LCD display. The LCD display is used to prompt the user for numeric or alpha-numeric entries, to display filter status numerically and graphically, and to display the menus that control the ADF's mode of operation.

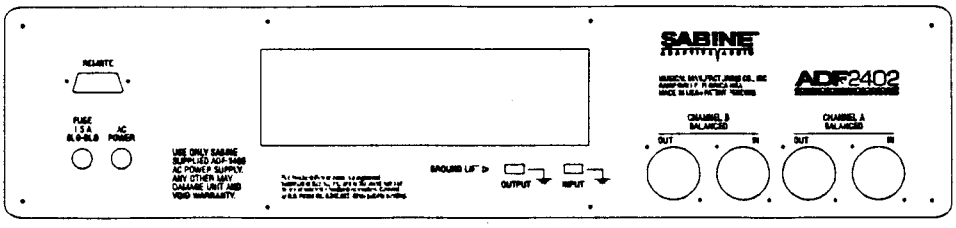
Keypad: The keypad is used for menu selection and numeric or alpha-numeric data entry. It consists of twenty keys with the following functions:

1. Ten keys (0 through 9) for numeric entry.
2. Four arrow keys for moving the cursor within the menus.
3. + and - keys for toggling or slewing certain alpha-numeric fields within menus.
4. An A/B key for toggling control and status between channels (ADF-2402 only).
5. ENTer key, which is used to indicate that an alpha-numeric entry is to be accepted as input, or that an action which is highlighted and blinking on the display is to be carried out.
6. ALT key that returns you to the MAIN MENU at any time.
7. RST key that resets all automatic feedback control filters to zero dB cut and zero frequency.

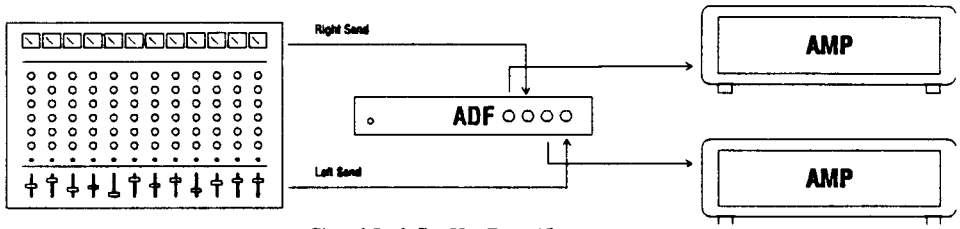
Clip Level Adjust: The CLIP LEVEL ADJUST potentiometer(s) on the front panel of the ADF is used to match the clip level of the ADF with the clip level of the power amp. Adjust the CLIP LEVEL so that the ADF clips at the same volume level as the power amp. This adjustment allows you to make optimum use of the system's dynamic range.

LCD Adjust: The LCD adjust allows the user to adjust the screen's contrast and viewing angle. Be sure to insert only non-conductive screwdriver tips into the unit.

ADF Back Panel



Signal Path: Insert the ADF in the signal path between the mixer and the power amp using three-prong XLR connectors (pin 2 high). The input signal should be balanced (symmetrical) and standard line level for best results. Place the ADF after the EQ, compressor and other effects. You can also insert the ADF on a channel insert point.



Signal Path Set-Up (Rear View)

AC Power: The ADF is equipped with an external power module (desk mount type). Make sure the power module has the correct primary plug configuration and input voltage specification required by your local power source (sometimes country specific). Plug the power module first into the ADF's AC POWER receptacle and then into the primary power outlet. (The power module has a screw mount hanger so that it can be mounted conveniently inside the equipment rack.) The power to the ADF is controlled by the ADF front panel POWER on-off switch.

Ground Lift: The ground lift switch lifts the cable shields off the chassis ground and removes inter-equipment ground loops. It is not recommended that both the input and output be lifted at the same time.

Remote: The remote connector provides optional interface via RS232.

HOW TO USE THE ADF MENU FEATURES

The ADF provides audio engineers with features and audio signal control never before available. The section below explains these features and how they can be used to best suit your specific application.

ADF MAIN MENU

When the power is turned on for the first time, the screen shown below appears. This screen is the ADF's MAIN MENU and provides access to the sub-menus.

MAIN MENU

Main Menu

1. Filter Control
2. Filter Diagram
3. R-Time Analyzer
4. Digital Delay
5. Noise Gate
6. Preset Table
7. Global Params
8. Change Passwd

Select: ■

MENU ITEM 1: FILTER CONTROL

Pressing 1 followed by ENTer takes you to the FILTER CONTROL MENU, as shown below. This is where the current configurations of the system's filters are numerically displayed by the ADF and controlled by the user. The column marked T displays the filter type (Fixed, Dynamic, or Parametric). The column marked FREQ displays the filter's center frequency. The column marked WIDTH displays the filter's width in decimal fractions of an octave. The column marked DEPTH displays the filter's depth in dB (or boost or cut in the case of parametric filters). The fields marked HPASS and LPASS display the shelving filter cutoff frequencies. The LOCKED field allows fixed filters to be "frozen" so they will not go deeper than the original level at set-up.

