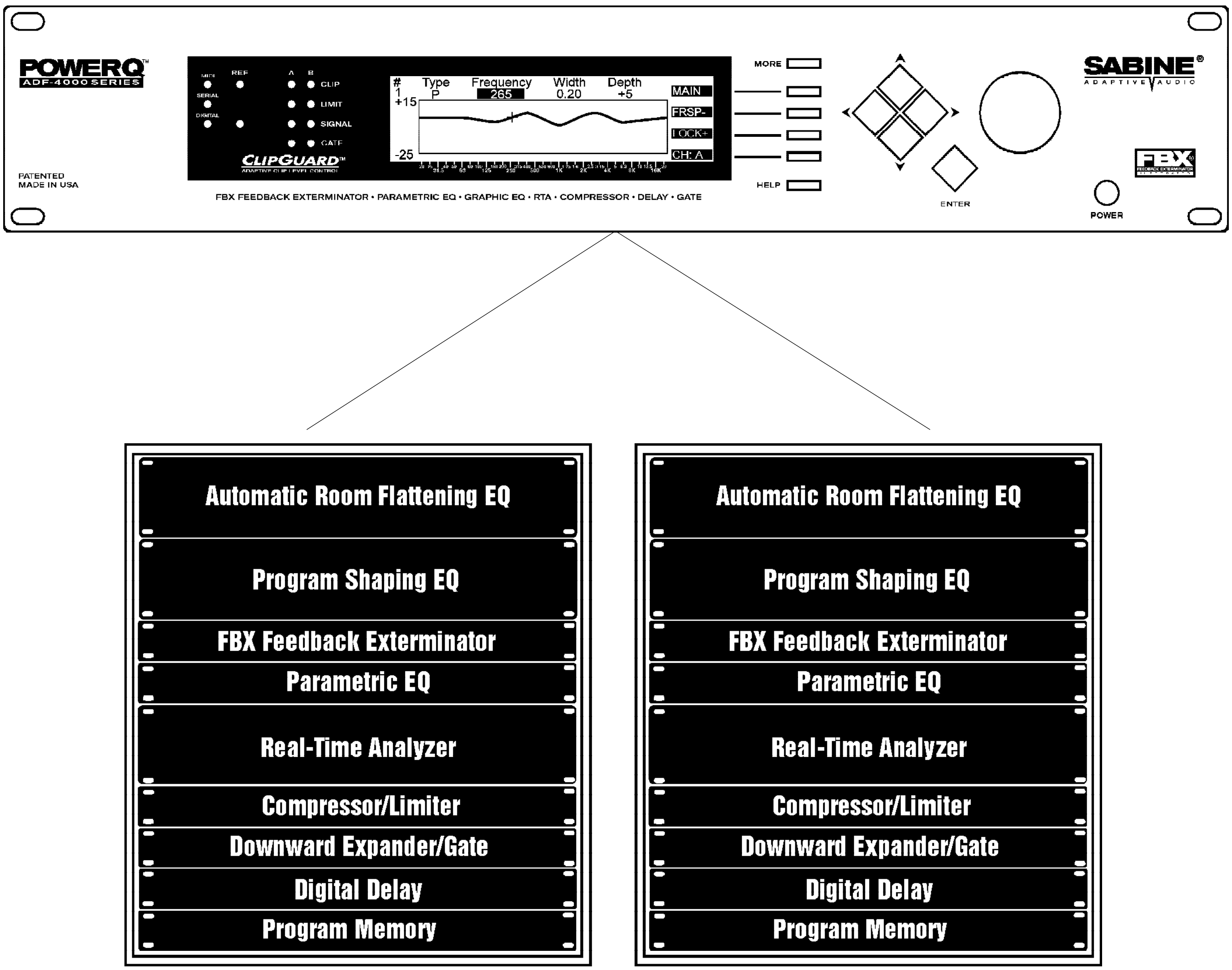
The two-channel POWER-Q gives you a full rack of gear -- all in one!



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POWQMAN4.V12/9/97jpm; (Software V2.30)

Special thanks to Hans Drobilitsch of Hans Drobilitsch Audio GmbH. (Austria) and Andreas Schneider (Chief Sound Engineer, Austria Center Vienna) for their valuable contributions to the design and implementation of the POWER-Q.

Section 1: Introduction

Congratulations and welcome to the new digital equalization and signal processing power of the Sabine POWER-Q ADF-4000, a whole rack's worth of power in a single 2-U unit. Simply insert the POWER-Q between the output of your mixer and the input to your crossover or power amp, and you're ready to harness the power of an arsenal of digital signal processing. (Note: You can also use the POWER-Q in other configurations. See Section 4.1.)

The POWER-Q is truly remarkable because of its multi-tasking ability and ease of operation. It's the latest breakthrough in the Sabine ADF Adaptive Digital Filter Workstation product line and offers many of the features of the Sabine REAL-Q2 Real-Time Adaptive Equalizer - all in one package.

The POWER-Q encompasses the following functions, all of which can operate concurrently:

- 2 channel 31-band digital graphic equalizer
- Up to 12 additional filters per channel, configurable as any combination of:
  - fully parametric filters
  - fixed or dynamic automatic feedback filters with Sabine's patented FBX technology
- High and low-pass filters for each channel
- 2 channel fully adjustable compressor/limiter
- Full featured filter-based RTA
- 2 channel digital delay for programmable time alignment of speakers by up to 80 mSec.
- 19 memory settings. Instant recall of all or selected parameters.
- Automatic Room EQ. Calibrate a system in any acoustic space to a flat response curve automatically in just a few minutes.
- ClipGuard™. The Sabine patent-pending adaptive clip level control automatically prevents digital clipping and expands dynamic range to over 110dB.
- Optional remote control via MIDI or RS-232 interface (for Windows™ computer)
- Optional Digital I/O: Allows user-selectable sample rate and input source.
- Optional Jensen isolation transformers

You may quickly refer to the sections you need by scanning for the appropriate icons:

This manual is written to provide background information for implementing the POWER-Q's features in appropriate situations. These sections of the manual are denoted with the "BACKGROUND" icon.

Sections of the manual pertaining to operating the POWER-Q are indicated by the "HANDS ON" icon.

Any information we think is essential is highlighted with an "IMPORTANT! READ THIS!" icon.

If you really can't wait to get started using your POWER-Q immediately, refer to our "quick start" section (section 5). We do recommend you read the manual for a fuller understanding and more complete utilization of the POWER-Q's features.

Section 2: Front & Back Panel Views & Controls

BACKGROUND

Fig. 1: POWER-Q Front Panel

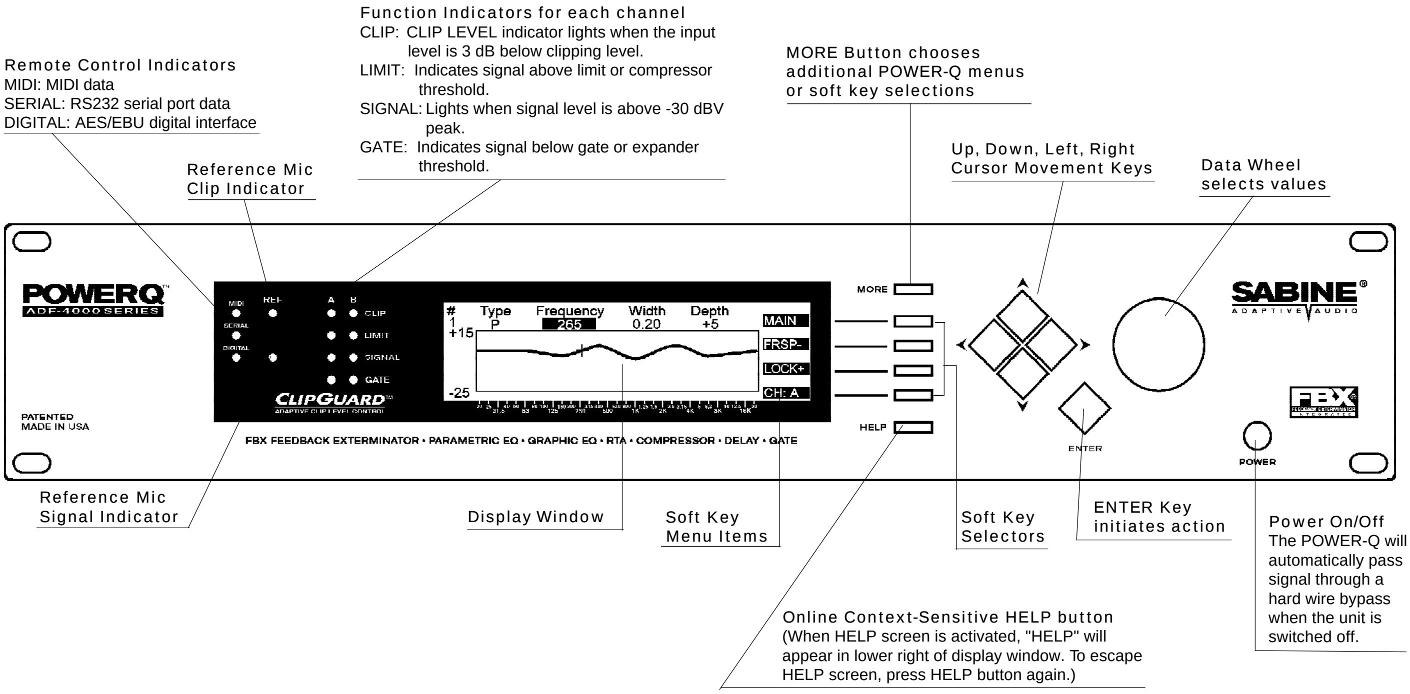
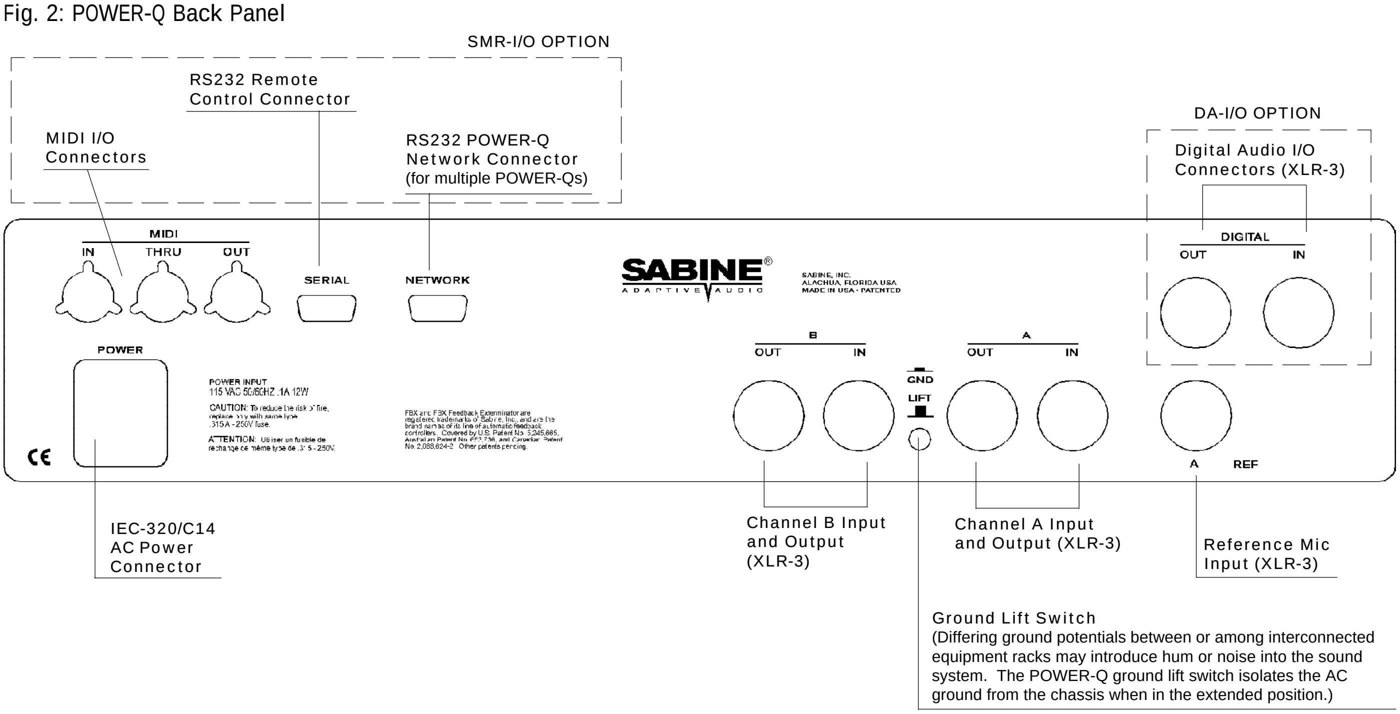


Fig. 2: POWER-Q Back Panel



2.1 Main Menu Windows (Quick Reference)

Fig. 3: Main Menu Items

MAIN MENU Page 1 of 4
1. Automatic Room EQ
2. Graphic EQ
3. FBX and Parametric Filters
4. Real-Time Analyzer
Copyright 1997 Sabine, Inc (MORE)

MAIN MENU Page 2 of 4
5. Compressor/Limiter
6. Expander/Noise Gate
7. Digital Delay
8. Stored Configurations
Copyright 1997 Sabine, Inc (MORE)

MAIN MENU Page 3 of 4
9. Global Parameters
10. Remote Control
11. Password
12. Digital I/O
Copyright 1997 Sabine, Inc (MORE)

MAIN MENU Page 4 of 4
13. About POWER-Q
Copyright 1997 Sabine, Inc (MORE)

Section 3: Block Diagram/Internal Signal Path

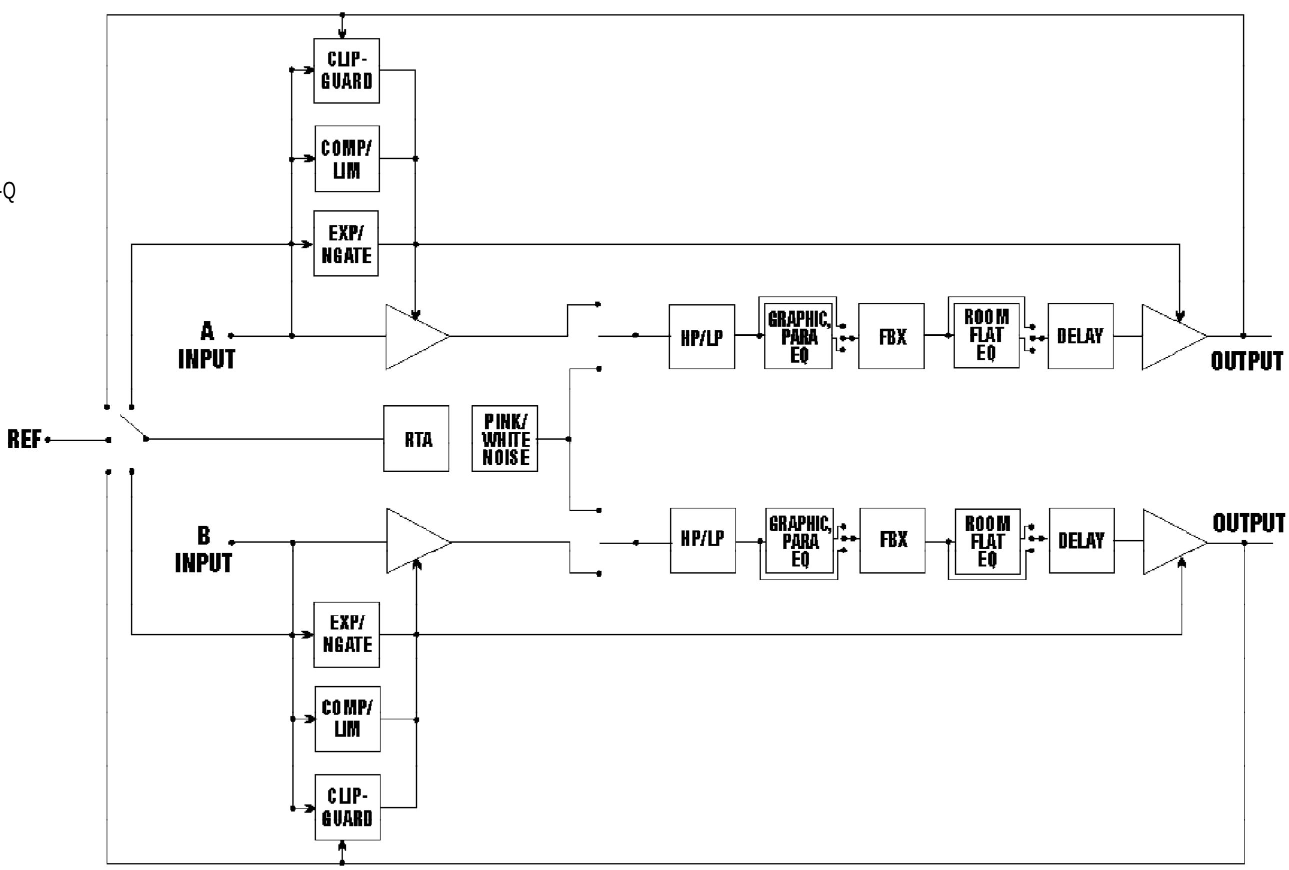


Fig. 4: POWER-Q Internal Signal Path

Section 4: Installation

4.1 WHERE TO INSTALL YOUR POWER-Q IN THE SOUND SYSTEM. The most common placement of the POWER-Q is between the output of a mixing console and the input to a power amplifier. If your system requires a crossover or additional delays (such as the Sabine DQX-206), put the POWER-Q in line after the mixer, but before those units. The configuration looks like this:

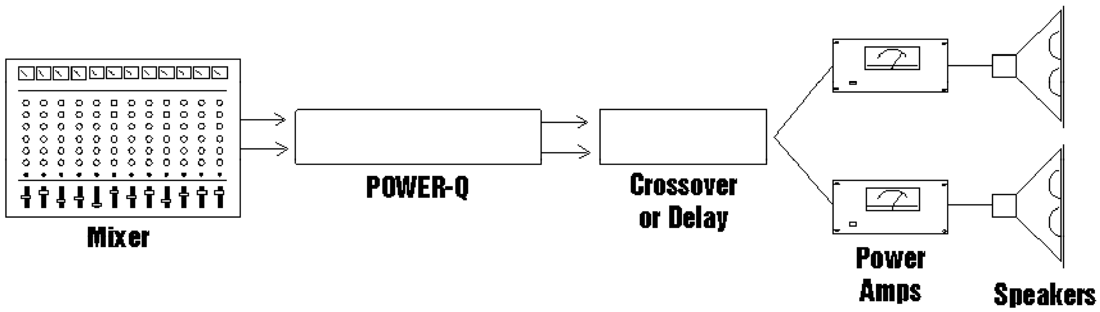


Fig. 5: Most common setup

A variation of this set-up might involve using the POWER-Q as a "dual mono" unit. Patch the main output of your mixer into Channel A of the POWER-Q, and the monitor output into Channel B. Then plug Channel A POWER-Q output into your main power amp, and Channel B POWER-Q output into your monitor power amp. This will allow use of the POWER-Q as if it were two separate mono units. The configuration looks like this:

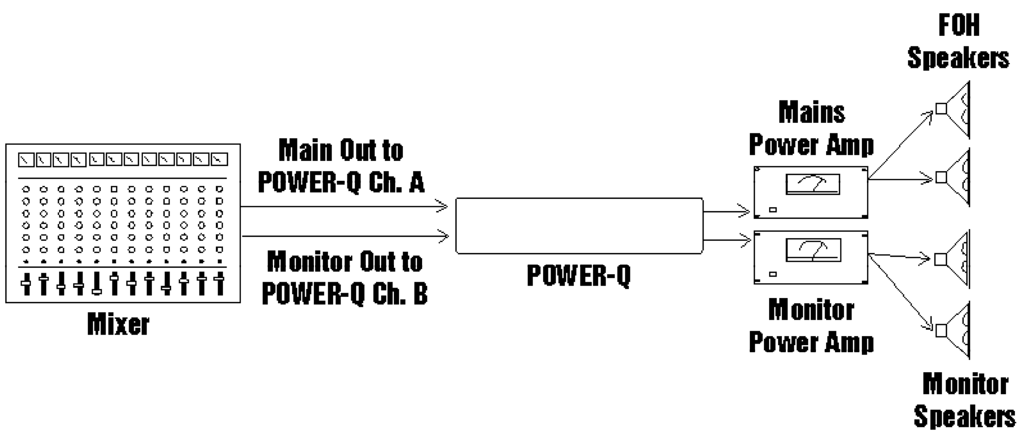


Fig. 6: Using the POWER-Q as a "dual mono" unit

The POWER-Q may also be used at a mixer insert point, either for a single input channel, or for a group or bus insert point. This will dedicate all of the features of the POWER-Q to a pair of single channel inputs on your mixer or to a sub-group of inputs (for example, all the drums in your mix). The patching will look like this:

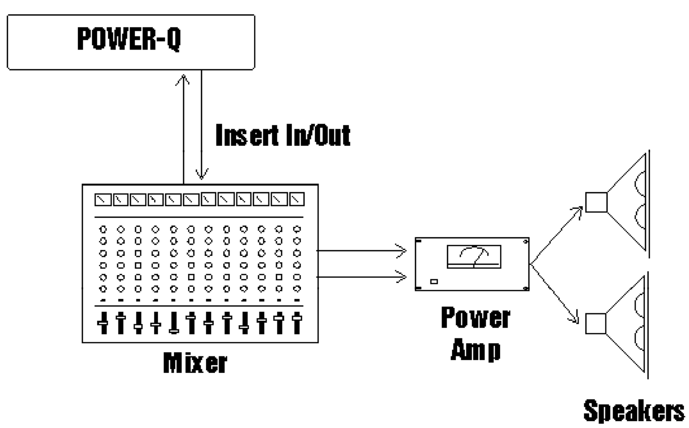


Fig. 7: Using the POWER-Q at a mixer insert point

The reference microphone for the POWER-Q plugs in the back of the unit, into the jack labeled “Ref A.”   The “Ref B” jack is blank.