FILTER CONTROL MENU

	T	Freq	Width	Depth	T	Freq	Width	Depth
	P	0	0.10	-0	P	0	0.10	-0
	F	0	0.10	-0	F	0	0.10	-0
	F	0	0.10	-0	F	0	0.10	-0
	F	0	0.10	-0	F	0	0.10	-0
	F	0	0.10	-0	D	0	0.10	-0
	D	0	0.10	-0	D	0	0.10	-0
Channel indicator →	A	Locked: No		Hpass: None			Lpass: None	

Values shown are the factory default settings

It is very simple to control the ADF's filters using the keypad and the filter control screen. This screen serves two purposes: Current ADF filter status is displayed and user instructions are passed to the ADF. As the ADF sets automatic feedback control filters, their center-frequencies, widths, and depths are displayed on the screen. Any time the status changes, these changes are reflected by the display. Perhaps more important, the user controls filters from this screen. Initially, a filter which is not set is indicated by a center-frequency equal to 0 Hz, a width equal to 0.10-octave, and a depth of 0 dB (factory default). Filters not set retain the settings in the GLOBAL PARAMETERS menu.

TYPES OF FILTERS

Use the cursor keys to move the cursor to the Type field of the filter you wish to change. Pressing the + or - keys toggles the filter type between Parametric, Fixed and Dynamic, indicated by a P, F or D respectively.

TYPE F (Fixed) and D (Dynamic)

The unique and most useful feature of the ADF is its ability to automatically sense and eliminate feedback. There are two types of FBX automatic feedback control filters: FIXED and DYNAMIC. FIXED filters provide gain before feedback. DYNAMIC filters eliminate transient feedback that comes and goes throughout the program. Follow the QUICK START procedure described previously to "teach" the FBX automatic feedback control filters the positions and depths necessary to control feedback in your system.

BACKGROUND: The ADF uses a sophisticated algorithm (patented) to monitor the input signal and detect the onset of feedback. It then makes a precise determination of the feedback frequency and sets a filter of prescribed width (typically 0.10 octave, but this can be modified by the user in the GLOBAL PARAMETERS menu) at this frequency. Initially, the filter is only -3 dB deep, which is often sufficient to eliminate the feedback. If the same feedback frequency persists, the filter is deepened progressively to a maximum depth (which is also user-selectable) or until the feedback is eliminated.

FIXED FBX filters are used to eliminate feedback due to characteristics which are unlikely to change, such as room acoustics and fixed microphone installations. Once a FIXED filter is set, its center frequency remains fixed, but it may be deepened automatically, if necessary, to control additional feedback at the same frequency. In some cases the ADF may mistake music for feedback and drive the fixed filters deeper than necessary, such as in churches with pipe organs. The ADF gives you the option of locking the fixed filters so they won't go deeper than the original setting. This feature is especially useful in unattended PA systems. Look for the LOCKED selection at the bottom of the screen. Using the + /- keys, choose "Y" to lock the filters or "N" to disengage the LOCK FIXED feature.

DYNAMIC FBX filters are used to deal with transient feedback that comes and goes during a program. When a new feedback frequency occurs, a new DYNAMIC FBX filter is automatically assigned to eliminate the feedback. When all of the DYNAMIC filters have been used, the filter that was set earliest is reassigned to handle subsequent feedback, and so on.

One trait shared by both the FIXED and DYNAMIC FBX filters is their ability to track feedback. We have already said that DYNAMIC FBX filters will set new filters or recycle old filters to eliminate feedback and that FIXED FBX filters are not recycled. If, however, the feedback frequency is detected to be very close to an existing feedback frequency, it is presumed that this new feedback is the result of a "drift" in the resonating frequency of the original feedback. Drifting can be the result of changes in air temperature or humidity. In this case, the closest filter will be moved slightly to the new frequency to track this feedback. You can adjust the tracking values in the GLOBAL PARAMETERS menu.

If the P.A. system is moved from the original set-up, the ADF must be "re-taught" where to place filters to eliminate feedback. To reset: Press and hold the RST button for approximately three seconds to reset all FBX fixed and dynamic filters to 0 dB cut and 0 frequency. Both channels must be reset individually on the ADF-2402 if in mono mode.

(See the section on GLOBAL PARAMETERS for more information about controlling FBX automatic feedback control filters.)

TYPE P (Parametric)

Once a filter has its type set to P, it is possible to edit its frequency, width and depth by moving to the appropriate field with the cursor keys. You can place the center FREQuency of an ADF parametric filter anywhere between 20 Hz and 20 KHz with 1 Hz placement resolution. The filter's WIDTH ranges from 1.00 octave to 0.01 octave. The DEPTH can be set anywhere from -84 dB cut to +12 dB gain in 1 dB increments. The + and - keys can be used to alter frequency and width by pressing either key down. Holding the + or - key down will cause the numeric field indicated by the cursor to slew. When the desired frequency or width is reached, simply release the + or - key. The data will be entered automatically at that moment and during the slew operation. You can also edit the parametric filters in the FILTER DIAGRAM. This gives you a graphic view of the filters as you edit them. See the FILTER DIAGRAM section for more information.

Please note: If the filter frequency is 0, the width and depth cannot be entered; the ADF assumes the filter does not exist.

Unlike conventional analog parametric filters, the ADF's digital filters do not drift with temperature or cause phase-shifting outside of the filters, and for the first time, engineers can set extremely narrow parametric filters to eliminate power-line hum.

For example, to set the first filter as a parametric notch filter centered at 60 Hz with 0.01-octave width and -65 dB depth, move the cursor to the type field for the first filter. Press the + key until a P is displayed; then press ENTer. Using the cursor keys, move the cursor to the center FREQuency field. At this point, type 6 0, followed by ENTer. Move the cursor to the WIDTH field, where you next type 1, followed by ENTer, then move the cursor to the DEPTH field. Type - for the sign, then 6 5, then ENTer. The filter is now set. In some cases, additional filters may be required at 120 Hz and 180 Hz to eliminate power-line hum (50, 100 and 150 Hz in 50 Hz power systems).

Similarly, a fixed or dynamic FBX filter that has been set by the ADF can be "frozen" by moving to the appropriate type field and changing it to a P. At this point, the filter can be left as is or edited further.

High- and Low-Pass Filters

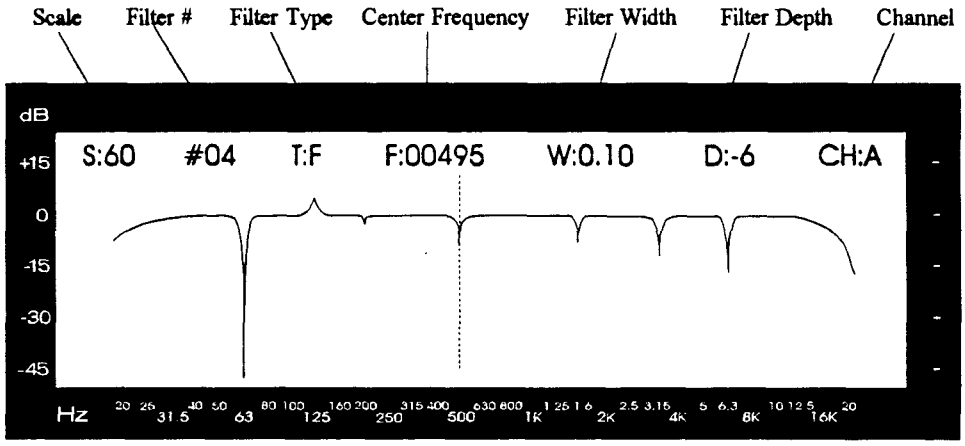
At the bottom of the filter control menu, you will find two fields that are labeled HPASS and LPASS. These are the high- and low-end roll-off filters that are used to custom-tailor the low- and high-end frequency response. The high pass filter can be used to suppress phenomena such as low-frequency rumble, while the low pass filter can be used for attenuating high-frequency hiss or for band limiting the ADF's output signal for telecommunications or subsequent digital recording.

To change the roll-off filter settings, move to the HPASS or LPASS field and adjust the cutoff frequency using the + or - key, respectively. HPASS frequencies are on ISO 1/3 octave centers, starting from 20 Hz. LPASS frequencies are on 1/6 octave centers, starting from 20 KHz. You can also choose the NONE option.

Once you are done editing the filters, you can either return to the main menu by pressing the ALT key or leave this screen on the display so you can continue to monitor the filter status. In ADF-2402 mono mode, you can use the A/B key to move from channel A to channel B. In ADF-2402 stereo mode, the channel indicator reads S (for total system control). The channel indicator is found in the lower left-hand corner of the screen (ADF-2402 only).

MENU ITEM 2: FILTER DIAGRAM

Main Menu Selection 2 brings up the filter display screen that gives a graphical representation of the notch and shelving filter settings. You may edit the filters here, as well as in the filter control menu described previously.



Filter Display Menu (sample)

The frequency response curve is calculated directly from the filter generation algorithms of the ADF and are accurate within the resolution of the LCD display. The display has three selectable scales available: 60 dB, 30 dB and 15 dB full scale. The scale selected is indicated in the upper legend of the LCD display. Use the +/- keys to scroll through the values. The zero dB reference point remains the same on each scale.

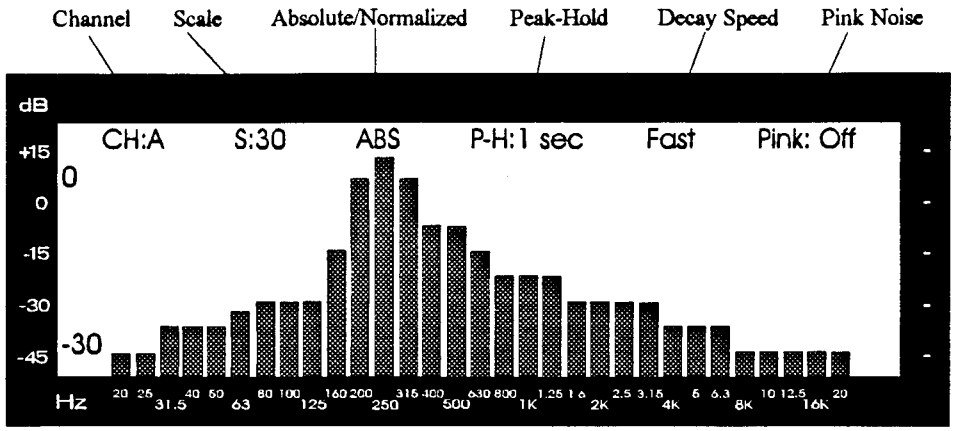
You can edit the parametric filters using the top row of fields, which match those in the filter control menu. The left and right arrow keys allow you to move from field to field, and the +/- keys are used to change the value of each field. The vertical dotted line cursor moves to the filter you are editing. You will be able to see and hear the changes you make as you move the filter; just place the standard field edit cursor in the frequency, depth or width field and scroll the values using the +/- keys.