THE POWER-Q SHOULD NOT BE USED IN THE FOLLOWING CONFIGURATIONS:

- Do not plug a microphone directly into the channel A or B XLR input connections on the back of the POWER-Q. These plugs are for balanced line level inputs, not microphones.
- Do not use the POWER-Q in an effects or auxiliary loop. Since such a patch is designed for mixing processed (wet) and unprocessed (dry) signals together in a variable proportion, the processing of the POWER-Q will be mixed with the unprocessed signal.

4.2 BYPASSING THE POWER-Q

The bypass function is on page 2 of the POWER-Q's Global Parameters screen. “BYPASS POWER-Q” routes the input signal directly to the output jack, completely bypassing the POWER-Q circuitry. This can be a useful feature when comparing “before” and “after” POWER-Q settings. Care must be taken in placing the unit in bypass after setting FBX filters, as the feedback you’ve been eliminating may pay you an instant unwelcome visit. Note that when the POWER-Q is in bypass, “BPASS” will flash in the upper right corner of the display in every screen.

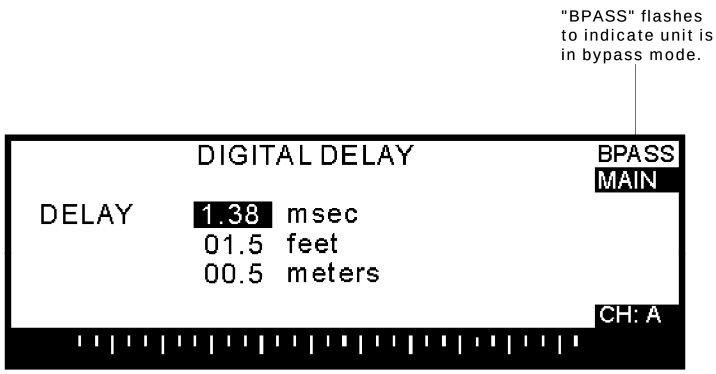


Fig. 8: Example of BYPASS indicator; shows in every screen

PLEASE NOTE: You can also bypass the various graphic EQ curves available in the POWER-Q. See section 16.3 for a complete explanation of this special bypass option.

Note also that turning off the POWER-Q or unplugging the unit from the wall will automatically route the input signal to the output as well, bypassing the circuitry and signal processing. When the POWER-Q is turned back on it will remember all of its settings at the time it was turned off.

Section 5: Quick Start-Up Reference

If you want to get going in a hurry, first make sure your POWER-Q is correctly patched into your system. There are two main tasks the POWER-Q will perform: equalizing the tonal balance of the system and eliminating feedback.

5.1 EQing AN ACOUSTIC ENVIRONMENT: QUICK INSTRUCTIONS. There are three basic methods of equalizing a sound system with the POWER-Q: Automatic Room EQ, RTA analysis, or by ear.

Automatic Room EQ. Make sure you have a reference mic positioned correctly and plugged into “Ref A” on the back of the POWER-Q (very important for optimal results). Select #1 (“AUTOMATIC ROOM EQ”) from the MAIN MENU. Follow the on-screen instructions. Your system will be analyzed and the EQ adjustments in the POWER-Q will be made in 5 minutes per channel.

Real Time Analysis. Alternatively, you may elect to play pink noise through your speakers and adjust the Graphic EQ to balance the frequency response. To do this, first make sure you have a reference mic positioned correctly and plugged into “Ref A” on the back of the POWER-Q. Then select #4 (“REAL-TIME ANALYSIS”) from the MAIN MENU, press the “MORE” button and turn on the pink noise independently for Channel A. Use the superimposed Graphic EQ sliders to make additional adjustments. Repeat the procedure for Channel B.

Equalizing By Ear. You won’t need to plug in a reference mic to EQ your system by ear. Select soft key #2 (“GRAPHIC EQ”) from the MAIN MENU. This will display the graphic EQ screen, one channel at a time (you may select which channel with a soft key). You may adjust individual filters using a combination of left/right arrow keys and the data wheel.

In addition to or instead of the Graphic EQ, you may effect system equalization with the POWER-Q parametric EQ. This is accessed by selecting MAIN MENU option #3, “FBX AND PARAMETRIC FILTERS,” and playing an audio source through your sound system and the POWER-Q. Twelve filters and a high and low pass filter are available for each channel. These filters are accessed using the up/down arrow keys and can be set to parametric (“PARAM”) using the data wheel. The data wheel and left/right arrows set the frequencies, filter widths, and filter depths of all filters. Any changes made to the parametric filters will add to the overall EQ of the system, including graphic EQ settings.

5.2 EXTERMINATING FEEDBACK: QUICK INSTRUCTIONS.

1. Once your room is equalized to your liking (or you may elect not to equalize and proceed directly to feedback control, though we don’t recommend this), set up your microphones and acoustic instruments in the positions where they will be used.
2. Set all controls for your sound system at the settings that will be used for performance. Select soft key #3 (“FBX AND PARAMETRIC FILTERS”) from the MAIN MENU. The POWER-Q defaults to seven fixed FBX, three dynamic FBX and two parametric filters, and has up to 12 total filters available for each channel. (This means that using parametric filters for room equalization by changing these defaults reduces the number of FBX filters available for feedback control.)
3. Slowly raise the gain of your microphones. As feedback begins to occur, the FBX filters will automatically detect the correct frequency of the feedback and apply a narrow filter to control it.
4. When all FBX-F (fixed) filters have detected feedback, press the “LOCK+” soft key to lock the filters (prevent them from changing or notching more deeply).
5. Do this procedure separately for both channels. (This is just one scenario for setting FBX filters. Please refer to Section 11 for additional details.)

Section 6: Overview & Philosophy

Live sound reinforcement can be a challenging business. Look what we have to deal with: The guitar player turns up to 11 and still complains that she can't hear herself. The podium speaker points the mic at his sternum and mumbles, drowned out by the chatter of people eating dinner, in a boxy hotel convention room. The rock singer asks for - no, DEMANDS - a monitor level loud enough to hear over a drag race in a hurricane. The minister clips on the lavalier mic and wanders around while preaching, sometimes right past the speaker cabinet...

As an antidote to premature aging and undue stress, we at Sabine have dedicated ourselves to simplifying the demands of live sound amplification by creating adaptive equipment that handles some of the tedious (but important) mixing chores automatically. This allows sound engineers to concentrate on making a mix sound good instead of dealing with acoustical problems!

The POWER-Q's features are designed to help you achieve two important goals in sound reinforcement: getting more gain before feedback and more clarity and definition in the sound. At the risk of sounding like a drill sergeant, let's call this the desire to be LOUD and CLEAR.

6.1 QUEST FOR LOUDNESS. At least two bad things happen in the pursuit of loudness (putting aside the deafness potential): feedback and lack of headroom.

Let's consider headroom first. The dynamic range of DSP is limited by the word length of an individual datum: The more bits in a word, the greater the dynamic range. The POWER-Q offers 24 bit resolution and a dynamic range spec of >110 dB (with ClipGuard™). Plus, our ClipGuard™ system is designed to make it all but impossible for our units to clip digitally; so if you're hearing distortion in your system, it's not likely coming from your POWER-Q, because we've taken steps to prevent that. (Note: Make sure the FBX TURBO set-up mode is off before your program begins; see section 11.2.) Likewise, the compressor/limiter built into the POWER-Q will raise the average gain level of your mix while protecting your speakers from hot shot sound engineers whose goal in life is to explore the extremes of speaker cone flexion. All of these functions are designed to maximize your gain without distortion.

Maximizing gain would be a far simpler matter if it weren't for the problem created by adding gain to a microphone in the presence of a speaker, which in turn reproduces the mic's sound, with the mic in turn amplifying its own amplification, and so on. At some point, at least one frequency will re-generate. This is techno-speak for that dreadful ringing sound commonly known as feedback. The nature and severity of the feedback will depend on the sound system and the acoustical environment, but feedback generally will occur before you reach the limits of your potential amplification. This means that the most likely volume limitation of your sound system is not the power of your amplifiers or the size of your speakers, but the occurrence of feedback.

Enter the Sabine FBX. In the pre-FBX dark ages, feedback was often controlled by passing a mix through a graphic equalizer and pulling out frequencies as close as possible to the ringing feedback. While this technique can reduce feedback, it also reduces the sound quality of the overall mix. The one octave-wide filters of a third-octave equalizer (you read that right—the filters are usually an octave wide, spaced on overlapping third-octave centers) are far too clumsy and inaccurate to target feedback specifically. You don't shoot a mosquito with a shotgun or do brain surgery with hockey gloves. Shotguns, hockey gloves and graphic equalizers are valuable tools, but only in the right applications. A graphic EQ is great to shape the overall sound of your mix (and that's why we've loaded your POWER-Q with a dandy one), but when you use it to control feedback by pulling down frequencies, you're also pulling out a big chunk of audio that is NOT feedback.

An FBX filter automatically detects feedback within a 1Hz resolution, slaps a tenth- octave wide filter on it, and pulls down the level only as far as necessary to get rid of the feedback at a given gain level. It is thus far more accurate in identifying and eliminating feedback, and far less destructive to your sound, than even the best graphic equalizer. Plus it finds the feedback automatically in a fraction of a second. You'd need to drink A LOT of coffee to react that quickly.

The POWER-Q provides up to 12 feedback filters per channel, thus allowing you to excel in your quest for loudness without compromising your second, equally important goal: QUEST FOR CLARITY.

6.2 QUEST FOR CLARITY. Clarity in the sound coming out of your speakers is a result of a myriad of considerations: the quality of the components you're using, the skill you and others demonstrate in setting up and operating the system, and the all-important acoustic properties of the room where you're operating your system.

Now, it's no secret that some architects skipped class the day of the 20 minute lecture on room acoustics, which is why so many rock concerts take place in basketball arenas and live sound people sport premature gray hair. Standing waves, flutter echoes, boominess...the list of problems is as big as a lead singer's ego. (Our congratulations to those architects who DO pay attention to room acoustics.)

The good news is that in addition to all the graphic, parametric, and FBX control in your POWER-Q, you are also the proud possessor of a full-blown Real-Time Analyzer. You can generate pink noise to help determine the frequency response curve of the surrounding acoustic space and compensate for its peaks and valleys with your EQ controls, while viewing a graphic display of the results in real time.

If you'd rather use your time to set up mics, patch in the rest of your gear or take a break, the POWER-Q Automatic Room EQ feature will analyze the room for you and optimize your system EQ to the response curve you specify. And the next time you come back to that venue or work with the same performer, you can punch up the memory of all your EQ settings and every other parameter set on the POWER-Q. You'll be done before the guitarist finishes tuning (unless he uses a Sabine tuner, in which case it might be a tie).

You can also use the POWER-Q for clearer sound by time aligning speaker stacks with our built-in digital delay. You can delay sound in one set of speakers by as much as 80 milliseconds (with 20 microsecond resolution) to allow sound to reach listeners' ears at the same time. This improves the phase consistency of the program, greatly enhances intelligibility, and synchronizes the sound origination directional cues from your ears to match the visual cues from your eyes.

So, in conclusion, the POWER-Q is your friendly rack of goodies conveniently condensed to a 2U box. It will automatically align your speakers, tune your system to any room, detect and eliminate feedback before and DURING performance, compress your mix bus, and remember your set-up for a given artist or venue. Sorry, it doesn't make coffee, but with everything the POWER-Q does for you, you'll have plenty of time to make it yourself.

Section 7: Optimizing The Sound System And The Room With The POWER-Q: Five Steps

Remember, our quest is to amplify sound in a room to a desirable level without creating feedback and distortion or sacrificing clarity. To make the most of a sound system in a particular acoustical environment, you will need to follow five simple steps:

1. Optimize the physical arrangement of your stage set-up, speaker placement, and room acoustics;
2. Time align your speaker stacks so that sound traveling from displaced speakers (and from sound sources on stage) arrives at a designated reference position at the same time or provides audio cues consistent with visuals;
3. “Flatten” the frequency response of your sound system in the acoustical environment so all frequencies are heard in equal proportion at the reference position;
4. Adjust the equalization of the system to your personal preference or the requirements of a particular application or performer (the POWER-Q will remember these settings and load them from memory);
5. Apply FBX filters to live microphones to increase gain before feedback and insure maximum clarity, volume and microphone mobility.

The POWER-Q is amazingly useful for realizing steps 2 through 5, especially the last three (see sections 9, 10, 11, 12 and 13). Here are some suggestions for implementing all five steps.

7.1 STEP ONE: THE PHYSICAL SPACE. Unfortunately, the POWER-Q cannot physically rearrange your stage set-up or dampen reflective surfaces in your room. You may not be able to build a bass trap in a boomy room, find enough stage space to set the front-of-house speaker cabinets far enough in front of the mic line to avoid howling feedback, or convince a night club owner to carpet the dance floor. Ideally, a room with non-parallel, non-reflective surfaces that is large enough to accommodate a full wave length (30 feet+) low bass frequency will provide you with fewer resonance points, a more evenly balanced room curve and less feedback. This acoustic ideal is seldom found in the real world, so you should make the best of the situation with careful speaker and microphone placement. To go beyond the limitations inherent in a less-than-desirable acoustical space, you’ll have to call in the artillery (electronics and equalization) to optimize your sound system. This is why a device like the POWER-Q is worth every cent (and then some) of its very reasonable price.

7.2 STEP TWO: TIME ALIGNMENT OF SPEAKERS. Compared to light or electronic signals, sound travels very slowly. Sound traveling across a night club or concert hall, or from speakers at the front of the stage to speakers half way to the back of the hall, is slow enough to often warrant speaker time alignment with a digital delay. The sound emanating from a speaker farther away from the stage is delayed relative to speakers close to the stage. The sound traveling to the listeners’ ears from stage speakers takes longer to get there than the sound from the closer speakers because most of the path is electrical. The correction is designed to allow the sound from both sets of speakers to arrive at the listening position at the same time. (Obviously, in a situation where there is only a left and right front of house speaker stack, time alignment may be less of an issue. See Section 8.1 for a full discussion of these issues.)

The POWER-Q allows you to delay each output by up to 83.2 milliseconds. The delay time for each channel can be set independently. For more details about setting the delay in your POWER-Q output, refer to section 8.2. For more demanding delay applications involving up to six separate outputs, automatic calculation of delay times and air temperature compensation, use the Sabine DQX-206 delay/equalizer/limiter.

7.3 STEP THREE: SETTING THE SYSTEM AND THE ROOM TO A “FLAT” RESPONSE CURVE. Once the system is in place and the speakers are time aligned, you are ready to even out (make equal, or “equalize”) the frequency response of the system in the room. This typically is done with broad filters, such as the octave-wide filters of a 31-band graphic equalizer. Graphic EQ filters are spaced on third-octave centers, but are typically an octave wide, overlapping across adjacent filter controls. You can vary the width of the POWER-Q's filters by accessing the "GLOBAL PARAMETERS" option on the MAIN MENU (see Section 17).

The POWER-Q excels at the task of room equalization, offering both automatic and manual frequency response adjustment. In the Automatic Room EQ mode (see section 9), a reference microphone (with a flat frequency response) is placed at your choice of listening positions in the acoustical environment. The POWER-Q automatically plays a series of tones (sine waves across the whole audible spectrum). The energy of the tones as heard at the reference mic results in automatic adjustments of the POWER-Q's 31-band graphic equalizer, producing equal energy at all frequencies. (If you want to specify a curve, load it on top of the flat curve after the Auto Room EQ procedure is finished.) The whole process takes less than 6 minutes per channel.

Alternatively, you may choose to equalize your system by playing pink noise over the loudspeakers and observing the REAL TIME ANALYSIS of the propagated energy across frequencies as heard by the reference microphone (see section 13). You must then manually adjust the equalization faders of the POWER-Q's graphic equalizer section to produce the desired frequency response. Note that the POWER-Q allows you to see both your equalization faders and the RTA response in the same window, which spares you the cumbersome task of scrolling from one window to another while you make adjustments.

7.4 STEP FOUR: TWEAKING THE EQ. Once the room is flat, you may want to customize the room equalization to meet your own personal tastes or to match the entertainers' performance style. You may do this either by making further adjustments of the graphic equalizer (see Section 10) or by inserting up to 12 parametric filters per channel for very precise adjustment (see Section 11). These filters can be added in list (tabular) form or drawn as a response curve using the data wheel. Once you have room and system tweaked to your ideal, you can save and name your setting for future quick recall (Section 16).

7.5 STEP FIVE: FBX FILTERS. In live sound reinforcement, the true limitation for system loudness is not usually the wattage of the amplifiers, the headroom of the mixer or the power handling maximum of the speakers. Before the system clips, you will almost certainly encounter feedback. And for eliminating feedback, there is no better system than the Sabine FBX.