Please note: Although only the +/- keys can be used to make changes in most fields in this filter display screen, a numeric value may be entered for the frequency. **You must enter a non-zero frequency before changing the depth and width.**

In ADF-2402 mono mode, you can use the A/B key to move from channel A to channel B. In ADF-2402 stereo mode, the channel indicator reads S (for total system control); **any change or filter set in channel A is also implemented in channel B.** The channel indicator is found in the upper right-hand corner of the screen (ADF-2402 only).

MENU ITEM 3: REAL-TIME ANALYZER

The Real-Time Analyzer (RTA) can be viewed by pressing the 3 key followed by ENTer from the main menu. The initial RTA display, as shown, indicates the power spectrum of the input to the selected channel of the ADF. The representation is in the 31 frequency band EQ format with each band centered on the respective ISO frequency.



Real-Time Analyzer (sample)

There are three scales available: 60 dB, 30 dB and 15 dB. They may be selected by pressing the +/- keys. The currently selected full scale is displayed in the upper left center of the LCD display. In all cases, the scale is linear in dB. Whereas the 60 dB full scale representation fills most of the screen, the 30 dB and 15 dB scales are far larger. A moving window has been implemented to allow for a more optimal display of portions of the 30 dB and 15 dB scales. Using the up/down arrows, you can move up-scale or down-scale as indicated by the full scale range at the upper left of the LCD display. The 30 dB or 15 dB ranges may be useful in an RTA and room equalization scheme where increased display resolution or sensitivity is required.

The absolute function displays the actual levels of signal power with respect to clip level; for example, 0dB of amplified pink noise may reach a level on the RTA screen of about -35dB because its energy is distributed across 31 frequency bands. Select ABS for analytical applications. Normalized values show the relative amplitude of signal power with respect to overall signal amplitude; the bandwidth of the highest peak is "normalized" to the signal amplitude at the moment. For example, 0dB of white noise will appear on the ADF RTA screen as 0dB. Use NRM for most applications.

Peak-hold displays a horizontal bar to momentarily hold the position of the peak value in each bin. Peak-hold times are adjustable from 1 second, 3 seconds, to infinite. Choose OFF to disengage peak-hold.

The display ballistics have been adjusted so that the attack is instant but the decay is slower. Two decay speeds are available: FAST and SLOW. They may be selected by the 4 or 5 keys. The currently selected speed is indicated in the upper right corner of the LCD display, just after the full scale range. The FAST decay mode would be used to monitor program material, if desired. The SLOW decay would be used in conjunction with a noise generator in an RTA and room equalization application.

You can use the A/B key to move from channel A to channel B. In ADF-2402 stereo mode, the channel indicator reads S (for total system control). The channel indicator is found in the upper left-hand corner of the screen (ADF-2402 only).

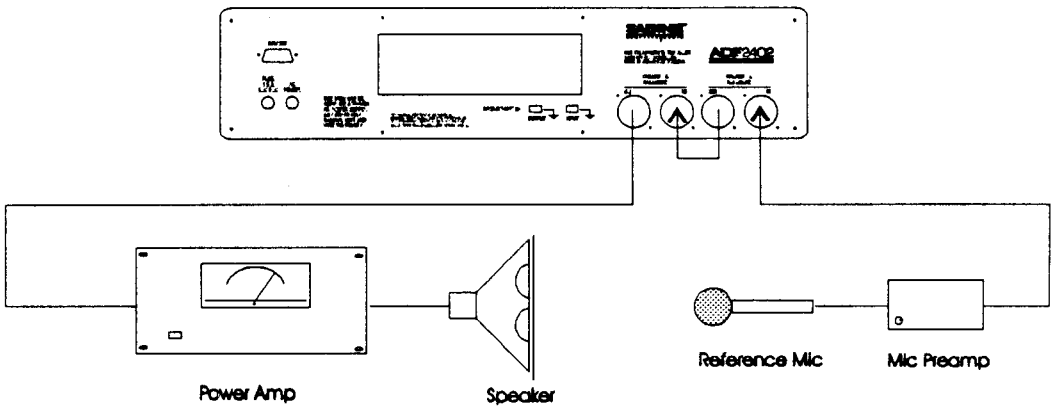
Please note: When you power down the ADF, all RTA settings are saved, and the same settings reappear upon power up.

BACKGROUND: The ADF's real-time analyzer (RTA) is used to measure the power content of the input signal in the FREQuency domain in real time before filtering. The RTA displays the signal power over 31 FREQuency bands between 20 Hz and 20 kHz. Each bar on the RTA display gives a measure of the relative signal power over a 1/3-octave interval centered at the standard 1/3-octave ISO center frequencies. The RTA operates in conjunction with the parametric filters, shelving filters, noise gate, digital delay, and FBX filters that have already been set.

The RTA is useful for visually analyzing the frequency content of the ADF's input signal. For example, a sinusoidal input will yield a peak in the RTA display in the band closest to the frequency of the input. Pink noise gives a display that is roughly flat with a height that is proportional to the pink noise power. The ADF allows you to choose A (for generating pink noise through channel A), B (for channel B), or OFF (to disengage the pink noise function).