Once the sound system is properly equalized, the narrow filters of the FBX will go a long way towards increasing loudness without feedback. Placing the FBX filters in line WITHOUT first using a graphic equalizer may make several narrow FBX filters cluster together closely at a point where the system and the room have a strong resonance. This means you will quickly exhaust available FBX filters trying to do a job better suited to a wider filter (i.e., a graphic EQ fader), you'll reduce the amount of potential increased loudness, and you won't get the maximum benefit of FBX.

The POWER-Q is the most complete system on the market that allows this much control over the steps to maximize clarity and gain in any acoustical environment, using any sound system.

Section 8: Using the POWER-Q Digital Delay

8.1 Digital Delay Applications and Use. This section goes beyond the typical operating guide that only explains the front and back panel adjustments of a piece of equipment. Instead, we discuss the basic acoustical concepts needed to get the most out of the use of digital delay in sound systems. If you are familiar with these principles, feel free to skip ahead. Some principles may require additional delay channels and options available with the Sabine DQX-206.

Why Digital Delays? The most intelligible sound occurs when two people speak face to face. The sound is loud and dry, and the direction of the sound aligns with the speaker. The most intelligible sound systems are the ones that come closest to emulating face to face communication. If this is your goal, a digital delay is essential to your sound system.

There are three distinct applications for digital delays. The first and most important is **synchronization of the loudspeakers** to control excess reverberation and echo. Second, digital delays help **control comb filter distortion,** and finally, digital delays are useful for **aligning the acoustic image** so the direction of the sound seems to be coming from the performer rather than from the loudspeaker.

Loudspeaker Synchronization

Sound travels at about 1,130 feet per second in air, or about 1 millisecond per foot. On the other hand, electronic signals travel almost one million times faster through your sound system to the loudspeakers. The main task for digital delays is to synchronize multiple loudspeakers so the sound traveling different distances arrives at the listener's ears at about the same time. Synchronizing the loudspeakers reduces reverberation and echoes for improved intelligibility.

How to Synchronize Your Signals

There are several powerful tools available for precisely measuring the time a loudspeaker signal takes to arrive at a certain point in the audience. Most of these tools are very sophisticated and tend to be quite expensive. Fortunately, simpler tools are sufficient for most applications.

In the 1930's, engineers synchronized the low and high frequency speakers in movie theaters by feeding a sharp click through the system. They moved the speakers until they could only hear a single sharp click coming from both speakers. You can use this same method with a common child's toy called a clicker. Pressing the thin metal strip makes a loud sharp click. A clicker is especially useful when synchronizing the direct sound from the performer with the sound from the loudspeakers.

Alternatively, you can use a phase checker especially for synchronizing the signals of two loudspeakers (either LF and HF or two full range systems) since most of the phase checkers include a click generator and receiver. Phase checkers are quite affordable and have other uses besides synchronization.

Processing (or Group) Delays

Converting signals back and forth from the analog to digital domain always delays the signal a little. These conversion delays are often called processing (or group) delays, and usually range between 0.9 and 5 milliseconds. You will notice that Sabine delays always display the processing delay as the smallest possible delay value. For the POWER-Q, the processing delay is 1.38 milliseconds. You can bypass the unit for 0 seconds delay.

Not all manufacturers acknowledge processing delays in their specifications, but you must take them into account when synchronizing your system. Make sure all digital equipment is on and not bypassed when synchronizing. Also, be careful to make an appropriate adjustment in your delay lines if you later add any type of digital equipment to the system.

Center Cluster Speakers

Center cluster speakers offer several advantages over systems that have speakers mounted on the sides. The most obvious advantage is that the distance to the closest and most distant locations in the audience is almost equal, so most listeners hear about the same level. Center clusters also offer two other advantages regarding visual imaging.

Studies have shown that people can detect even small horizontal changes in the direction of a sound source, but vertical shifts are much less noticeable. This suggests that the sound from center-cluster speakers is more likely to be visually aligned with the performer than loudspeakers placed on each side of the stage.

All those in the audience who are closer to the performer than the center cluster will hear the direct sound from the performer before they hear the sound from the loudspeakers. This makes the sound seem to come from the performer, not the loudspeakers. (See the Precedence Effect below.)

Comb Filter Distortion

Many who took high school science may remember ripple tank experiments where waves are generated from two separate point sources. The waves from each source combine to form visible interference patterns. In some places the wave crests and troughs are in phase so they combined to make a larger wave. In other places the crests are out of phase, so the crest of one wave source is canceled by the trough of the other. Ripple tank experiments show the interference patterns are strongest when the amplitudes of the waves from each source are equal.

A similar interference occurs in sound systems when a signal is delayed and mixed back into the original signal. These interference patterns are called COMB FILTERS because their frequency response plots look like the teeth of a comb (see Figs. 9 & 10). There are a number of common situations that cause comb filters. For example, when the program is played through two loudspeakers, the loudspeaker that is farther away interferes with the closer loudspeaker. Comb filters are also created when a performer is picked up by two microphones, one closer than the other. You even introduce comb filters by mixing digital effects back into the “dry” signal at the mixer’s effects loop.

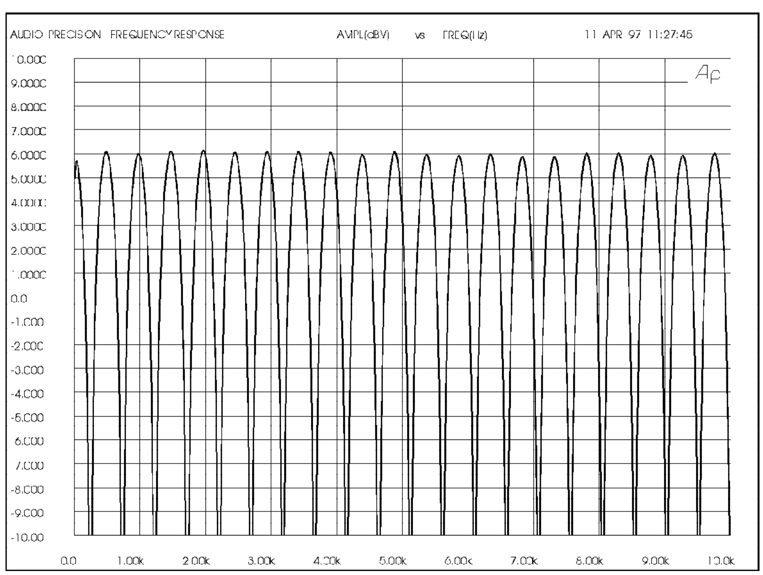


Fig. 9: COMB FILTERS. Input signal mixed with a 2 msec. delayed signal. (Both signals have the same amplitude. Max. filter gain is +6dB, and max. depth is -4.)

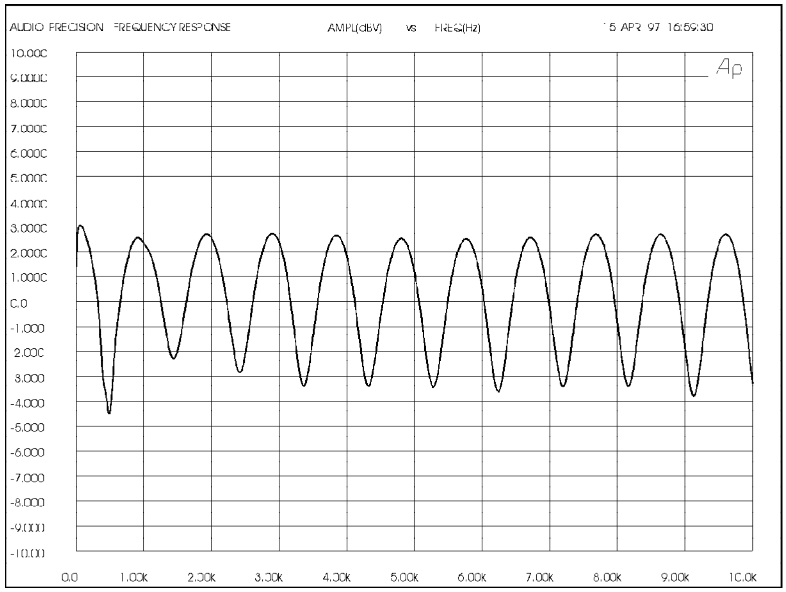


Fig. 10: COMB FILTERS. Input signal mixed with a 2 msec. delayed signal. (Delayed signal has 10dB less amplitude. Max. filter gain is +2.5dB, and max. depth is -3.) Reducing the amplitude of the delayed signal reduces the comb filters' effect.

Calculating Comb Filter Frequencies

The frequencies of the reinforcements and cancellations depend on the delay time (the time difference between the arrival time of the original signal and the delayed signal). The frequency of the first cancellation occurs at 1/(2t) Hz, where t = the delay time in seconds. The cancellations are separated by (1/ t) Hz. Fig. 11 shows how the comb filters change with the delay time.

Delay time = 0.002 sec.		Delay time = 0.003 sec.		Delay time = 0.004 sec.	
Cancellation Freq. (Hz)	Reinforcement Freq. (Hz)	Cancellation Freq. (Hz)	Reinforcement Freq. (Hz)	Cancellation Freq. (Hz)	Reinforcement Freq. (Hz)
250	500	167	333	125	250
750	1000	500	667	375	500
1250	1500	833	1000	625	750
1750	2000	1167	1333	875	1000
2250	2500	1500	1667	1125	1250
2750	3000	1833	2000	1375	1500
3250	3500	2167	2333	1625	1750
3750	4000	2500	2667	1875	2000
4250	4500	2833	3000	2125	2250

Fig. 11: Comb filters get closer as delay time increases.

Comb Filter Amplitude

If the original signal and the delayed signal are the same amplitude, the reinforced frequencies increase in amplitude by 6 dB, while the out-of-phase frequencies cancel completely to -4 dB.

Comb filters cause a lot of problems. The frequencies that are reinforced are prone to excite feedback, while the out-of-phase cancellations make the program sound thin and over equalized.

Try this simple experiment to hear what comb filters do to your sound.

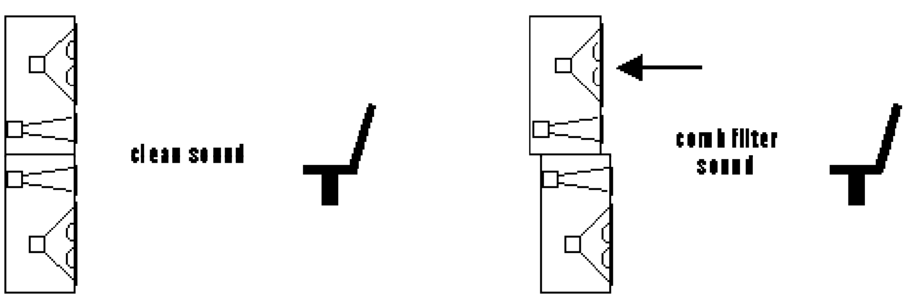


Fig. 12: Comb filters noticeably affect your sound.

Stack two identical full-range loudspeakers as shown in Fig. 12. Carefully align the HF horns and wire the speakers in mono. Stand in front while listening to your favorite full-spectrum CD. Ask a friend to move the top speaker slowly away from you. The degradation in sound quality you hear is caused by comb filters. The experiment is most dramatic when you use good quality speakers.

Correcting Comb Filters

Comb filters are inevitable to some degree in every live sound system, and they cannot be corrected with equalization. Fortunately, most comb filter problems can be reduced to a minimum by synchronizing the signals and reducing the amplitude of the delayed signal. The examples below show several practical applications.

The Precedence Effect: Aligning the Acoustic Image

Helmut Haas published a study in 1951 describing a series of experiments that demonstrated how people perceive delayed signals and echoes. In his experiments, a listener was positioned between two speakers placed 3 meters away; one was placed 45 degrees to the right and the other was placed 45 degrees to the left. When the same program was played through both speakers simultaneously, the listener perceived the acoustic image (the direction from which the sound seemed to be coming) centered between the speakers.

When Haas delayed the signal going to one of the speakers by somewhere between 5 to 35 milliseconds, the listener perceived a shift in the acoustic image to the speaker heard first. While the delayed speaker did not contribute to the apparent direction of the sound, it did make the program seem louder and “fuller.”

Haas showed that you must increase the loudness of the delayed signal by about 8 to 10 dB (twice the perceived loudness) in order for the acoustic image to move back to the original center position. Increasing the loudness more than this, or increasing the delay somewhat more than 35 milliseconds, makes the delayed signal sound like an echo.

The phenomenon describing how the acoustic image follows the signal we hear first is called the Precedence Effect. The phenomenon that makes two distinct sounds heard less than 35 msec. apart seem like only one sound is call the Haas Effect. However, the terms are often used interchangeably in the sound industry.

THREE APPLICATIONS FOR DIGITAL DELAYS

APPLICATION I: Under-The-Balcony Speakers

Fig. 13: Overhead view of under-balcony application.

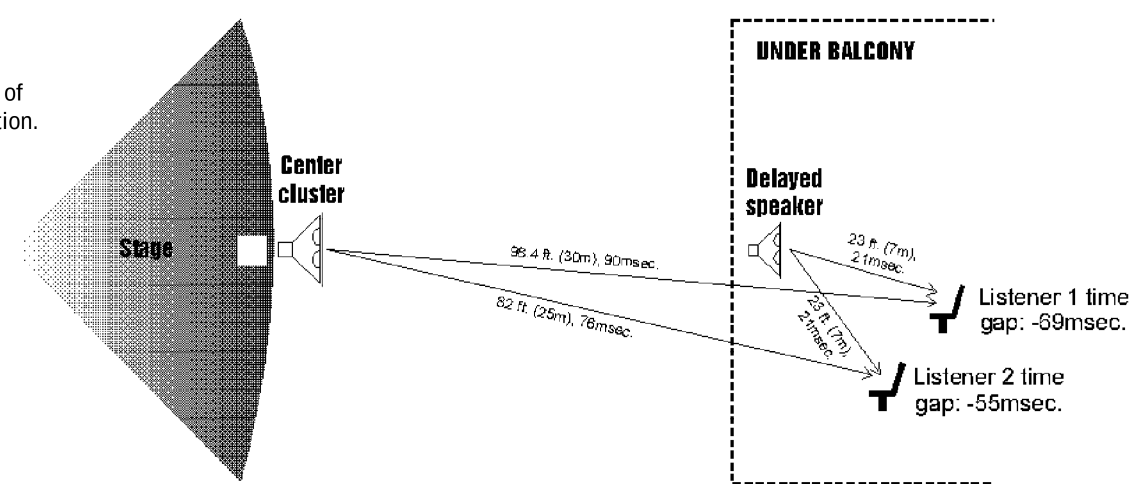


Fig. 13 shows a typical situation where the performer is amplified by a center cluster hanging above the stage. Almost everybody in the audience will enjoy good sound, except those seated in the shadow of the balcony. So we add an under-balcony speaker to fill in the shadow.

Now we have sufficient volume under the balcony, but the sound from the two speakers arrives at the listener's ears some 55 to 69 milliseconds apart. The two signals, along with their echoes, result in an unintelligible cacophony. We must delay the sound from the under-balcony speaker to synchronize the signals. Do we set the POWER-Q delay to 55 or 69 milliseconds? Obviously, the geometry will not allow us to exactly synchronize every location under the balcony; we have to compromise.

First, consider the program type. For spoken word programs, you will produce the best intelligibility if the signals from the under-balcony speakers arrive within 10 msec. of the signals from the center cluster. Therefore we should set the delay to 65-69 msec. You can allow a little more reverberation for programs that are mostly music.

Next, we must eliminate comb filter distortion. Find the axis where the levels of the center cluster and under-balcony speaker are equal. (See "Comb Filter Distortion," p.15.) Use the POWER-Q to precisely synchronize the speakers along this axis to eliminate the most severe comb filters. Comb filters off the equal-level axis are much less of a problem since a louder signal is not affected very much by a weaker signal.

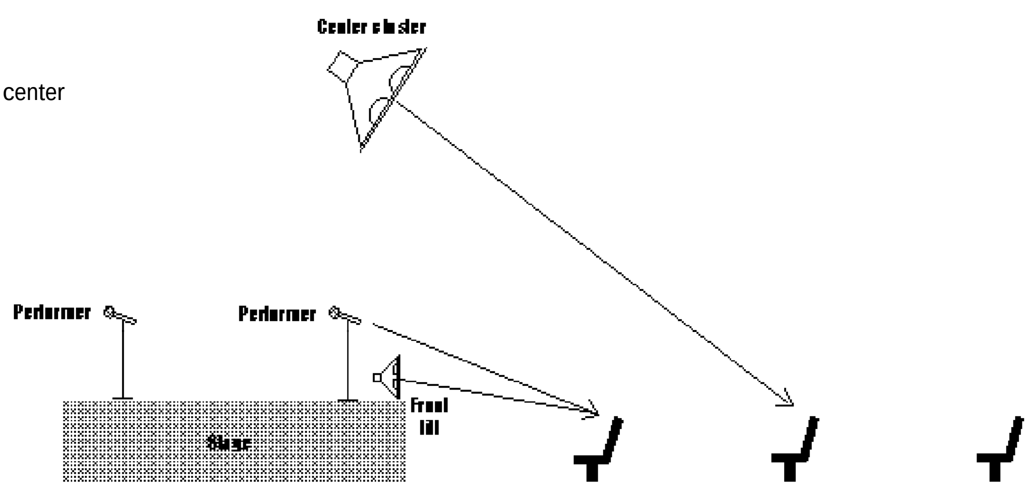
Finally, you can experiment with adding 5 to 10 milliseconds delay to both sets of speakers to enhance the Precedence Effect for the audience seated near the performer.

In the final analysis, every setting is a compromise, and your ear has to be the final judge. Check the sound in several different locations throughout the auditorium and correct the most severe irregularities.

Application II: Center Cluster with Front Fills

Fig. 14 below describes a typical application that has a stage with a microphone, a center cluster above the stage, and front fills in front of the stage. There must be thousands of installations throughout the world like this that "get by" without digital delays. But with the POWER-Q, you can improve the intelligibility and add a new quality without ringing up any significant costs. Use the POWER-Q in this situation to align the visual image with the acoustic image. The program is much more enjoyable when the amplified sound seems to be originating with the performer, not the loudspeakers.

Fig. 14: Synchronizing center clusters and front fills.



Find a central place in the audience where the center cluster is 6 to 8 dB louder than the direct sound from the performer. Delay them so that their sound arrives 5 to 8 milliseconds after the direct sound from the performer. Experiment by bypassing the POWER-Q in and out to hear how the source of the sound seems to move from the loudspeakers to the performer and back. Now your ears have the same directional information as your eyes, so the performance will sound more natural and exciting. The best seats in the house just got better.

What about the front fills? Their purpose is to add intelligibility and listening comfort to the first few rows nearest the stage by filling in the areas missed by the center clusters. Add about 8 msec. to the front fills to take advantage of the Precedence Effect.

The 8 msec. setting presumes the performer is standing on the front few feet of the stage. But some stages are well over 30 feet deep. What if there is a second performer standing 25 feet behind the first? The direct sound from his or her voice will reach the first few rows about 25 msec. after the first performer's. The audience will perceive the first performer directly and the second performer through the loudspeakers.

We can add the advantage of the Precedence Effect to the second performer by placing the POWER-Q in the mixer's channel insert point and adding a 25 msec. delay.

Certainly taking advantage of the Precedence Effect is not as obvious to the audience as eliminating feedback, but it is nice to know you did all that is possible to make the performance enjoyable.

Application III: Synchronizing the signals of a far-throw and short-throw loudspeaker.