The RTA is most often used for room set-up to bring the sound system to a state where the frequency response of the room is flat or optimal for a given performance venue. This can be accomplished by sending pink noise through the system, feeding a reference microphone input into the ADF and monitoring the RTA display. The objective is to bring the RTA display to the desired profile, which can be done by boosting or attenuating each frequency band, as appropriate. The boost or cut can be obtained by using either a graphic EQ or some of the parametric filters in the alternate channel of an ADF-2402 in mono mode. If you also wish to perform feedback elimination, be sure to leave some fixed or dynamic FBX filters enabled for the FBX mode of operation.

ADF-2402 RTA Setup:



Step 1: Configure your system as shown above.
Step 2: Adjust the power amp volume to minimum.
Step 3: Press ALT on the ADF to get to the main menu. Select menu item 3 (RTA); then press ENTer.
Step 4: Using the arrow keys, move the cursor to the pink noise field and press the + or - keys to select channel A; then press ENTer. Pink noise will now be output on CH A.
Step 5: With the CH A clip level control set to mid position, slowly raise the power amplifier volume until pink noise is heard through the speaker.
Step 6: Press the A/B button to display CH A's RTA, and adjust the reference mic preamp level to obtain an adequate reading on CH A's RTA.
Step 7: Adjust the CH A RTA mode, scale, peak hold, response time and window placement to most accurately display the incoming signal.
Step 8: Observe the CH A RTA to determine which frequencies need to be cut or boosted. Make those changes in the appropriate filters in the CH B filter edit screen.
Note: Exiting the RTA screen turns off the pink noise function.

MENU ITEM 4: DIGITAL DELAY

Main Menu Selection 4 takes you to the digital delay screen shown below. When you first enter the Digital Delay Menu, the cursor is in the FEET field, but you can move it to either the METER or MILLISECOND fields with the cursor keys. Key in the desired audio signal delay and press the ENTer key. Once the cursor is moved to the desired field, the + or - keys may be used to increment, decrement or slew the value up or down. The value is automatically entered when the + or - key is released.

DIGITAL DELAY MENU

A Digital Delay

Feet:	001.2	(1.2--192.3)
Meters:	000.4	(0.4--58.4)
Milliseconds:	001.1	(1.1--170.0)

Values shown are the factory default settings for ADF-2402

If you enter the value of the delay in terms of FEET, the equivalent values will be entered automatically in the METERS and MILLISECONDS fields. The same applies to METERS and MILLISECONDS. The ADF-1201's maximum delay is 340 milliseconds. The ADF-2402's maximum delay is 170 millisecond per side. The smallest incremental adjustment (resolution) is 0.1 milliseconds.

You can use the A/B key to move from channel A to channel B. The channel indicator is in the upper left-hand corner of the screen (ADF-2402 only). The delay can be set independently for each channel in both stereo and mono modes.

APPLICATION NOTE: It is not recommended to change the digital delay while live performance program material is present in the system in the presence of an audience unless it is well known that system tests or adjustments are taking place. Because of the nature of digital delay, any change in delay time causes the signal to shift instantly in time from one point in the signal to another. This may cause signal discontinuity in the form of a loud "pop" or click in the signal output. To minimize the problem, the ADF reduces signal level for a short time or until the transient time has passed. To avoid all transient phenomena, change the digital delay only when no program material is going through the system.

BACKGROUND: The ADF's digital delay facility can be used for many purposes, but perhaps most importantly for speaker signal alignment. Consider the case where an audio signal is emitted simultaneously from two speakers that are separated by some distance. The clarity and intelligibility of the program in the far sound field is greatly enhanced if the two signals arrive at the same time. In order for this to happen, the signal that emanates from the closer speaker must be delayed to compensate for its proximity to the listener.

The delay can be calibrated in terms of milliseconds, feet, or meters using the digital delay screen. For example, if the speakers were separated by 50 feet, the appropriate delay would be 50 feet. The conversion between milliseconds, feet, and meters is made automatically by the ADF. This conversion is based on the fact that the speed of sound in air at 20 degrees Centigrade temperature and 760 mm Hg atmospheric pressure is 344 meters per second. (Velocity increases about 0.61 meters per second per degree Centigrade between 0 degrees Centigrade and room temperature.) Since the speed of sound is a function of the air temperature and pressure, it is important that this be taken into account to precisely compute delay time. In this case, it is best to specify the delay in terms of milliseconds after you have computed it yourself as a function of the distance and the air temperature and pressure. For rough calibration, however, delay time specification in terms of distance should suffice.

MENU ITEM 5: NOISE GATE

Main Menu Selection 5 calls up the noise gate control screen. It allows you to specify the noise gate ATTACK TIME and the THRESHOLD level below which the noise gate should suppress the output signal. Move the cursor to the field you wish to edit with the ARROW keys and enter the desired numeric value. The changes take effect when you press the ENTer key after editing. The range of permissible values is in parentheses. Once the cursor is moved to the desired field, the + or - keys may be used to increment,

decrement or slew the value up or down. The value is automatically entered when the + or - key is released.