In order to reach the proper coverage in larger venues, we often stack two full range speakers - a short-throw center cluster for the audience below and a far-throw speaker for the back of the auditorium. It is almost impossible to perfectly align the stacked speakers mechanically, so comb filter distortion becomes a problem in the area where the levels from both speakers are equal. The same thing happens with speakers mounted on the right and left sides.

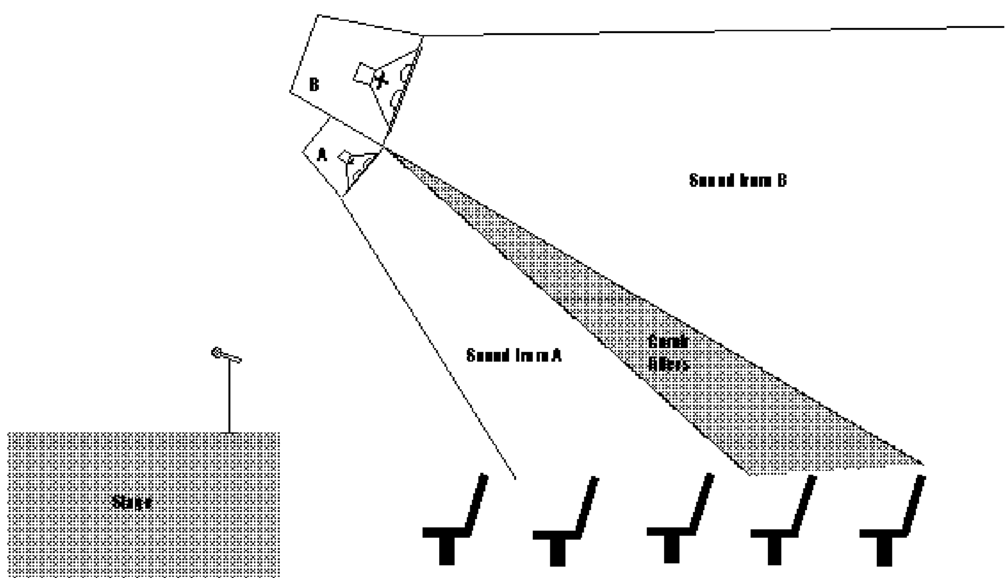


Fig. 15: Aligning far- and short-throw speakers. (The level from both speakers is equal.)

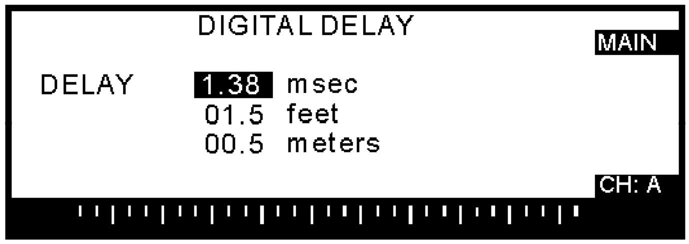
It is impossible to remove comb filters with equalization, but the POWER-Q eliminates them in short order without affecting the spectral balance for the rest of the audience. Find the axis where the levels from the two speakers are equal. This is where the comb filters are most severe. Carefully adjust the POWER-Q so that the signal from both speakers arrives at precisely the same time. The POWER-Q provides 20 microsecond resolution for this purpose.

Use the same procedure to align speakers within a cluster when necessary.

8.2 POWER-Q DIGITAL DELAY ADJUSTMENTS.

To access the POWER-Q digital delay controls, choose #7 (“DIGITAL DELAY”) from the MAIN MENU. Use the up and down arrow keys to scroll the MAIN MENU screen.

Fig. 16: Digital delay window



You may set the delay time for channels A and B independently. Delay time can be set in milliseconds, feet, or meters, with a resolution of 20 microseconds. Adjusting any one delay parameter automatically changes the corresponding read-out in the unselected measurement units. The time unit display will always be the most accurate. (The distance displays are approximations based on the speed of sound at standard temperature and pressure conditions: 1127 feet/second at 20 degrees C and 760 mm Hg atmospheric pressure.) The minimum delay time allowable is 1.38 milliseconds per channel; the maximum is 83.2 milliseconds.

Note that adjusting the digital delay during audio program may cause discontinuities (popping sounds) while the adjustments are made. This is unavoidable and will cease when the delay is set.

Delay times may also be set from inside the POWER-Q REAL-TIME ANALYZER window. Refer to section 13.3 for more details.

Section 9: Using The POWER-Q Automatic Room EQ

9.1 HOW AUTOMATIC ROOM EQ WORKS

Sabine's Automatic Room Equalization works by playing a series of sine wave tones - comprised of the entire range of audible frequencies - through the sound system and measuring their respective energies at a reference microphone. The POWER-Q will automatically boost or cut frequencies to achieve as flat a room response as possible when the tones are played through the sound system and heard at the reference microphone. This flat room response is abbreviated "EQ RM" and is meant to serve as a consistent starting point for further adjustments.

Both microphone placement and the acoustics of the environment are important considerations for performing an acoustic analysis such as the POWER-Q's Automatic Room EQ. It is an unfortunate reality of room acoustics that acoustic analysis may vary quite a bit with differing reference microphone positions, depending on the size, shape and surface reflections of the environment. For example, in very small rooms, the number of paths for sound reflection may produce less than optimal results due to the phase interference patterns created by direct and reflected sound arriving at the reference microphone at different times. Keeping the mic away from reflections (e.g., walls and corners) and relatively close to the front of house speakers may minimize these interference patterns and the effects of a reverberant field. For similar reasons, when running a mono sound source through two or more speaker stacks or enclosures, we recommend playing the tones through only one speaker cabinet or stack, since sound arriving from multiple sources will arrive at the microphone at different times and create similar analysis problems. When running a stereo source through a stereo sound system, the POWER-Q will automatically analyze one channel at a time in sequence.

NOTE: Do not plug a microphone preamp or balanced line transformer into the reference input. This may cause the reference mic board to overheat and damage the POWER-Q.

9.2 POWER-Q CONTROLS FOR AUTOMATIC ROOM EQ. Select "AUTOMATIC ROOM EQ" from the display window by pushing soft key #1 from the MAIN MENU.

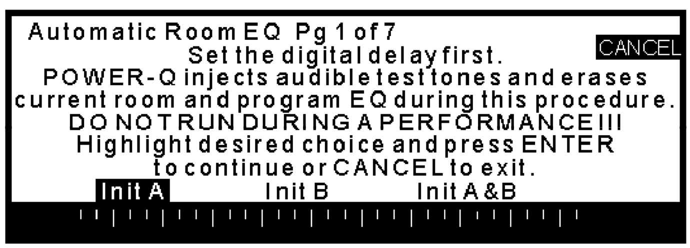


Fig. 17: Automatic Room EQ, page 1.

Because tones are played at audible levels, do not run this procedure during a performance, or even during rehearsal. You should run the Automatic Room EQ procedure BEFORE the performers arrive, but AFTER you have done any speaker delay alignment (see section 8). To prevent accidental injection of test tones in your sound system, the Automatic Room EQ function requires you to deliberately choose the procedure by hitting the "ENTER" button when prompted.

You may run the Automatic Room EQ analysis for either the A channel alone, the B channel alone, or both channels in sequence. Select "INIT A," "INIT B," or "INIT A&B" using the left/right arrow keys before hitting the "ENTER" button to begin the analysis. NOTE: Any time you run the Automatic Room EQ for a given channel, it will erase and replace the previous analysis for the channel(s) chosen. In addition, the current program shaping curve will be erased from active memory (though you may save the program curve and reload it from memory). For more information about program shaping curves and memory options, see sections 10.2 and 16.0.

Pressing ENTER displays Page 2 on the screen:

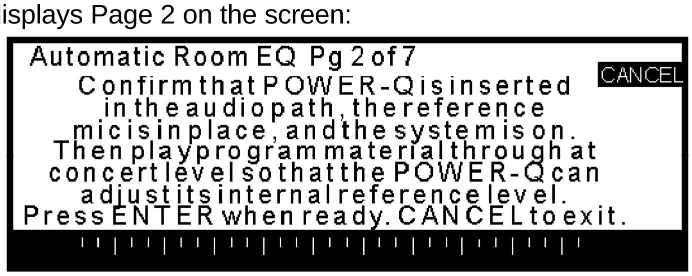


Fig. 18: Automatic Room EQ, page 2.

Make sure your POWER-Q is in place in the system (typically between the mixer output and the power amp or crossover input), your system is powered up, your reference microphone is in the correct position and plugged into the “Ref A” jack on the POWER-Q back panel, and you are playing music (a prerecorded CD will work fine) at performance levels. Hit “ENTER” again.

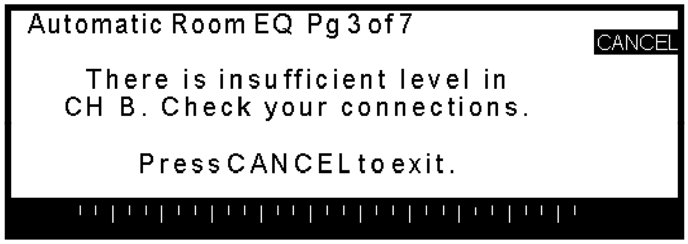


Fig. 19: Automatic Room EQ, page 3.

If the POWER-Q detects insufficient level at its inputs, this screen will notify you of the absence of signal in either or both channels. If there is no signal in channel A or B, either your connections to the POWER-Q are bad, or the POWER-Q may be in bypass (“BPASS” will flash in the upper right of your screen. Turn it off in the “GLOBAL PARAMETERS” menu.) If signal is present in one channel but not in the other, the POWER-Q will ask you to check your connections. If you’re using the unit as a single channel processor, you may proceed despite the indication of “insufficient level” in the unused channel. Simply hit “ENTER.”

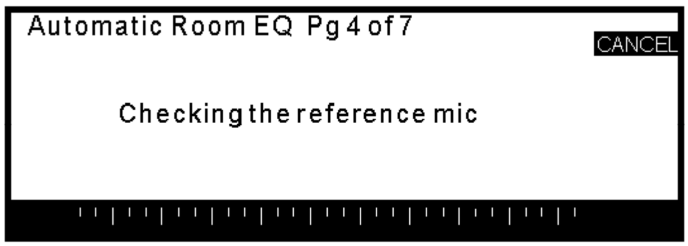


Fig. 20: Automatic Room EQ, page 4.

Page 4 of the Automatic Room EQ menu indicates that the POWER-Q is testing the reference microphone for the presence of signal (a 1 KHz tone) and calibrating the POWER-Q to the reference level heard by the microphone. If no signal is detected at the reference microphone input, the POWER-Q will prompt you with Page 5:

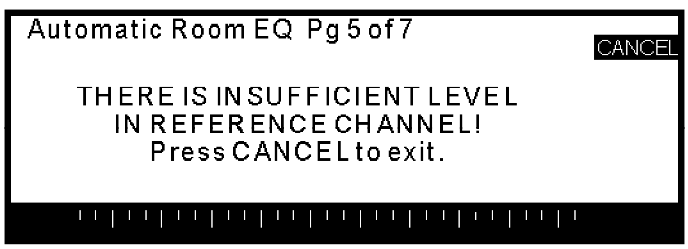


Fig. 21: Automatic Room EQ, page 5.

Make sure you are using a working microphone plugged into the “REF A” input on the back of the POWER-Q with a correctly wired cable. If you are using a reference mic that requires phantom power, turn on the phantom power in the “GLOBAL PARAMETERS” section, accessed from the MAIN MENU.

Assuming your reference microphone is plugged in, positioned correctly and working, the Automatic Room EQ analysis will begin, as indicated by page 6 of the menu.

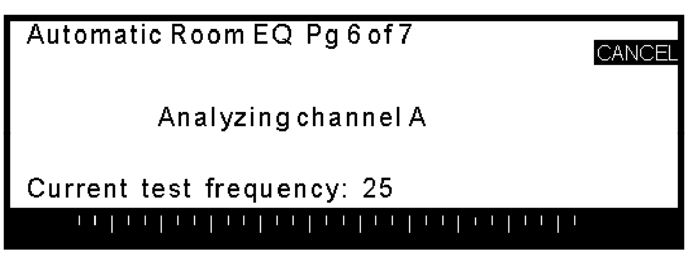


Fig. 22: Automatic Room EQ, page 6.

The POWER-Q plays a series of tones, from 20 Hz to 20 KHz, the range of human hearing. This process will take about 5 minutes per channel. The page 6 screen will indicate the POWER-Q channel being analyzed and the tone being played for the analysis.

When the analysis is complete, the POWER-Q will display page 7:

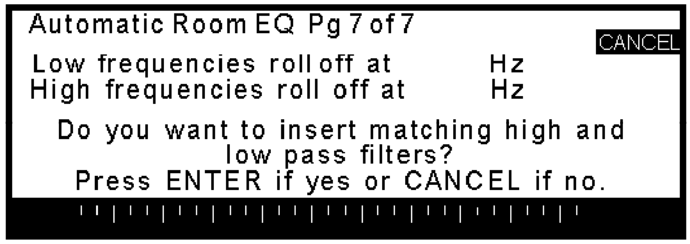


Fig. 23: Automatic Room EQ, page 7.

The analysis shows the high and low frequency roll-offs inherent in your sound system, and this screen allows the user the option of having the POWER-Q insert matching high pass and low pass filters. Press ENTER to insert the HPF and LPF, or press CANCEL if you choose not to insert roll-offs. Either ENTER or CANCEL will return you to the MAIN MENU.

It is possible to interrupt the Automatic Room EQ function at any time by pressing the “CANCEL” soft key. This will return you to the MAIN MENU.

Note that the AUTOMATIC ROOM EQ function will NOT add more than 6 dB boost to any EQ slider.

Section 10: Using The POWER-Q Graphic Equalizer

10.1 GRAPHIC EQUALIZER APPLICATIONS

Automatic Room EQ or an RTA analysis will produce a room EQ curve that may best be regarded as a starting point or baseline for further fine tuning with the POWER-Q graphic EQ. Experienced graphic equalizer users employ a variety of methods to arrive at their preferred sound, and the POWER-Q allows your personal touch in determining the way your system sounds.

A graphic equalizer is an engineer's most common choice for compensating for less than ideal acoustical reproduction. The quality of equipment used, the placement of speakers and the acoustical properties of a room will rarely result in tonally balanced sound reproduction. Even the best sound equipment will not produce sound to its optimal potential in most acoustical environments; for example, reflective parallel room surfaces (wall to wall, floor to ceiling) will create standing waves and acoustic resonances that will vary as a function of room dimensions.

A multiband graphic equalizer can compensate for unequal acoustical energy across the frequency bands of the audio spectrum. The POWER-Q graphic EQ provides you with a choice of measurement and calibration aids to perform automatic equalization, or it will allow you to rely on your own experience and hearing acuity to equalize an acoustical space. You can even combine methods. In fact, the POWER-Q goes a step further, providing you with some very powerful advantages for making a sound system sound as good as it can. You can arrive at a baseline "flat response" curve by one of two methods: by performing the POWER-Q Automatic Room EQ function (see Section 9), or by using the RTA to analyze pink or white noise played through your sound system into a reference microphone (see Section 13). However, you may not want to stop there. Many sound engineers can improve the sound of a flat response system and will tweak the sound of the EQ further, perhaps changing the overall frequency balance to match the performers' style or the requirements of a particular application. In such situations, the POWER-Q truly lives up to its name.

To fully understand the powerful options made available with the POWER-Q, it may be helpful to think of the unit as containing two separate graphic EQs. The "first" EQ is automatically set by the POWER-Q using the Automatic Room EQ function (see Section 9). Its function is to make the frequency response of the acoustical environment as even as possible, or flat. Let's call this group of EQ slider settings the "Room EQ." Think of this as the default starting point for sculpting sound in any environment.

In actual practice, many or most sound engineers will choose to adjust EQ settings differently for various applications. For example, the system EQ desirable for a hip-hop artist would likely require more low end boost than would be appropriate for an acoustic folk artist. This additional application-specific EQ is the "second" EQ added to the sound. Let's call it the "Program EQ."

The actual "Total EQ" added to your program is the combination of Room EQ and Program EQ (the actual settings of each slider added together constitute the setting for Total EQ). The EQ curve displayed on the POWER-Q Graphic EQ screen will vary according to selection made with the BYPASS options, accessed from the second page of the Graphic EQ screen. You may choose to bypass either the Room EQ, the Program EQ, both, or neither. Both the slider settings and the actual sound at the output of the POWER-Q will be determined by choice of bypass settings (see Section 10.3 for details).

Only the Program EQ will be saved and recalled by the POWER-Q. The Room EQ will be stored in memory as the default Room EQ, until the Automatic Room EQ procedure is run another time. Then the results of this new anaysis will be the Room EQ default. The reasoning for this is quite simple: every room is different (even the same room will have differences from one performance to the next), and you must calibrate your system (run Automatic Room EQ, which will take a little longer than 10 minutes for two channels) for optimal performance in that room. Once you do this, and establish the same “flat” starting point for further EQ changes, you may then add your favorite Program EQ. You can change from one Program EQ to another at the touch of a button. The ultimate sound heard in the room should be very consistent, since the room acoustics’ effect on system frequency response is measured and compensated.

Please note that the total boost for any EQ slider is limited to a total of 12 dB; the limit for any slider’s cut is 15 dB. Automatic Room EQ will not impose a boost greater than 6 dB, or a cut greater than 15 dB.

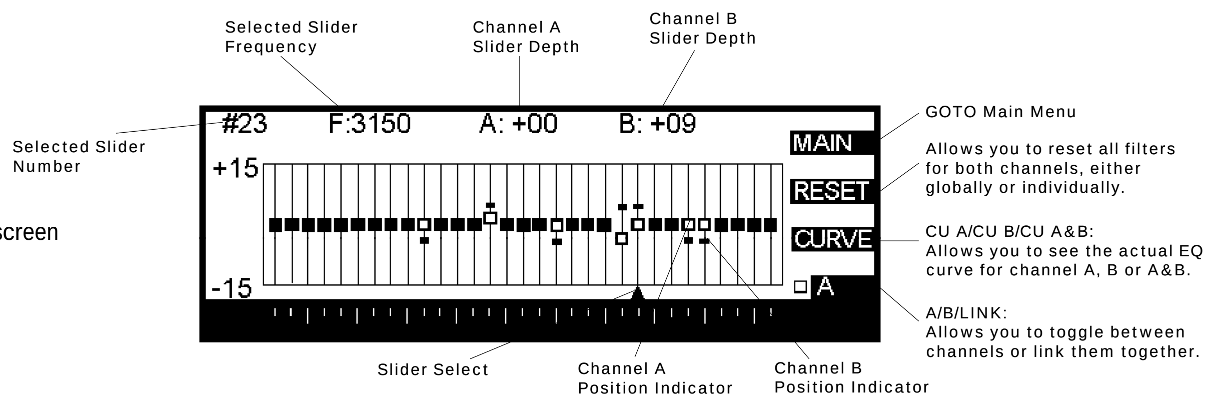
10.2 POWER-Q GRAPHIC EQ SCREENS AND OPTIONS: GRAPHIC EQ, CURVE DISPLAY, RESET FILTERS

To access the POWER-Q Graphic Equalizer controls, press soft key #2 (“GRAPHIC EQ”) from the MAIN MENU. (The MAIN MENU options can be scrolled using the “MORE” button or up/down arrow keys.) Pushing this key will display the image of a 31-band graphic equalizer on the screen. The frequency midpoints of each band are displayed at the bottom of the equalizer window.