NOISE GATE MENU

```
A  Noise Gate

        Attack Time (msec): 999        (100-999)
             Threshold (db): -95       (1 --- 95)
```

Values shown are factory default settings

BACKGROUND: The purpose of the ADF's noise gate is to suppress low-level broadband noise, such as tape hiss between songs. Noise can arise from a number of different sources, including quantization error in the signal path of certain digital signal processors, cable noise and surface noise in magnetic media. The noise is most noticeable during the quiet times between program material — while there is no sound to mask the low-level noise.

There are two parameters that govern the behavior of the ADF's noise gate — THRESHOLD and ATTACK TIME. THRESHOLD is indicated in dB and tells the ADF the level below which the input should be regarded as noise. ATTACK TIME is indicated in milliseconds and tells the ADF how long the input signal must stay below the THRESHOLD level before the input signal should be regarded as noise. If the input signal level stays below the THRESHOLD setting for a period of time greater than the ATTACK TIME setting, the ADF turns off the D/A converter and the output signal is set to zero. The instant the input crosses above the noise gate THRESHOLD, the output is turned back on and the dwell time counter is reset.

The noise gate level is measured relative to the clip level of the analog-to-digital (A/D) converter inside the ADF. Recall that the clip level is determined by the position of the clip level adjust potentiometer. When this potentiometer is in the full counterclockwise position, the clip level is 29 dB peak above the level obtained with a 0 dBV RMS sine wave input. Therefore, with the noise gate level set to -80 dB and the clip level adjust set full counterclockwise, for instance, signals that are 80 dB below the clip level of the A/D converter or below -51 dBV peak will be suppressed.

MENU ITEM 6: PRESETS

Main Menu Selection 6 takes you to the PRESETS screen shown below. It allows you to name and save the current ADF parameter configuration or to load and run a configuration saved previously.

PRESETS MENU

```
Preset Table

  *  0. Default
     1.                          2.
     3.                          4.
     5.                          6.
     7.                          8.

              Save:0      <  >      Load:
```

Values shown are factory default settings

To load a PRESET configuration, move the cursor to the LOAD field using the cursor keys, and enter the number of the preset you wish to load. An asterisk (*) will appear next to the preset number that has been loaded.

To save the current configuration to a preset location, move the cursor to the SAVE field using the cursor keys, enter the number of the desired preset location, and press ENTer.

Each PRESET can be assigned a name of up to 8 alpha-numeric characters long. For example, you may pre-configure the ADF specifically for a warm-up band and the main performer during separate rehearsals and then quickly recall the configurations between sets of the performance. For example, you can name the configurations WARM UP and MAIN 1. Up to eight user-definable configurations can be stored.

To enter or edit a name, move the cursor to the position you wish to change with the ARROW keys. Change the values of the alpha-numeric characters at the cursor with the + and - keys.

The presets can also be used to compare the effects caused by changes in the system's configurations. For example, you may save a configuration in preset 7, then modify it and save the new configuration in preset 8. Now if you toggle between 7 and 8, you can hear the difference between the two configurations.

Factory Default Presets: Preset #0 is of special interest. It contains the factory default configurations and cannot be modified by the user. Preset #0 is a useful starting point for programming new configurations. It is also a useful reference if you cannot remember which settings are appropriate to use.

Power-up Default Setting: Each time the power is turned on, the ADF will automatically return to the same configuration it was in when the power was last turned off. You will not have to waste time recalling a configuration each time you start up the ADF. The first time you turn on the ADF, it will load and execute the factory default configuration.

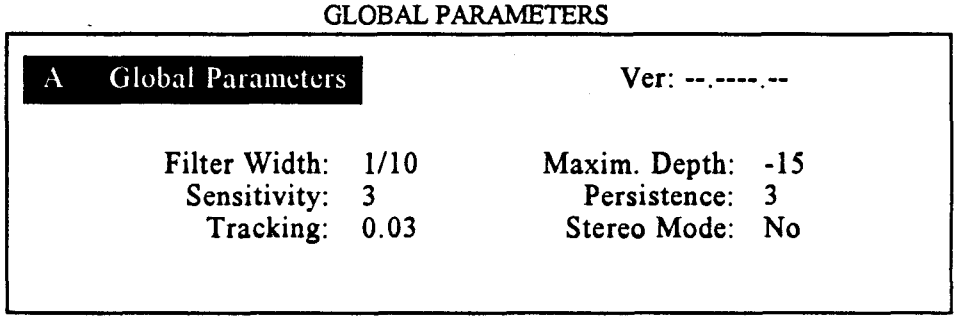
APPLICATION NOTE: It is not recommended to load PRESETS while live performance program material is present in the system in the presence of an audience unless it is well known that system tests or adjustments are taking place. Because PRESETS may cause dramatic changes to filter and delay settings, any change may cause a signal discontinuity which may be heard in the form of a "pop" or click in the signal output. To minimize the problem, the ADF reduces signal level for a short time or until the transient time has passed. To avoid all transient phenomena, load PRESETS only when no program material is going through the system.

MENU ITEM 7: GLOBAL PARAMETERS

Both the ADF-1201 and 2402 give you control over minute details of the system parameters so you can optimize the settings for a particular venue. These parameters pertain primarily to the feedback detection and tracking performance of the ADF. They also give added control to the fixed and dynamic FBX filter characteristics. Precise level of control has never been possible with conventional analog filters and is unique to Sabine's patented DSP-based feedback filtering system. Global parameters are saved as part of each preset. For example, each preset can have totally different FBX characteristics.

Main Menu Selection 7 invokes the GLOBAL PARAMETERS control screen shown below. Use the +/- keys to make changes to each selection. A description of each menu choice follows:

GLOBAL PARAMETERS



Values shown are factory default settings for ADF-2402

FILTER WIDTH: This selection affects only FBX filters, not parametric filters. In general, narrower filters are more transparent and mute the program less. Wider filters provide more gain before feedback and allow you to move the microphone a greater distance and still protect against feedback. A value of 1/10 octave is recommended for musical performances, and a value of 1/5 octave is recommended for lecture halls and other spoken word applications, such as teleconferencing. Permissible values: 1/1-1/20.

MAXIMUM DEPTH: determines the maximum depth to which the automatic FBX filters are allowed to grow in the presence of persistent feedback. For example, if only 8 dB additional gain before feedback is desired, it is not necessary to set the automatic MAXIM. DEPTH any lower than -16 dB. This setting will guarantee at least 8 dB cut at all frequencies above 50 Hz. Permissible values: 0 dB to -80 dB.

SENSITIVITY: determines the harmonic content of the suspected feedback signal before it can be classified as feedback. Used in conjunction with PERSISTENCE, SENSITIVITY discriminates between feedback, which tends to have low harmonic content, and music tones, which tend to have more harmonic content. Some musical instruments and singers are capable of producing tones which have very low harmonic content and are easily mistaken for feedback by the ADF. Higher SENSITIVITY settings indicate a higher sensitivity to feedback and require a lower harmonic content in the signal. Lower SENSITIVITY settings indicate a lower sensitivity to feedback and therefore require a higher harmonic content in the signal. Smaller values of SENSITIVITY will allow feedback to grow larger in magnitude before it is detected and eliminated; too high a value can result in mistaking certain musical tones for feedback. Use the DEFAULT setting of 3 for most venues; use the value 2 for more classical venues and 4 for spoken word applications. Permissible values: 1 (least sensitive) - 9 most sensitive).

PERSISTENCE: determines the relative length of time that a suspected feedback tone or signal must be present before it is classified as feedback and automatically suppressed. PERSISTENCE is used in conjunction with THRESHOLD to determine the authenticity of feedback and to discriminate real musical or non-feedback tones and signals from real feedback. Higher values of PERSISTENCE indicate a higher sensitivity to feedback and therefore require the suspected signal to be present for a shorter time. In a classical or slow venue wherein long sustained notes or tones are present, set the PERSISTENCE value low (2 or 1, for example) to minimize the chance of mistaking the long sustain for real feedback. Set PERSISTENCE to 4 for spoken word applications. Permissible values: 1 (longest persistence, least sensitive) - 5 (shortest persistence, most sensitive).

TRACKING: The TRACKING parameter controls how close FBX filters may be automatically set together. You may enter a value between .01 and .05 octave; this is the width of the window surrounding the current filter. If feedback occurs within your specified window, the ADF will "track" the feedback with the existing filter. The default setting is .03 octave.

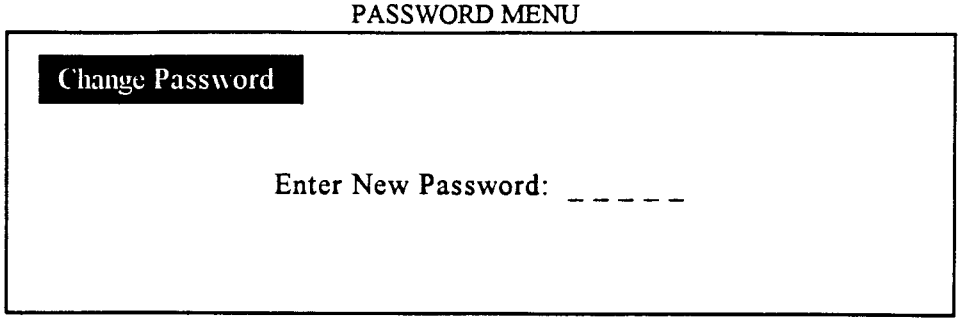
MODE: The ADF-2402 can be set to operate in either DUAL MONO mode or STEREO mode. In DUAL MONO mode, each channel is completely independent. In STEREO mode, a filter set in one channel is also automatically set in the other channel. There is only one FILTER EDIT screen when in the STEREO mode.

In ADF-2402 mono mode, use the A/B key to move from channel A to channel B. In ADF-2402 stereo mode, the channel indicator reads S (for total system control). The channel indicator is found in the upper left-hand corner of the screen (ADF-2402 only).

MENU ITEM 8: PASSWORD

The ADF password provides a mechanism for you to prevent unauthorized persons from accessing the MAIN MENU or sub-menus and changing the ADF's configurations. Enter MAIN MENU Selection 8 to move to the PASSWORD screen, shown below. Type in the new password and press ENTer. ALT takes you back to the MAIN MENU.