There are two pages to the GRAPHIC EQ controls, accessed by pressing the “MORE” button. Pressing “MORE” changes the bottom soft key function from Channel Selection to Curve Selection.

Channel Selection and Slider Adjustment. Press “MORE” until soft key #4 reads “A,” “B,” or “LINK.” This selects the POWER-Q audio channel to be controlled by the EQ sliders.

Fig. 27: Graphic EQ screen

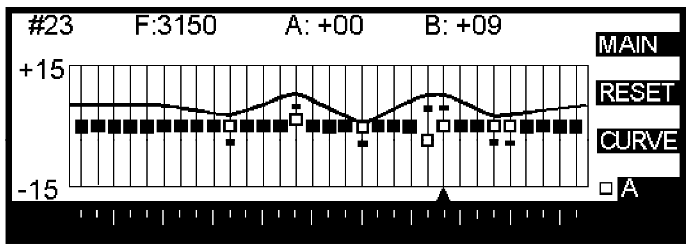


The "LINK" feature maintains the relative levels of the two channels while moving both. Use the left and right arrow buttons to choose the filter you wish to adjust (as indicted by the cursor at the bottom of the screen). The data wheel will move the sliders up or down. The "A" channel position indicator is a hollow square; the "B" channel is indicated by a smaller, solid rectangle. When the two channels' position indicators are at the same point, they will show as a large solid square.

The limits of the graphic EQ boost for any slider is 12 dB; the limit for a cut is 15 dB.

Curve Display. One powerful feature of the POWER-Q is its ability to display the total frequency response curve at the output stage of the unit, integrating all its equalization (graphic, parametric, FBX, and high and low pass filters).

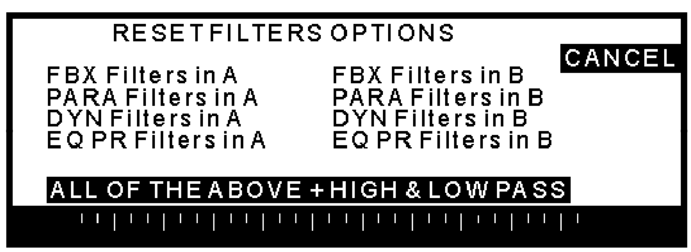
Fig. 28: Curve display



This curve is superimposed on the graphic screen and is accessed by pushing the soft key labeled "CURVE." This key can be toggled to show the frequency response curve of channel A ("CU A"), channel B ("CU B"), or both channels together ("CU A&B"). These curves represent the combination of ALL (graphic, parametric, FBX, and HPF/LPF) EQ settings in the POWER-Q. The superimposition of the response curve on the slider screen allows you to see the overall EQ resulting from graphic EQ adjustments with the front panel controls. The response curve is offset above the "flat" (zero boost or cut) center position of the EQ sliders in order to facilitate the display. This does not indicate any overall level boost from the POWER-Q.

Reset. The "RESET" button (soft key #2) opens up a screen that will allow global resetting of all filters for both channels or separate resetting of FBX, parametric or EQ PR filters for the A and B channels. It is possible to reset all FBX filters or just dynamic filters.

Fig. 29: Reset filters screen



To execute an option, arrow to your choice and push the "ENTER" button. Each type of filter for each channel must be reset individually unless every filter on both channels is reset. Pressing the "CANCEL" button will exit this screen without resetting any filters.

IMPORTANT NOTE: Reset ALL will NOT remove the Room EQ calculated by the POWER-Q during Automatic Room EQ. The only ways to remove or change Room EQ are: (1) Reinitialize the box by running a new Automatic Room EQ set-up, or (2) Bypass the Room EQ using the BYPASS function described below. When bypass is no longer selected, the POWER-Q will add the last Room EQ determined by Automatic Room EQ back into the signal path.

10.3 USING THE BYPASS CONTROL TO SELECT AND DISPLAY GRAPHIC EQ OPTIONS

A fresh conceptualization of live sound EQ. Although we have programmed the POWER-Q's Automatic Room EQ analysis to make the frequency response of a particular acoustic environment flat, in day to day usage many sound engineers will make additional modifications to a flat EQ. These changes are made to suit the engineer's ears or the performers' style, or for a variety of other reasons.

As discussed in Section 10.1, the POWER-Q should be thought of as two separate graphic equalizers that combine to function as one. The "first" EQ (Room EQ) sets the curve necessary to make the room response flat (measured and adjusted for you with the POWER-Q Automatic Room EQ feature), and the "second" EQ (Program EQ) is set by the engineer once the room response is flat, to suit a particular application. The Automatic Room EQ and the engineer's alterations combine to create the sound that emerges from the loudspeakers.

The POWER-Q allows you to develop and save Program EQs and load them into the POWER-Q graphic EQ with the touch of a button. An engineer who works with a variety of acts that each require a specific program curve will be able to customize an EQ curve for each act and name, save, and recall each curve quickly. In a new acoustical space, by performing the Automatic Room EQ procedure followed by the recall of a Program EQ, a sound engineer will achieve these powerful advantages:

- Consistency of sound.
- Quick set-up. You should be able to EQ the system in just a few minutes.
- Quick and reliable change of EQ from one performer to another.
- Predictable change of EQ in mid-performance.
- Quick loading of EQ templates to match the style of new or unfamiliar performers. Amaze them with how quickly you catch on to their sound!

Displaying and Hearing Room EQ, Program EQ, and their Combination.

Within the GRAPHIC EQ window, press the "MORE" button until soft key #4 reads "BYPAS," as shown below:

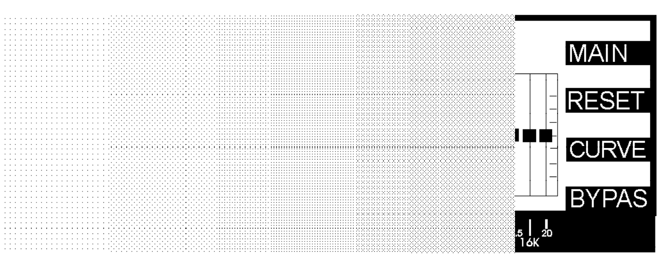


Fig. 30: Pressing MORE gives you these options

Press the “BYPAS” button to reveal this screen:

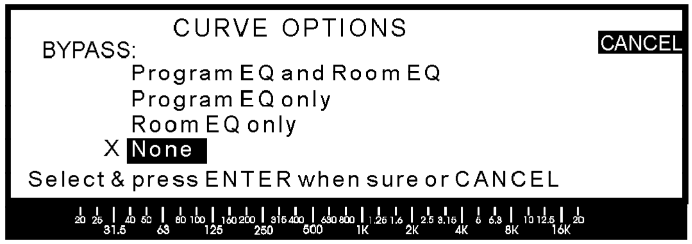


Fig. 31: Curve options screen

These four choices represent the graphic EQ curve that the POWER-Q is currently applying to your audio program, and displaying on the EQ screen. The “X” indicator shows which of the possible EQ curves are bypassed; in other words, “NONE” means both Program and Room EQ are inserted in the signal path. Make your selection using the arrow keys and ENTER button. Bypassing the Program EQ will show the Room EQ calculated by the Automatic Room EQ analysis, and bypassing the Room EQ will show the changes you have made to the EQ after the Room analysis. Both the audio and the display will change as a function of the selection you make.

PLEASE NOTE: You can also hard-wire bypass the entire POWER-Q. Choose BYPASS in the Global Parameters screen (Page 9 in the Main Menu). See section 4.2 for more information.

The POWER-Q will allow you to save and recall your Program Curves with the POWER-Q “STORED CONFIGURATIONS” function (see Section 16.0). If you have a Program EQ with a nice fat bass for a hip hop group and another for a crisp clarity for a vocal ensemble, you can load both of these curves from the POWER-Q’s memory with the touch of a button.

We believe this innovative conceptualization of room and program equalization affords you a uniquely powerful tool for quick, accurate, consistent and flexible equalization for a changing array of performers, applications and acoustics.

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We believe this innovative conceptualization of room and program equalization affords you a uniquely powerful tool for quick, accurate, consistent and flexible equalization for a changing array of performers, applications and acoustics.

Section 11: Using The POWER-Q FBX/Parametric EQ

11.1 TYPES OF FILTERS: Fixed FBX, Dynamic FBX, and Parametric.

FBX Feedback Exterminator Filters. The POWER-Q is the most recent refinement of the patented Sabine FBX technology that allows the unit to automatically sense and eliminate feedback. The POWER-Q offers 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 procedures described on pages 9 & 31 to “teach” the FBX automatic feedback control filters the frequencies and depths necessary to control feedback in your system.

BACKGROUND: The POWER-Q 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 in the GLOBAL PARAMETERS menu) at that 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. The POWER-Q 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. The third soft key button (“LOCK+/-”) locks and unlocks the fixed FBX filters. The "FBX F" (fixed) filters change to "FBX L" when locked.

DYNAMIC FBX filters are used to deal with transient feedback that comes and goes during a program. Changes in room acoustics (due to temperature or audience size changes), changes in gain (musicians almost always play louder during a performance than at sound check) and changes in mic position often result in new feedback frequencies arising during performance. 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 automatically move 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 POWER-Q must be “re-taught” where to place filters to eliminate feedback. To reset, press the “MORE” button until the option “RESET” appears next to the second soft key button. Press “RESET,” and you will be given the option of resetting all filters or just the FBX, graphic, or parametric filters for one or both channels. (See the section on GLOBAL PARAMETERS for more information about controlling FBX automatic feedback control filters.)

PARAMETRIC FILTERS. If a filter type is set to “PARAM” (parametric), it is possible to edit its frequency, width and depth by moving to the appropriate field with the arrow keys while in the LIST mode (press the second soft key button, “LIST/CURVE,” until you see a table of filters). You can set the center frequency of POWER-Q parametric filters anywhere between 20 Hz and 20 KHz with 1 Hz placement resolution. The filter’s WIDTH ranges from 9.99 octaves to 0.01

octave. The DEPTH can be set anywhere from -84 dB cut to +12 dB gain in 1 dB increments. These parameters are set using the arrow keys to choose the correct parameter and the data wheel to change the parameter value.

You can also edit the parametric filters in the CURVE mode, accessed by pressing the second soft key button (“CURVE/LIST”) until you see a frequency by amplitude display. This gives you an EQ-curve view of the filters edited in the LIST mode, plus allows “click and drag” graphic editing for parametric filters using the arrow keys to choose a filter’s frequency, width, and depth, and the data wheel to vary the parameter. The change will be displayed visually and audibly as the parameters are varied.

Note that the curve displayed is a combination of all filters (parametric, FBX, HPF, LPF and graphic) in the signal path for the selected channel. This is not only a powerful visual aid, showing the combined effects of all the EQ and filter changes you’ve made, but also a handy “final tweak” stage for your overall EQ. Assuming you have at least one unused parametric EQ filter, you can add this to your overall EQ curve with “click and drag” editing, which is easily done on the fly in the heat of live mixing.

If the filter frequency is set to "off," the width and depth cannot be entered. The POWER-Q assumes the filter does not exist.

Unlike many conventional analog parametric filters, the POWER-Q’s digital filters do not drift with temperature or cause phase-shifting outside of the filters.

A fixed or dynamic FBX filter that has been set by the POWER-Q can be “frozen” by moving to the appropriate type field and changing it to “PARAM.” At this point, the filter can be left as is or edited further. (See Section 21, A Pro’s Guide to Using the ADF Products, by Ken Newman.)

11.2 TURBO MODE. The "TURB+/-" soft key activates the POWER-Q TURBO set-up mode when "TURB+" is selected. (A warning screen will appear before TURBO is turned on so you will be aware that the POWER-Q is making the change.) TURBO mode maximizes the POWER-Q clip level and sets all FBX parameters to maximum feedback sensitivity. TURBO mode should only be used when "ringing out" feedback during set-up or sound check, since it maximizes the response speed of the POWER-Q automatic feedback control. The front panel CLIP LED will blink when TURBO is on.

TURBO mode should be **turned off** when the POWER-Q is in normal operating mode. It can be turned off in at least five ways:

1. By toggling the soft key to read "TURB-";
2. By hitting the "LOCK+/-" soft key (accessed by pressing the MORE button);
3. By setting at least one dynamic FBX filter (automatically takes the POWER-Q out of TURBO mode; TURBO soft key selection disappears);
4. By loading any program curve from the POWER-Q memory; or
5. By turning the POWER-Q off and back on.

If the TURBO mode is somehow left on when the FBX set-up procedure is completed, you may experience distorted audio through the POWER-Q since the clip level will be turned up fully. This problem is easily remedied by turning TURBO mode off. TURBO mode will NOT affect the output level of the POWER-Q.
NOTE: The TURBO soft key option disappears from the screen after it is disabled. It will reappear after you reset the FBX filters (see Section 10.2).

11.3 POWER-Q FBX/PARAMETRIC FILTER ADJUSTMENTS. There are two pages of soft key options under "FBX and Parametric Filters" (MAIN MENU option #3). Press the “MORE” button to access all the available options of this menu.

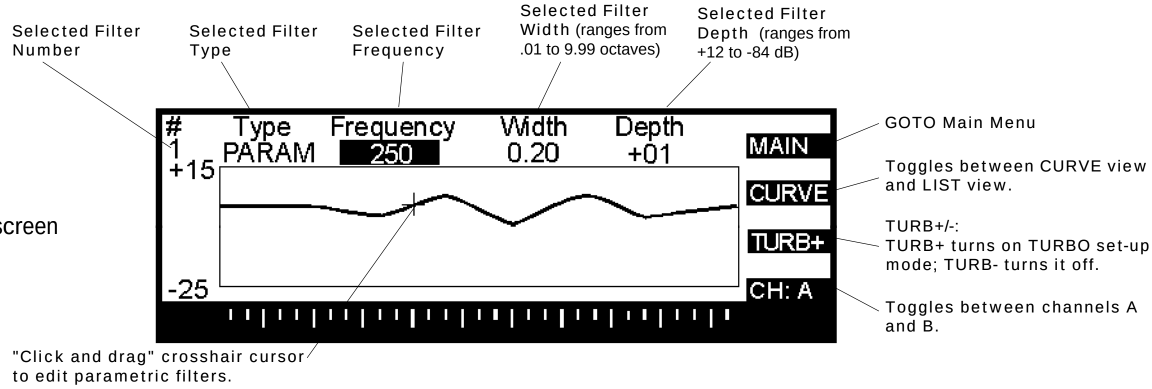


Fig. 32: Filter options screen

The “MAIN” soft key returns you to the MAIN MENU.

The “CURVE/LIST” button toggles between the LIST mode, a tabular listing of filters and control parameters, and a graphical representation of a frequency response curve (CURVE mode), showing the total curve for either channel of the POWER-Q as it is affected by all of the EQ adjustments you have made (including graphic, parametric, FBX, and high and low pass filters).

Both the Curve and List displays are available for either the A or B channel, selectable with the fourth soft key button.

11.4 FILTER CONTROL MENU: LIST MODE AND CURVE MODE

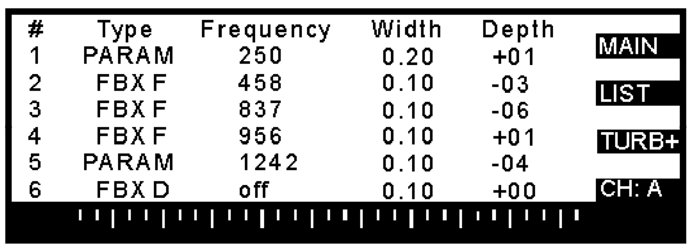


Fig. 33: List mode

LIST MODE: The POWER-Q provides up to 12 filters per channel for parametric equalization or feedback control. Each of these 12 filters can be set to any of the three types of filters: parametric, FBX fixed or FBX dynamic. Only six of the filters are displayed at a time; the remaining six will be displayed when the arrow keys are used to scroll “above” or “below” the screen currently displayed. Arrowing down from the 12th filter accesses the controls for setting high pass (HPF) and low pass (LPF) filters for each channel individually.

Use the arrow keys to move the cursor to the "Type" field of the filter you wish to change. Rotating the data wheel will select among three filter options: parametric, FBX D (for dynamic) and FBX F (for fixed). The third soft key button (“LOCK +/-”) locks the FBX F filters (they are locked when the display reads “LOCK+”) and will cause this display to read “FBX L.” This button should be toggled to “LOCK-” until the FBX fixed filters are set.

CURVE MODE:

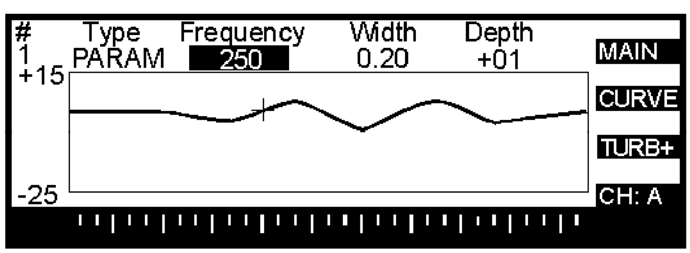


Fig. 34: Curve mode

Pressing the “CURVE/LIST” soft key switches the tabular LIST mode to a frequency response plot representing the sum total of all EQ settings (graphic, parametric, FBX, HPF, LPF) in channel A or B. Soft key #4 selects which channel’s curve is displayed.

The top line of the "Curve" mode display shows the filter number (1 through 12, high pass, low pass), the type of filter that corresponds to that number, the frequency of the filter, and its width and depth. These values will correspond to the settings in LIST mode. The filter chosen for this top line can be changed using the up/down arrow keys. (NOTE: This is important, as all other parameters are adjusted using the data wheel after selecting parameters with the left/right arrow keys.) When the displayed filter is set to “PARAM” using the data wheel, the frequency, width and depth of the parametric filter can be changed by using the arrow keys and data wheel.

For example, let’s say we want to add a parametric notch at 60 Hz, 10dB deep, and 0.20 octaves wide. While “PARAM” or “FBX L/F/D” is highlighted, use the up/down keys to select the number of the filter (1 through 12) you wish to use. Using the data wheel, select “PARAM” as the type of filter. Press the right arrow key to highlight "Frequency," and turn the data wheel to set “60.” (You will see the position indicator on the screen move as you change this value. If you’re adding to an existing EQ curve, the position indicator will follow the line of the curve.) Use the right arrow key to move to “Width,” and set this to “.20.” Move to the right one more time to “Depth,” and use the data wheel to set this value to “-10.”

Both the graphic display and the audio program will change in real time to reflect these changes. The cursor traces the existing displayed curve until you change the “Depth” value up or down from zero, and the actual curve display and audio will only change when a boost or cut is set. It is also possible to preset a boost or cut with frequency set to “off,” and then sweep the boost/cut through all the audio frequencies until you find the sound you like. (This makes for quite an interesting flanging sound if you use a wide filter while listening to audio through the POWER-Q!)

This is a very powerful feature that allows visualization of “on-the-fly” fine adjustments of EQ. You can refine your system EQ curve with a touch of parametric EQ added to the response curve you’re already working with - and you can see and hear the results instantly.

11.5 THE “MORE” BUTTON FOR THE FBX/PARAMETRIC WINDOW

Pressing the “MORE” button in the FBX/Parametric window accesses the “RESET” and "LOCK+/-" soft keys.

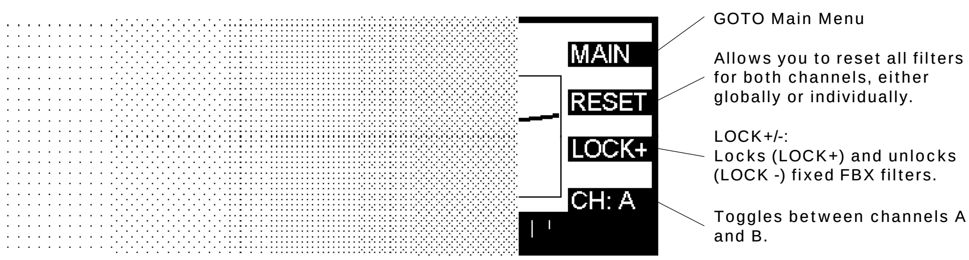


Fig. 35: Pressing "MORE" gives you these options

The "RESET" key gives you the choice of resetting all filters or selected banks. These choices are identical to the options for resetting filters in the Graphic EQ window (see section 10.2).

The "LOCK+/-" soft key locks (+) and unlocks (-) fixed FBX filters. A locked FBX filter will not change or notch any deeper.

11.6 STEP-BY-STEP FBX GUIDE

Follow this step-by-step guide to set up the FBX function:

1. Place the microphones and speakers in the locations where they will be during the program, and patch in the POWER-Q using one of the applications described on page 7. Set all controls for your sound system at the settings that will be used during the performance.
2. Making sure the main sliders on the mixer are pulled down, power up the sound system, the POWER-Q and then the power amp.
3. Select soft key #3 ("FBX and PARAMETRIC FILTERS") from the MAIN MENU, and make any adjustments to the default configurations.

4. Slowly raise the gain (for one channel only). As feedback begins to occur, the FBX filters will automatically detect the correct feedback frequency and apply a narrow filter to eliminate it. Continue raising the gain until all FBX fixed filters are set. If you wish to lock the FBX fixed filters to prevent them from notching more deeply, press the "LOCK+/-" soft key at the right of the screen so it reads "LOCK+."
5. Repeat this procedure for the other channel.

Section 12: Using The POWER-Q High & Low Pass Filters

The HIGH and LOW PASS FILTER controls are accessed via the “Parametric and FBX Filters” menu selection (option #3). Pressing the down arrow key repeatedly (while in the LIST mode, or under “TYPE” in CURVE mode) through all 12 filters ultimately shows the HIGH and LOW PASS FILTER menu.

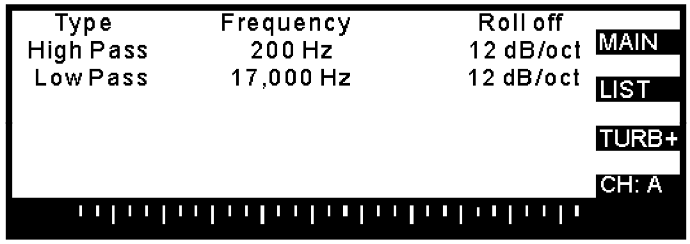


Fig. 36: High and low pass filters (list mode)

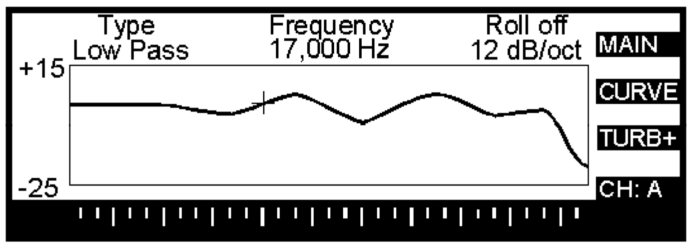


Fig. 37: High and low pass filters (curve mode)

These are the high- and low-end roll-off filters that are used to custom-tailor the low- and high-end frequency response. A “high pass filter” allows frequencies higher than a set value to pass; in other words, it filters out low frequencies, and the opposite happens with a “low pass filter.” 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 POWER-Q’s output signal for telecommunications.

This screen allows setting of the high and low pass filter points. The HIGH PASS FILTER can be inserted starting at any frequency up to 3,000 Hz. The LOW PASS FILTER can be inserted from 1,000 to 20,000 Hz.

Note that turning the data wheel to the extreme low range of the HPF and the extreme high range of the LPF turns the filters off.

Section 13: Using The POWER-Q Real-Time Analyzer

13.1 USING A REAL-TIME ANALYZER

Although it is possible to achieve excellent results setting up the EQ for a room using the POWER-Q's Automatic Room EQ (see section 9), many engineers prefer to perform a REAL-TIME ANALYSIS of a sound system and make equalization adjustments based on the results. You may elect to do this in lieu of, or in conjunction with, Automatic Room EQ.

RTA analysis involves generating noise (usually pink noise) through a sound system and observing the energy across frequency bands as heard by a reference mic positioned in the acoustical environment. Adjustments are made to an equalizer until the response curve of the system, as heard by the reference mic, matches the specifications of the operator.

The RTA feature of the POWER-Q offers a no-compromise, filter-based RTA that will equal or surpass the performance of other brands and models on the market that cost as much or more, yet offer none of the POWER-Q's other features.

13.2 POWER-Q RTA ADJUSTMENTS

An RTA analysis will require the POWER-Q's built-in noise generator (pink or white) and a reasonably flat response microphone (not supplied) plugged into the "REF A" jack on the back of the POWER-Q. You may select the "REAL-TIME ANALYZER" option by pressing the #4 soft key in the main menu. The screen will look like this:

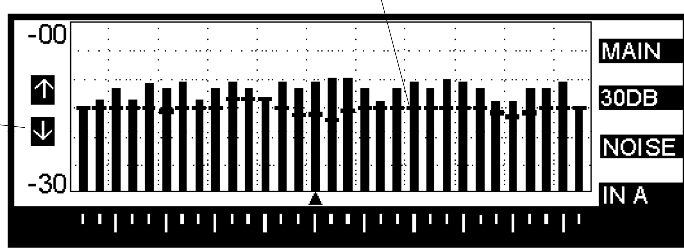


Fig. 38: RTA screen

The "MAIN" soft key returns you to the MAIN MENU.

The second soft key allows adjustment of the vertical scale of the RTA, giving you choices of 15 dB, 30 dB, or 60 dB. Note that when you switch from a 60 dB range to either 30 or 15 dB, the RTA response may disappear from the screen because it is below the range being displayed. You may move the range displayed on the screen with the up/down arrow keys to bring the RTA response into optimal view. When the up/down arrow keys are enabled to perform this function, an up/down arrow icon will appear at the left extreme of the screen display.

Pressing the third soft key ("NOISE") accesses the NOISE GENERATION OPTIONS screen:

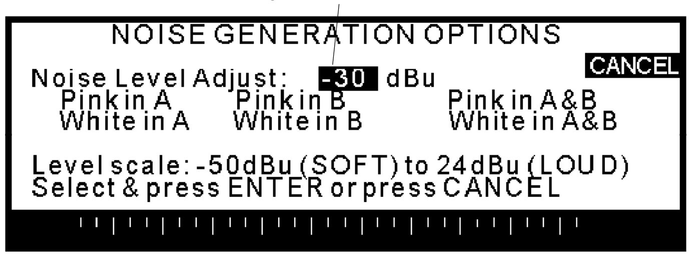


Fig. 39: Noise generation option menu

These controls allow you to choose the type of noise generated at either or both channel outputs of the POWER-Q, and to set the level of the noise with the data wheel. Make sure you set the noise level before you press ENTER. The noise level defaults to -30 dBu. (The noise level can still be adjusted after ENTER is pressed and noise operation begins.) When you have selected the desired type of noise and output channel(s) with the arrow keys, press ENTER to begin noise generation. The POWER-Q display will automatically switch to show the real-time analysis in progress (displays the response you choose under soft key #4). Pressing the NOISE soft key once more turns off the noise generator.

The fourth soft key chooses either the A or B channel input or output response of the POWER-Q, or the frequency response of the program heard at the reference microphone ("REF").

Press "MORE" to see the following screen:

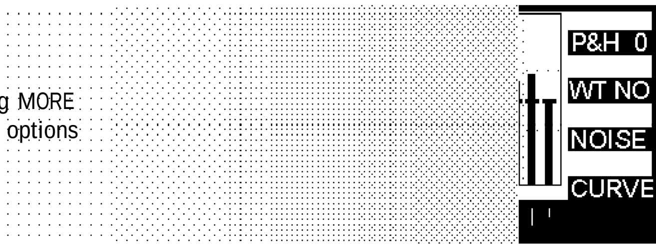


Fig. 40: Pressing MORE gives you these options

The first soft key varies the peak and hold reset time of the RTA, from instantaneous ("P&H 0"), where there is effectively no peak and hold; to 1 second ("P&H 1"); to 4 seconds ("P&H 4"); to infinity ("P&H 4"), which will display and hold the highest peak until it is reset. Peaks will show in addition to the RTA display.

The second soft key chooses the weighting for the RTA response: none ("WT NO"), A weighting ("WT A"), B weighting ("WT B"), and C weighting ("WT C"). In normalized mode, the RTA display is scaled so that the highest peak is at the same level as the input level.

The third soft key opens the noise generation option menu (see description on page 34).

The fourth soft key ("CURVE") shows on the screen the EQ response curve of the sum of all the graphic, parametric, HPF, LPF, and FBX filters placed in the signal path. This can be displayed for channel A ("CU A"), channel B ("CU B"), or both ("CU A&B").

Press "MORE" again to see the following screen:

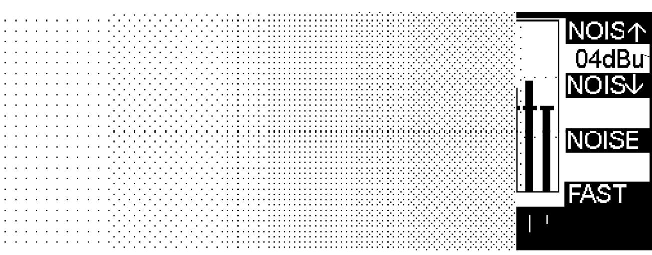


Fig. 41: Pressing "MORE" gives you these options

"NOIS ↑" and "NOIS ↓" are "phantom" soft keys that only appear when you select an option in the noise generation options window. Pressing "NOIS ↑" increases noise level, and pressing "NOIS ↓" decreases noise level.

Press "MORE" again to see this screen:

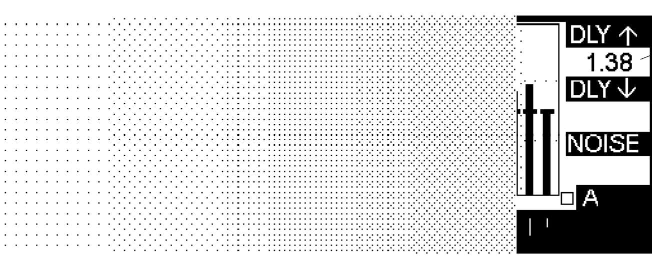


Fig. 42: Pressing "MORE" gives you these options

"DLY ↑" and "DLY ↓" enable adjustment of the digital delay time for channel A or B. Delay is adjusted using soft key #1 to increase delay and soft key #2 to decrease. Channels A and B delays can only be set independently of each other.

13.3 USING THE POWER-Q RTA AND DIGITAL DELAY SETTINGS TO MINIMIZE COMB FILTERS

Comb filters arise in any acoustical setting where direct and reflected sounds (or two or more speakers playing the same sound source) combine in equal, or near-equal, amplitude (i.e., are close to the same volume). Slight delays in the arrival of sounds at a particular location in an acoustic environment will cause a phase interference pattern. Some frequencies will be attenuated, while others will be reinforced, producing a frequency response showing many "peaks and valleys" and also resembling the teeth of a comb (see Figure 9 on page 15). The specific frequencies that are boosted or cut will depend on the physical layout of the acoustical space and the location of the speakers and your reference listening position. The attenuated frequencies will be heard at a lower volume, and the reinforced frequencies will be emphasized and are prone to feedback.

Unfortunately, equalization cannot cure comb filtering. However, digitally delaying the arrival of some of the sound creating the phase interference pattern can minimize the comb filters. You can experiment with the digital delays in the POWER-Q while analyzing pink noise as heard at the reference microphone, and observe the results. Here's a step by step guide to this comb-filter minimization procedure:

- Make sure your system is set up correctly and passing audio signal properly. Make sure your two speakers pass signal at equal volume. (NOTE: The steps described here will apply to a system with two or more speakers or speaker stacks aimed at the audience.)
- Set your reference microphone up at an appropriate listening spot. To maximize the potential for comb filtering, the microphone should be close to equidistant from the two speakers (to hear the sound from the speakers at equal volume).
- From the MAIN MENU, select option #4 ("REAL-TIME ANALYZER"). Select "REF" at soft key #4 (this will display the RTA response of the reference microphone). Then press the "MORE" button 3 times to reveal the screen in Fig. 39 on page 35 of this manual.
- Press soft key #3 ("NOISE"). The NOISE GENERATION OPTIONS screen (Fig. 36, page 34) will appear.
  Set the gain of the noise generator to the desired setting, then select "PINK in A&B" and hit ENTER. This will play pink noise through both speaker stacks.
- While observing the RTA as heard by the reference microphone, experiment with delays in both channels A and B until the comb filtering is minimized, i.e., the frequency response as heard by the reference microphone shows minimal peaks and valleys.
- When you're satisfied your comb filtering is minimized, proceed with equalization.

See Section 8.0 (DIGITAL DELAY) for a detailed discussion of comb filtering.

13.4 USING THE POWER-Q RTA DURING PERFORMANCE

The POWER-Q RTA need not be used only as a set-up tool. It can also be employed as a visual aid when mixing live or recorded sound. If the reference microphone is set up and feeding the RTA input during performance, you will see a frequency map of the sound energy emanating from the speakers, as heard at the reference mic. Superimposed on top of the RTA you will see the graphic EQ sliders in the positions you have set them.

The RTA may prove useful in the following ways:

- During sound check, when it is possible to use the mixing console's "Solo" function, or by using an auxiliary bus during performance, it is possible to analyze the frequency map of an individual voice or instrument with the POWER-Q RTA. This may show useful frequency information for optimizing an input's EQ using the console's equalizer controls on the appropriate channel.
- During performance, when the entire audio output of the console is routed through the POWER-Q, you will see the audio program's frequency map. This may help you determine a proper frequency balance of the overall mix using POWER-Q graphic EQ controls.
- If feedback arises during sound check or performance, the swelling frequency of the feedback will be apparent on the RTA, thus indicating the frequency to attenuate in order to remove the feedback. (The POWER-Q FBX feature will do this for you automatically.)

Section 14: Using The POWER-Q Compressor/Limiter

14.1 COMPRESSOR/LIMITER APPLICATIONS AND USE

The dynamic range (how loud to how quiet) of the human ear is something on the order of a ratio of a billion to one; i.e., the loudness of a sound like a close jet engine is a billion times louder than the level of air molecules striking the human eardrum. Any sound reproduction that depends on electronics will fall far short of the dynamic range of our hearing, limited at the quiet extreme by noise inherent in the system and at the loud extreme by clipping (distortion).

Passing an audio signal through any electronic circuit will create noise (often a VERY tiny amount), and much of the skill in operating sound equipment consists of the fine art of optimizing the balance of clean signal and noise. A compressor (or, in its most powerful form, a limiter) is probably the most widely used tool in the control of dynamic range. In simple terms, a compressor is designed to restrict the dynamic range of an audio program; i.e., to make quiet signals louder, and loud signals quieter. A compressor becomes a limiter when the compression ratio (the ratio of the output gain change to the input gain change) is so high that the output level won't rise above a "brick wall" ceiling regardless of how loud the input gets.

The practical benefits of compression include: (1) speaker protection, as the compressor will restrict the musical peaks from excessive amplitude; (2) a greater average volume and a "fatter" sound, since turning down the peaks allows the rest of the program to be raised in volume; and (3) more gain-consistent mix components. For example, a compressor will restrict the dynamic ups and downs of a vocalist so the vocal won't fall below the rest of the mix when the singer sings quietly, or rise out of the mix when the vocals are loud.

The most common negative side effect of compression is a pumping or "breathing" sound that sometimes accompanies constant gain modulation. This can be minimized by proper adjustment of the "attack," "release," "thresh," and "knee" controls.

14.2 POWER-Q COMPRESSOR/LIMITER ADJUSTMENTS

The POWER-Q compressor/limiter menu is accessed from the MAIN MENU by pressing soft key #5 ("COMPRESSOR/LIMITER").

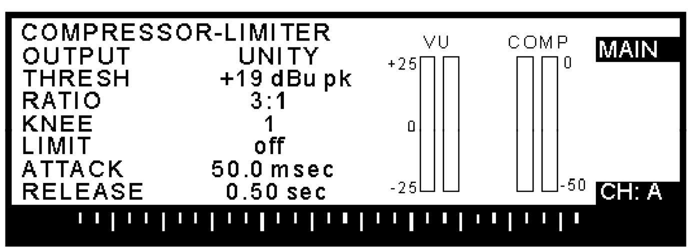


Fig. 43: Compressor/limiter screen

"OUTPUT" allows you to vary the signal gain after it has been compressed. Without this control, the output signal level would fall below the input, since the function of the compressor is to restrict the increase in gain of signals louder than a user-defined threshold. "OUTPUT" allows you to make up the gain (and then some) lost by compressing the signal. Note that the OUTPUT control operates regardless of the other COMPRESSOR/LIMITER settings, so it can serve as an output level control for the POWER-Q. Exercise caution when raising the output level of the POWER-Q, as this may have consequences for the equipment downstream in the signal chain.

"THRESH" sets the threshold at which the compressor begins to affect the signal. Any signal below this threshold will pass through the circuit with the gain unaltered (i.e., unity gain).

"RATIO" is the compression ratio, designated by two numbers separated by a colon. The first number represents a potential change in gain at the input stage of the compressor; the second number represents the corresponding change that will be allowed at the output of the compressor. In other words, a 3:1 compression ratio means that the output signal level (in dB) will rise 1/3 as much as the gain change at the input level for signal levels that rise above the threshold. A compression ratio of 4:1 is effectively a limiter; no matter how much louder the input rises above the threshold, the output level remains unchanged.

“KNEE” refers to a changing compression ratio as the compression threshold is approached and exceeded. Most compressors allow a choice of either “hard knee,” which applies the full compression ratio dynamics control to the signal immediately as the input level crosses the threshold, or “soft knee,” which varies the slope of the compression as the threshold is approached and crossed. “Soft knee” smooths the onset of the compression.

The POWER-Q allows a variable, user adjustable knee. The degree of “softness” is adjusted with the data wheel, and is variable from 1 to 40dB. This value refers to the range in dB of the input signal (with the threshold as the midpoint) over which the slope of the compression will vary.