PASSWORD MENU



The ADF allows two levels of password control: a user level and system manager level. The system manager password is fixed as 13829 and cannot be changed. The system manager can initialize the system to default settings by pressing "9" and then ENTer in the main menu. **This procedure clears the unit's NVRAM and erases all presets.**

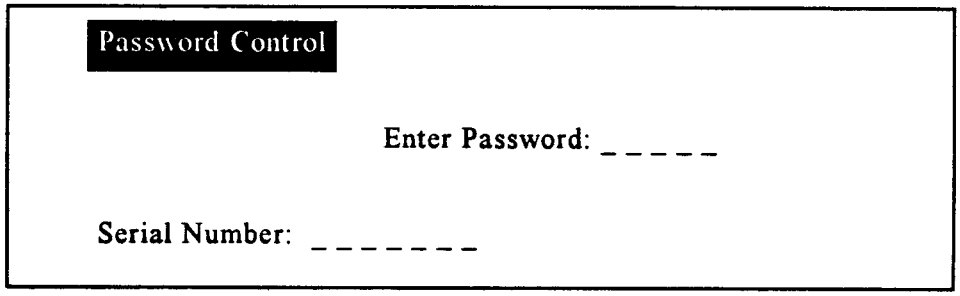
BACKGROUND: When the ADF is first powered up, the system automatically configures itself to the same settings as when the system was last powered down. If a password was previously assigned, the first ADF screen will be a request to input the password. Entering the correct password brings up the MAIN MENU and access to all sub-menus. If the current PASSWORD is the factory default password (00000), the password screen will be skipped and anyone can have access to the MAIN MENU and subsequent sub-menus.

The password must be 5 numbers long. Use the numeric keypad to change the number.

If the password 00000 is entered, the password protection feature is disabled until a new, valid password is entered in the CHANGE PASSWORD menu. You must turn off the ADF after entering a new password to initiate password protection.

IMPORTANT: From the main menu, entering 8 followed by ALT instead of ENTer causes the ENTER PASSWORD screen (same as at POWER ON) to appear, requesting password entry.

POWER UP/ENTER PASSWORD SCREEN



A FINAL WORD

There are millions of different combinations of user-selectable parameters in the ADF. As may be expected, some of these combinations can cause less than desirable side effects. So far, we have identified the following:

1. If several boost parametric filters are clustered together in the same frequency range, oscillation may occur.
2. Two boost filters close together add together, resulting in a net boost far greater than either filter alone. This will greatly reduce the headroom before clipping.

ENGINEERING SPECIFICATIONS

Filters

Twelve independent digital notch filters per channel which can be controlled automatically or parametrically from 20 Hz to 20 KHz

High pass filter with cutoff frequency, user-controllable in 1/3-octave intervals between 20 Hz and 1 KHz, 12 dB/ octave roll-off

Low pass filter with cutoff frequency, user-controllable in 1/6-octave intervals between 3.15 KHz and 20 KHz, 12 dB/octave roll-off

Notch filter depth: user-controllable in 1dB steps from +12 dB to -84 dB (parametric mode), 3 dB steps from 0 dB to -80 dB (FBX mode)

Filter width: user-controllable from 1.00 octave to 0.01 octave

Resolution: 1 Hz from 20 Hz to 20 KHz in FBX mode; 1 Hz from 20 Hz to 20 KHz in parametric mode

Time required to find and eliminate feedback: user-controllable from 0.1 seconds to 5 seconds (typically 0.3 seconds)

Number of parametric filters per channel: user-selectable, 0 - 12

Number of FBX fixed filters per channel: user-selectable, 0 - 12

Number of FBX dynamic filters per channel: user-selectable, 0 - 12

Total number of combined filters active per channel: user-selectable, 0 - 12; plus low and high pass shelving filters

Digital Delay

ADF-1201 — 340 msec total; ADF-2402 — 170 msec total per channel

Programmable in milliseconds, feet or meters.

0.1 msec resolution

Noise Gate

Attack time: 0 to 999 msec
Threshold: -1 to -95 dB (relative to peak amplitude)

Real-Time Analyzer

31 band, 20 Hz — 20 KHz

Password Configuration

5 alpha-numeric characters

Multiple Configurations Stored in Memory

8 user defined
1 factory default
1 most recent configuration (power down save)

LCD Display

User selection menus
Graphic filter placement screen
RTA display
Filter edit/status screen

Input/Output

Input impedance: Balanced > 10K Ohms, PIN 2 high
Output impedance: Balanced 10 Ohms nominal, PIN 2 high
Input/Output maximum signal levels: Balanced +29 dBV peak
Output load: >/- 600 Ohms balanced
Bypass: true power-off bypass
Headroom: +25 dBV peak @ 4 dBV nominal input
I/O connectors: XLR-3

Performance*

Spectral variation < 0.25 dB, 20 Hz to 20 KHz
SNR: > 100 dB typical (with Noise Gate)**
THD: < 0.02% @ 23 dBV at 1 KHz
Dynamic range: > 100 dB

Power Supply

50/60 Hz available in 100 V, 120 V, 220 V or 240 V; 22 W

Battery life: 10 years

Dimensions: 2-U rack mount 19 x 3.5 x 7.5 in. (48.3 x 9 x 18.3 cm); 9 lb. (3.9 Kg) nominal w/o power supply

Options

I/O Transformer isolation
RS 232 serial interface

Specifications subject to change without notice.

* Tests performed using an Audio Precision System One model 322 or equal

** Signal-to-noise ratio is the ratio of the maximum undistorted signal by specification (26 dBV RMS sinewave) to the noise floor

One-year limited warranty