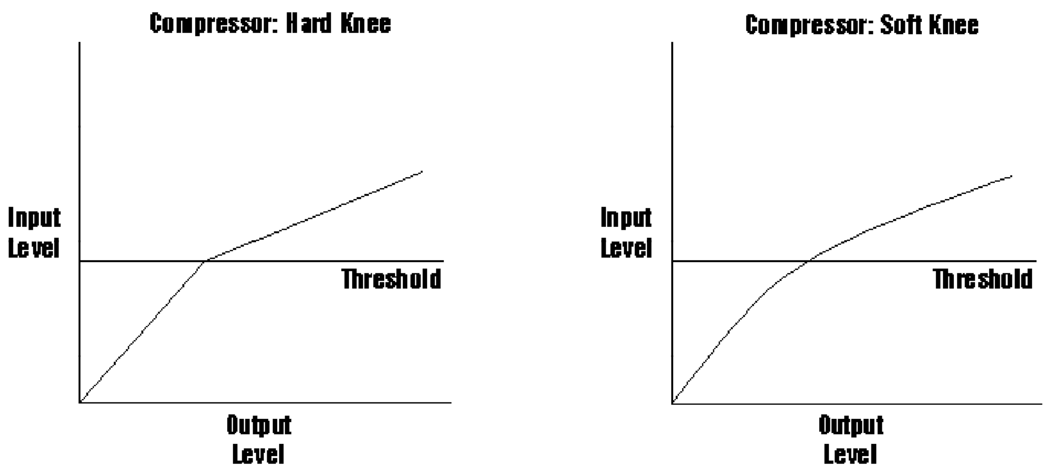


Fig. 44: Compressor hard knee (left) and soft knee (right)

The lowest “knee” value (1) represents instant full compression when the input level crosses the threshold; the highest value (40) represents a soft-knee compression that will begin to gently compress the input signal 20dB below the threshold, and allow the full compression ratio 20dB above, for a total “soft-knee” range of 40dB.

“LIMIT” sets an absolute output gain level. Peak input signals rising above the threshold value set with this parameter will be compressed so extremely as to reflect no additional gain at the output stage. Note that the “LIMIT” threshold and the compressor threshold (“THRESH”) can be set independently, allowing both a mild degree of compression and brick wall peak limiting. All other parameters besides “THRESH” and “RATIO” are common for both the compressor and limiter.

“ATTACK” sets the speed with which the signal is compressed once the gain exceeds the threshold. “ATTACK” is generally set to a very quick response and can be varied from 1 to 99 msec.

“RELEASE” sets the speed with which the output signal returns to unity gain when the input signal falls below the threshold. It can be varied from 50 milliseconds to 5 seconds.

The compressor window also displays metering of the input signal (both channels) and the compression added to the signal (both channels). When compression is engaged, the meters will work in opposite directions.

The compressor/limiter functions can be adjusted individually for channels A and B, using the selector knob to toggle between them. The front panel of the POWER-Q has a yellow LED (marked "LIMIT") for each channel that will illuminate when the compressor is engaged (when the input threshold is crossed for a knee setting of “1,” or 20dB below the threshold for a knee setting of “40”).

Section 15: Using The POWER-Q Expander/Noise Gate

15.1 EXPANDER/NOISE GATE APPLICATIONS AND USE

Just as compression is designed to restrict the extremes of the dynamic range of an audio program, an expander is designed to *increase* dynamic range. In normal usage, the dynamic range is increased by downward expansion, i.e., by turning down quiet signals to a lower level. The most common application is to reduce noise in a program, as the noise will generally be quieter than the signal. Any audio that falls below a gain threshold will be turned down; when it is turned down so far as to be turned off, the expansion function is known as a gate. Since the turned off/turned down audio is generally undesirable noise (tape hiss, component noise, guitar amplifier noise, leakage from an off-mic sound source, etc.), any device that automatically turns off such noise has come to be called a “noise gate.”

The major problem when using expanders and gates occurs as the audio crosses the gain threshold, engaging the expansion. An abrupt change from “on” to “off” or vice versa can produce an audible pop or click and tends to be more of a problem as audio signals sustain and decay. For audio with a slower attack (such as bowed string instruments), the gate may also abruptly open in mid note. For human speech, a gate may cut off the beginnings of consonant sounds. For this reason, you must exercise care in the setting of the gate’s threshold, knee, attack and release parameters. If you are using the POWER-Q as an insert point of a mixer channel, the settings may be less critical than for using the POWER-Q between a mixer and an amplifier, where you run the risk of abruptly turning off low level portions of your mix via poor gate settings.

15.2 POWER-Q EXPANDER/NOISE GATE ADJUSTMENTS

The POWER-Q expander/gate is accessed by pressing soft key #6 from the MAIN MENU (“EXPANDER/NOISE GATE”). The window looks like this:

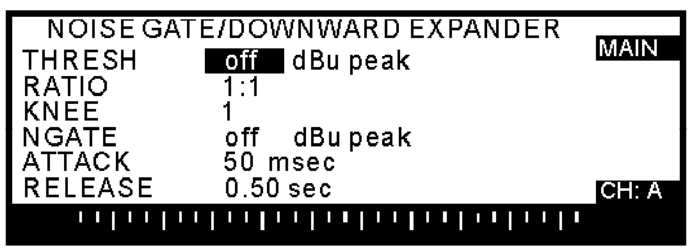


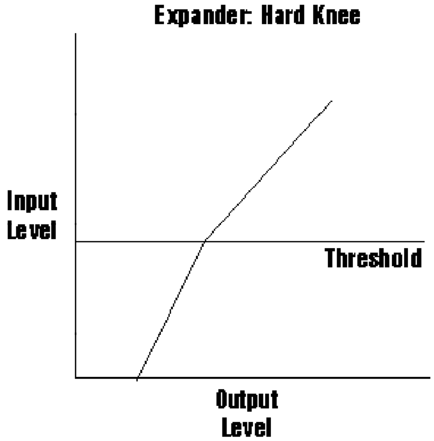
Fig. 45: Noise gate/ downward expander screen

Many of the EXPANDER/NOISE GATE parameters are essentially identical to COMPRESSOR/ LIMITER adjustments, only they act in reverse fashion.

“THRESH” sets the level at which the expander turns on and off. It is adjustable from -90 dBu (off) to -20 dBu peak.

“RATIO” is the expansion ratio. The first number represents a change in gain at the input stage of the expander; the second number represents the corresponding decrease in gain for the output of the expander when the input level falls below the threshold. In other words, a 1:3 expansion ratio means that the output level of the expander will fall three times as quickly as the input level for signals that fall below the threshold. An expansion ratio of 1:4 is a noise gate; signals that fall below the threshold will be turned down completely.

“KNEE” refers to the slope of the gain reduction that occurs as the threshold of the expander is approached. A value of 1 (the “hardest” knee) will simply turn on and off immediately as the threshold is crossed. Higher values will “soften” the knee with an accelerating onset of expansion as the input level approaches the gate threshold. This “softens” the turn-on and turn-off of the gate. A setting of “40” will begin to gently expand the signal 20dB above the threshold and achieve the full expansion ratio 20dB below the threshold.



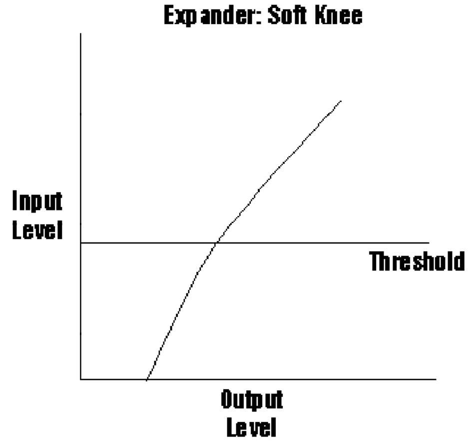


Fig. 46: Expander hard knee (left) and soft knee (right)

“NGATE” allows an independent threshold (in addition to the threshold setting for expansion) to be set for the onset of a noise gate, affording simultaneous operation of both gating and expansion. “THRESH” and “RATIO” values will differ for the gate and the expander; the settings of all other parameters will be common to both.

“ATTACK” sets the length of time the program output level takes to return to unity gain when the input level threshold is crossed. It is adjustable from 1 to 99 milliseconds, with 1 millisecond resolution.

“RELEASE” sets the speed of the expansion onset and is adjustable from 50 milliseconds to 5 seconds.

The expander/gate functions are separately adjustable for channels A and B. When expansion or gating is engaged (i.e., when the threshold is crossed for a low KNEE value or above the threshold for softer KNEE settings), the red LED (“GATE”) on the POWER-Q front panel will illuminate.

Section 16: Saving & Loading Stored Configurations

16.1 RECALL AND STORAGE OPTIONS AND USE.

The POWER-Q allows up to 19 memory locations for storing parameter settings you've determined for a particular application, acoustic venue or performer. These settings may be recalled later, saving you much time and effort.

The POWER-Q memory options are very flexible and powerful, allowing a variety of choices of how you store your settings. You may elect to save or load every POWER-Q parameter (graphic EQ, FBX, parametric EQ, compression/limiting, noise gate and delay) or the program shaping curve only.

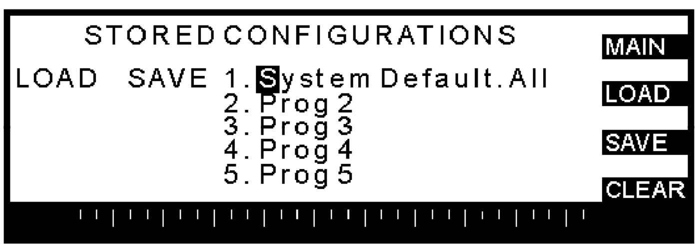
16.2. SAVING EQ SETTINGS: ROOM EQ AND PROGRAM EQ.

Section 10.3 describes the POWER-Q conceptualization of Room EQ and Program EQ. Please note that the POWER-Q will automatically store the most recent Room EQ calculated using the Automatic Room EQ analysis (see Section 9.2). Only one Room EQ is stored in the POWER-Q memory at a time, and it can only be changed (by performing a new Automatic Room EQ) or bypassed, but cannot be otherwise stored and recalled. However, up to 19 separate Program EQ settings can be stored and recalled by the user.

16.3 USING THE POWER-Q STORED CONFIGURATION WINDOW.

Select option 8 ("STORED CONFIGURATIONS") from the MAIN MENU soft keys to access these controls.

Fig. 47: Stored configurations screen

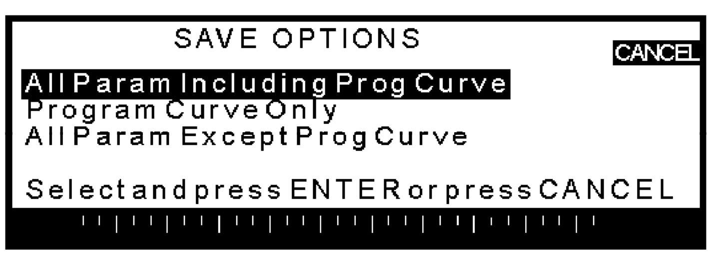


"LOAD," "SAVE" and "CLEAR" are "phantom" keys that only appear in certain situations. "LOAD" appears only when the cursor is on a named configuration (including system default). "SAVE" appears only when the cursor is on an unnamed configuration (NOT including the system default). "CLEAR" appears only when the cursor is on a named configuration (NOT including the system default).

There are 20 memory locations for loading and 19 for storing presets, accessible by scrolling with the up and down arrow keys. These are numbered consecutively, followed by a memory name (user definable) and a suffix. The suffix "CRV" indicates that the memory location holds only the program curve; "PRM" indicates that all parameters EXCEPT the graphic EQ curve have been stored; and "ALL" indicates that both the curve and parameters have been saved. Memory location #1 (System Default) cannot be saved or cleared - only loaded. All other memory locations are user definable.

To SAVE your settings, use the up/down arrow keys to select a memory location. Press the "SAVE" soft key. The POWER-Q will display this window:

Fig. 48: Save options screen



NAME A CONFIGURATION: Use the up/down arrow keys to select the parameters you wish to save (all parameters, Program Curve only, all parameters except the Program Curve). Press the “ENTER” button to save your selection. The POWER-Q will automatically return you to the “STORED CONFIGURATIONS” window, with the first character in the name of the new memory highlighted (the default name is “PROG”). Use the data wheel and the left/right arrow keys to write a new name for the memory location.

To LOAD a configuration (see the screen below), use the up/down arrow keys to select the desired curve. Press the “LOAD” soft key. You will be given a choice of parameters to load (all, Program Shaping Curve only, or all parameters except Program Shaping Curve), assuming you saved all the parameters when you originally created the memory. Otherwise your load options will be limited to the parameters you saved. Hit “ENTER” to load your selection.

Note that loading new parameters (with or without the Program Curve) will briefly mute audio that is playing through the POWER-Q when "LOAD" is executed. This will prevent popping if digital delay settings change with the loading.

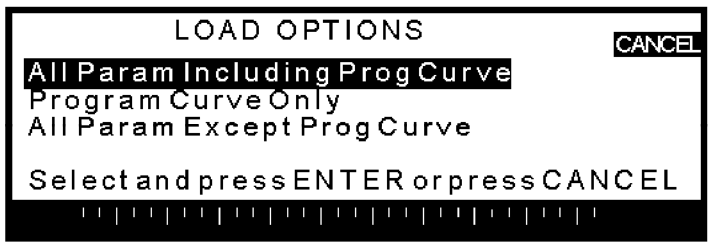


Fig. 49: Load options screen

Any memory (except the system default) may be cleared by pressing the “CLEAR” soft key. There is a built in fail-safe option to make sure you want to clear your memory.

The left margin of the “STORED CONFIGURATIONS” window will indicate the most recent memory loaded and saved.

Note that when the POWER-Q is turned off, or if the electrical service is interrupted, the unit will return to all its most recent settings upon power-up.

Section 17: Global Parameters: Configuring Internal Default Values

Choose option 9 ("Global Parameters") from the Main Menu to adjust the following:

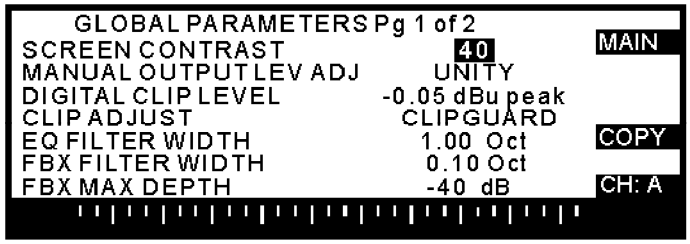


Fig. 50: Global parameters screen, page 1

"SCREEN CONTRAST" changes the "tilt" of the front panel LCD, allowing the display to be adjusted to optimal legibility.

"MANUAL OUTPUT LEVEL ADJUST" adjusts the signal level at the output stage of the POWER-Q. It is adjustable in half dB increments from -32 dB to +32 dB, relative to unity gain.

"DIGITAL CLIP LEVEL" allows you to adjust the analog input level to optimize the available dynamic range of the A/D conversion. The output level of the D/A conversion is compensated reciprocally to preserve unity gain from the input to the output. Too high an input level will cause clipping, and too low a level will result in noise. This parameter is adjustable from -0.05 dBu to +31 dBu, in half dB steps. When "CLIP ADJUST" is set to "AUTO," this control is defeated.

"CLIP ADJUST" allows either manual control over the clip level (see above) or automatic control (with Sabine's patent pending ClipGuard™). ClipGuard™ works transparently to optimize the dynamic range of the A/D converter, preserves unity gain and increases the effective dynamic range of the POWER-Q to over 110 dB. We highly recommend leaving the CLIP ADJUST control set to CLIPGUARD.

"EQ FILTER WIDTH" sets the width of the graphic EQ filters as defined by the width of a filter notch at the -3dB (half power) level. Note that the graphic EQ filters are constant-Q filters, meaning their width will not change as the filter depth increases. "EQ FILTER WIDTH" is adjustable from .5 to 1.0 octave, in .01 octave increments.

"FBX FILTER WIDTH" globally adjusts the width of fixed and dynamic automatic FBX filters. It does not affect parametric filters. Width is adjustable from .01 to 1.00 octave, in .01 octave increments. Narrow filters are more transparent and affect audio content less; wide filters provide more robust feedback control and allow greater microphone mobility before feedback occurs. A 0.10 octave setting is recommended for musical performances, and a setting of 0.20 octave is recommended for voice applications.

"FBX MAX DEPTH" globally adjusts the maximum cut an FBX filter can make. The range varies from -80dB to -6dB, adjustable in 1dB increments.

Access page 2 of "GLOBAL PARAMETERS" using the downward arrow key or the "MORE" key.

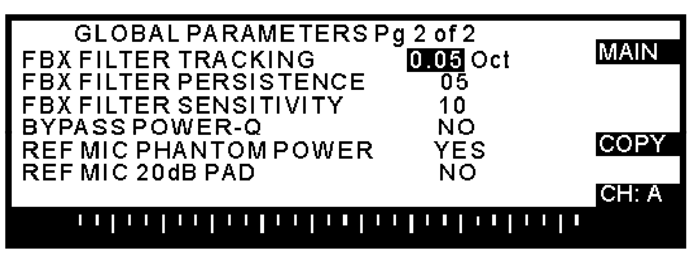


Fig. 51: Global parameters screen, page 2

"FBX FILTER TRACKING" globally adjusts the range over which an automatic FBX filter can move to accommodate feedback frequencies that may drift slightly as a result of humidity and temperature changes (see section 11.2). This parameter is adjustable to a width from .01 to .10

octave, centered around the original frequency of the automatic FBX filter. If feedback occurs within your specified window, the POWER-Q automatically “tracks” it with the existing filter.

“FBX FILTER 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 SENSITIVITY to determine the authenticity of feedback and to discriminate musical tones from real feedback. Higher values of PERSISTENCE require more time for the POWER-Q to decide whether a given signal is feedback. For musical styles that may feature long sustained notes or tones (e.g., classical), set the PERSISTENCE value high (4 or 5, for example) to minimize the chance of mistaking the long sustain for real feedback. Set PERSISTENCE to 2 for spoken word applications. Range of values: 1-5.

“FBX FILTER SENSITIVITY” adjusts the POWER-Q’s sensitivity to 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 can be easily mistaken for feedback by the POWER-Q. Higher values of SENSITIVITY will allow feedback to grow larger in magnitude before it is detected and eliminated; too low a value can result in mistaking certain musical tones for feedback. Use the DEFAULT setting of 5 for most venues; use the value 2 for more classical venues and 4 for spoken word applications. Range of values: 1-10.

For ultra fast feedback detection during set-up, set PERSISTENCE and SENSITIVITY to their lowest values. For performance, experiment until you find the best setting for your application.

“BYPASS POWER-Q” routes the input signal directly to the output jack, completely bypassing the POWER-Q circuitry. This is a useful feature when comparing “before” and “after” POWER-Q settings. Be careful when placing the unit in bypass after setting FBX filters, as the feedback you’ve been eliminating may pay you an instant unwelcome visit. Note that when the POWER-Q is in bypass, “BPASS” flashes in the upper right corner of the display (in every window). Note also that the POWER-Q defaults to a hard-wire bypass when turned off.

“REF MIC PHANTOM POWER” makes +48v available at the reference microphone input for microphones that require phantom power.

“REF MIC 20dB PAD” lowers the gain (by 20 dB) of the reference microphone preamp in the event that the signal coming from the microphone is clipping the input level. NOTE: DO NOT use a microphone preamp or balanced line transformer with this input. This may cause the reference mic board to overheat.

COPY FUNCTION. The third soft key in the Global Parameters screen allows you to select a "COPY" option with the up/down arrow keys and implement it by pressing ENTER. You may:

1. Copy parameters from A to B. This option will instantaneously copy all parameters currently set for channel A to channel B. Any further adjustments made to either channel will be independent until COPY is executed again.
2. Copy parameters from B to A.   This option will instantaneously copy all parameters currently set for channel B to channel A. Any further adjustments made to either channel will be independent until COPY is executed again.

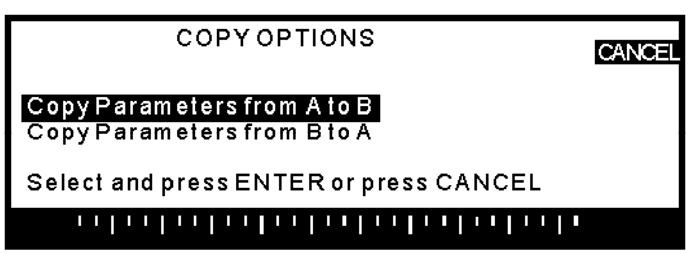


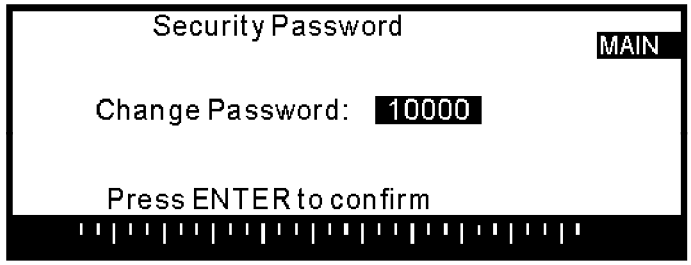
Fig. 52: COPY options screen

Canceling the COPY function will allow the POWER-Q to operate as an independent, two channel, "dual mono" unit.

Section 18: Password

To access the PASSWORD controls, press #11 (PASSWORD) from the POWER-Q MAIN MENU. The following window will appear:

Fig. 53: Change password screen



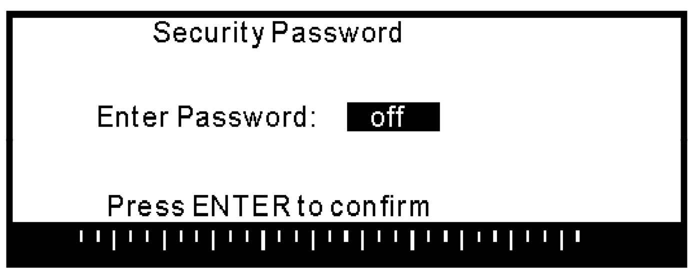
Security password protection will prevent unauthorized (or well-intentioned but unsophisticated) users from tampering with your POWER-Q set-up.

To change your personal security password from its default value "off," turn the data wheel clockwise one click. A five-digit number code will appear, initially set to 10000. The left/right arrow keys will move the cursor to any of the five digits, and the data wheel will change the value. Note that the changing values above or below zero increments or decrements all digits to the left of the cursor correspondingly. For example, moving the cursor to the far right digit of password 10000 and rotating the data wheel one click counterclockwise will change the password to 9999.

Once you have created the password you want, press ENTER, and the POWER-Q will remember your password and return to MAIN MENU. You may then proceed with normal POWER-Q operation.

When the POWER-Q is switched off and turned back on, the following window will appear:

Fig. 54: Enter password screen



To enter your password, turn the data wheel clockwise. The five-digit code (10000) will appear, and you must enter your password using the left/right arrow keys and data wheel. When you have scrolled to the correct password, hit ENTER. The POWER-Q will display the main menu.

Until the correct password is given, the POWER-Q front panel buttons will remain inoperative (with the exception of the HELP button). The POWER-Q will continue to operate and process audio using the parameters that were programmed in when the POWER-Q was turned off. You will be unable to change any of the settings until you enter the correct password.

If you forget your password, you may always gain access to the POWER-Q controls by entering the secret user backdoor password 13829. *(If you use this we'll have to kill you.)*

Section 19: POWER-Q Options

The POWER-Q is available in the following configurations:

- **Standard model:** Analog in/out.
- **Transformer analog I/O.** Analog connections with Jensen transformers.
- **Blank front panel:** For remote control via RS-232 serial port interface to Windows software or via MIDI.
- **Digital and analog I/O.** Adds an AES/EBU digital interface in addition to the standard analog connections.
- **Digital I/O only.** Replaces the analog connections with an AES/EBU digital interface.

Section 20: Remote Control of the POWER-Q

POWER-Q units with the Serial/MIDI remote control option, or blank front panel POWER-Qs (ADF-4SLU), can be remote controlled by a Windows-equipped computer. If your POWER-Q has this option, the back panel will have connectors labeled "Serial" and "Network". The Serial connector is for connecting the first POWER-Q to your computer. Additional POWER-Qs are connected together via the Network connector.

The Windows software necessary to operate the POWER-Q is included in your purchase of the Serial/MIDI option or the ADF-4SLU slave unit.

20.1 CONTROLLING POWER-Q FUNCTIONS WITH WINDOWS SOFTWARE.

When you order a POWER-Q with the Serial/MIDI Remote Control option, or you purchase an ADF-4SLU blank front panel verion, you will receive two 3.5” POWER-Q for Windows disks.
In order to use the software you will need the following:

System Requirements

1. Computer equipped with Pentium processor 100 Mhz or faster.
2. Hard disc with at least 1.5 MB of available space for program files.
3. Windows 3.1 or Windows 95.
4. SVGA or greater resolution graphic card and monitor.
5. One COMM port for a serial connection, with a 16550 or faster comm chip..
6. Cables & Connectors
See figures A & B for serial cable options, installation instructions and Pin to Pin map.

Cables & Connectors
If your computer has a 9-pin COMM port, use a standard 9-pin male to 9-pin female, RS232, standard computer-store connectors (Radio Shack Cat No 26-117). Plug into the POWER-Q’s back-panel RS232 male jack labeled SERIAL.

[Figure A below, is provided for your knowledge and convenience. No action is required for connectivity]

If you computer’s COMM port is has a 25-pin connector, use a standard RS232, 25-pin female to 9-pin male standard computer-store connector. Alternatively, use a 25-pin female to 9-pin male adapter (Radio Shack Cat No 26-287) and a standard 9-pin to 9-pin connector described above. Plug into the POWER-Q’s back-panel RS232 jack labeled SERIAL.

[Figure B below, is provided for your knowledge and convenience. No action is required for connectivity]

For connecting up to ten POWER-Qs, use standard 9-pin to 9-pin connectors described above. POWER-Q input is labeled SERIAL and the output is labeled NETWORK.the last unit in the chain back to the computer.

Installing POWER-Q for Windows:
1. Start Windows 3.1 or Windows 95.
2. Insert Remote Control disc into your disc drive.
3. Select the Windows 3.1 Program Manager FILE then RUN, or the Windows 95 start button.
4. Type A:SETUP and press ENTER. (NOTE: If you are using the B drive, substitute B for A).
5. Follow the instructions on the screen. You need only choose the directory where you want to install the POWER-Q program. The program suggests C:/ADF4000.
6. Now you have a Program group window called “ADF4000” and an icon called “ADF4000”.

Using POWER-Q for Windows:
1. Double-click on the ADF icon. Select the proper COMM port and then select “CONNECT POWER-Qs”.

2. All the POWER-Qs in the chain will be automatically connected to your computer. Each unit in the chain is represented by an icon at the top of the screen. The RTA/EQ screen of the first POWER-Q in the chain starts automatically. Select the icon of the POWER-Q you wish to monitor and edit.

3. This operating guide describes the features and controls of the POWER-Q from the front panel. POWER-Q for Windows closely mimics the POWER-Q front-panel. Select Main Menu from the tool bar at the top of the screen for a list of all the features. You can also use the feature icons or the Function keys to select various POWER-Q features.

4. You may regain front-panel control of any POWER-Q in the system by entering the password with the data wheel and arrow keys. POWER-Q for WIndows disconnects all units in the chain whenever front-panel control is regained on any unit.

20.2 POWER-Q MIDI CONTROL.

MIDI control of the POWER-Q will be implemented in a future firmware release.

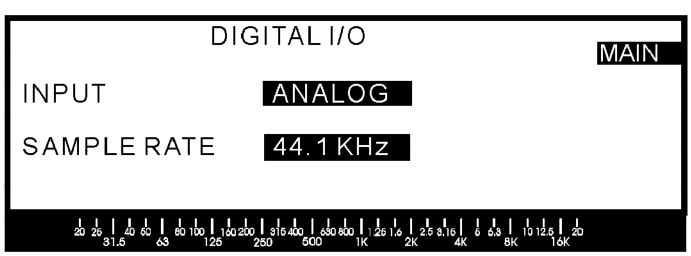
Section 21: Digital I/O Option

HANDS ON

To enable your POWER-Q Digital I/O option, your POWER-Q must have firmware 2.1 or higher, and must be equipped with the optional D-I/O board. If your unit is so equipped the POWER-Q back panel will feature AES/EBU digital in/out connectors. These are used to connect to other digital equipment in your signal path.

To set up your digital options, select #12 (“DIGITAL I/O”) from the POWER-Q MAIN MENU.

Fig. 55: Digital I/O screen



The POWER-Q will operate with either an analog or digital input source. Select the appropriate source using the arrow keys and data wheel. NOTE: when choosing a digital input, the POWER-Q's ClipGuard function is automatically disabled, so only the output level of the unit will be adjustable when using the controls found in the GLOBAL PARAMETERS screen.

Sample rate of the POWER-Q output is adjustable using the arrow keys and data wheel. Selectable samples rates include 32 KHz, 44.1 KHz, 48 KHz, or a match to the sample rate of the digital input source.

Regardless of the input chosen, the POWER-Q will provide simultaneous digital <u>and</u> analog outputs. However, please note one important consideration: when monitoring the digital output from an analog source, change the Clip Adjust setting in the GLOBAL PARAMETERS screen to "Manual" and adjust the POWER-Q gain structure to suit your application (see Section 17 for details about setting GLOBAL PARAMETERS).

Section 22: Using The ADF -- A Pro's Guide, With Ken Newman

Ken Newman’s career as a sound engineer encompasses over 20 years of working for performers such as Anita Baker, Barry Manilow, Chris Isaak, and many other artists who demand the best. Like his clients, Ken demands the best from his equipment, and his years of experience using Sabine ADF products have helped him earn ecstatic praise from performers and audience alike. Barry Manilow calls Ken “the finest sound mixer I’ve ever worked with.” Here are Ken’s own suggestions for using Sabine ADF equipment.

Ask concert-goers, “Was the sound good?” Chances are the average listener will base their response on a couple of factors. If the system is free of feedback, and the vocals are audible, most people regard this as “good sound,” with little or no consideration of the tonal balance or perspective of the mix. On the other hand, you could have the most happening mix in the world going, and if for some reason there are occasional bursts of feedback, the distraction and unpleasant nature of the squeals will earn your mix a very negative review. Likewise, if the audience members can’t hear what the singer is saying between songs, or understand the lyrics because you can’t get enough gain before feedback, then your mix will also be considered “bad sound.”

For many years, I used 1/3 octave graphic EQs to control feedback, but found this to be a compromise. Feedback is a result of peaks in the frequency response of the microphone, the electronics, the speaker system, and the room acoustics. These peaks are rather narrow, at least compared to the filters in a graphic EQ, which are based on 1/3 octave CENTERS, but actually are much wider than 1/3 octave (most often an octave wide). A graphic EQ’s fixed-point, octave-wide filters are too inexact and too global to carve out feedback without taking out a big chunk of music in the same slice.

I was introduced to the Sabine line of “feedback eliminators” a few years ago when I was working on Ann-Margret’s stage show with my friend John Reed. He had found that the FBX-900 afforded him a good deal more gain before feedback on Ann-Margret’s mic than what he had been able to attain manually in the past. And since she’s a quiet singer, this was an important breakthrough.

Since then, Sabine has continued to improve their whole line of FBX products, and with the introduction of the Sabine ADF (Adaptive Digital Filter) Workstations, the process of feedback control and gain maximization has become a great deal easier, more accurate, and more effective. Here’s a step by step guide that I’ve developed for using the ADF units to help get the best mix possible, full of clarity and free from feedback.

First, the ADF needs to have its system parameters set correctly. On the global parameters page, I set “Threshold” to a high number, so that the unit is more sensitive to feedback because it requires less harmonic content. Then I set “Persistence” to a low number, so that less time is needed for the unit to detect feedback. I set “Bandwidth” to 1/10 octave (a good starting point) and set “Maximum Cut” to 10 dB...so the most filtering that can occur is a very narrow 10dB cut.

Next, I begin the automatic “ringing out” of the PA (assuming the sound system is all set-up and tonally balanced), with the mic most likely to need the greatest gain positioned on stage. I set all of the ADF filters to “P” (parametric) except one, which I set to “D” (dynamic FBX), so that it will catch the first feedback. Then I boost the mic gain until it starts to actually feed back, and voila! the ADF catches the feedback, and makes a cut (only as deep as necessary to eliminate feedback at that mic volume) at the EXACT frequency of the feedback with the filter I had set to “D.” I then back off the mic gain, switch the “D” filter to “P,” and check the depth of the filter cut. If it’s deeper than 3-5 dB, I usually change it to 3 dB for starters.

I then move on to the next ADF filter, setting it to “D,” and repeat the process until I’ve caught the first 5 or 6 ringing frequencies (maybe more, in a tough situation). Then I examine the frequencies of the filters put in place automatically by the ADF. If it turns out that some of them are very close together, I’ll pick an in-between frequency and widen a filter there, eliminating others close by, in order to place fewer filters in the signal path. Also, since 1/10 of an octave is a very narrow filter, I sometimes widen other filters as well, especially considering that I’ve made my tests with a stationary mic. As soon as the mic is moved, the feedback frequencies will shift, and I want to cover that situation as well. Besides experimenting with widening filters, I also set an unused ADF filter to “D” for the show, to catch feedback that may occur under performance conditions. The beauty of the Sabine FBX filters is how well they detect feedback even when music is playing!

After I’ve completed the ringing out process, I’ll try and use the ADF’s integral delay and gate to further assist me in achieving the greatest gain before feedback. With the gate set at a very low threshold, but above the room ambience, the sound of a podium mic or omnidirectional lavaliere can often be improved. And by delaying the signal ever so slightly (perhaps 10 milliseconds), another slight increase in gain before feedback (especially at low frequencies) can be realized.

The ADF has become an indispensable tool in my quest for the maximum gain before feedback, and I’m sure you’ll find the same results.

Section 23: Troubleshooting Tips

In the unlikely event you should experience trouble with the unit, here are some suggestions about what might be wrong. Some of these are pretty obvious, but so sometimes are the solutions! For additional assistance, call the Sabine Customer Service Department at (904) 418-2000, Monday through Friday, 9:30 a.m to 5:30 p.m. Eastern.

PROBLEM	SUGGESTION
NO AUDIO COMING FROM POWER-Q OUTPUT	Check connections. Are input and output reversed? Is the POWER-Q LED showing signal? If no, make sure the unit is not in BYPASS mode, and that audio signal is feeding POWER-Q input. If yes, check connections and gain downstream from POWER-Q.
AUDIO CUTS ON AND OFF	Check POWER-Q noise gate settings. Check connections for intermittence.
AUDIO "PUMPS"	Check POWER-Q compression settings.
SPEAKER STACKS PLAY AUDIO OUT OF SYNC	Check POWER-Q delay settings.
FEEDBACK NOT BEING REMOVED	Check filter availability. Check "Persistence" and "Sensitivity" settings. Make sure unit is not in BYPASS.
AUTOMATIC ROOM EQ DOES NOT WORK	Check reference mic, mic cable, connections, and phantom power. Make sure audio signal is playing through the POWER-Q.
POWER-Q APPEARS TO BE CATCHING FEEDBACK, BUT FEEDBACK STILL PRESENT	See Section 7.2. If you have the POWER-Q patched in an effects or auxiliary loop, you will only catch the feedback in the effects loop, and not the mixer input channel. Or, you may have used up all the available FBX filters, leaving no additional filters for new feedback frequencies.
FEEDBACK FREQUENCIES CLUSTERED TOGETHER	Try using a graphic equalizer to "flatten" the room. There may be a big frequency "bump" in a room with less than ideal acoustics; this is better treated with a wider filter.
DISTORTED AUDIO	Most likely you are pushing a VERY HOT signal into the box. It's hard to make the POWER-Q clip. Check connections for intermittence, or check downstream from the POWER-Q. Check the POWER-Q "Digital Clip Level" in the "Global Parameters" window and turn it up, or adjust "Clip Adjust" to "AUTO." ALTERNATIVELY: The POWER-Q may still be in "TURBO" mode, which automatically maximizes the Clip Level until the first dynamic FBX filter is set. You may exit "TURBO" mode in several ways (see section 11.2).
NOISY AUDIO	Bypass the POWER-Q. If noise is still there, it's not the POWER-Q. If noise goes away, check the POWER-Q "Digital Clip Level" and turn it up, or adjust Clip Adjust to "CLIPGUARD."
AUTOMATIC ROOM EQ POOR RESULTS	Check the reference microphone you're using and make sure it's flat. Consider the reference microphone position and experiment.
YOU SPEND TOO MUCH TIME READING MANUALS	Take up a useful hobby like stamp collecting. Get out more often. Be glad you don't have to write one.

Section 24: Engineering Specifications

FBX/Parametric Filters

Twelve independent digital notch filters per channel, controlled automatically or parametrically from 20 Hz to 20 KHz, each switchable between FBX fixed filters, FBX dynamic filters and parametric filters

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); max. automatic depth adjustable from -6 to -80dB

Filter width: user-controllable from 9.99 octave to .01 octave (parametric), 1.0 to .01 octave (FBX)

High pass filter, user-controllable in 1 Hz steps between 20Hz and 3KHz; 12 dB/octave roll-off

Low pass filter, user-controllable in 1 Hz steps between 1 KHz and 20 KHz; 12 dB/octave roll-off

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

Time required to find and eliminate feedback: user-controllable from 0.2 seconds to 1 second (typically 0.3 seconds).

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

Filters controllable via table or graphic interface

Graphic Equalizer

31 digital filters on ISO center frequencies, width selectable from .5 to 1 octave in .01 octave increments, +12 to -15 dB boost & cut

Independent display and control or A & B channels, LINK or COPY

Real-Time Analyzer

31 band, 20 Hz — 20 KHz on ISO center frequencies

A, B, C or flat weighting

Fast/slow, peak/hold

Source selectable: reference mic, channels A or B, input or output

Reference mic input: ISO phantom power, +48VDC @ 10mA, 1.2K Ohm impedance

Compressor/Limiter

Threshold: +32dBu to -30dBu in 0.5dB steps
Ratio: 1:1, 1.4, 2, 3, 4, 5, 6, 7, 8, 9, 10, 16, 32, infinite
Knee: 1 (hardest) to 40 (softest)
Attack: 1.0 to 99 msec in 1 msec steps
Release: .05 to 5 sec in .05 sec steps

Expander/Noise Gate

Threshold: -20dBu to -90.0dBu in 0.5dB steps
Knee: 1 (hardest) to 40 (softest)
Attack: 1 to 99 msec in 1 msec steps
Release: .05 to 5 sec in .05 sec steps

Digital Delay

1.38 - 83.28 milliseconds/channel in 20 microsecond steps

Programmable in feet or meters.

Password Configuration

5 numeric characters

Load & Recall Configurations & Response Curves

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

Front Panel

LCD Display

Clip, limit, signal and gate LEDs for channels A and B; clip and signal LEDs for REF; MIDI, Serial & Digital LED indicators

Four menu/soft keys

More, Help & Enter keys

Data wheel and cursor keys

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 +26 dBV peak
Max. output load: 600 Ohms balanced
Bypass: true power-off bypass
I/O connectors: XLR-3

Performance*

Frequency response: 10Hz to 20KHz, 0.2dB @ +22dBV
SNR**: > 105 dB (with ClipGuard)
THD: < 0.01% @ 22 dBV at 1 KHz
Dynamic range: > 110 dB (with ClipGuard)
Headroom: +22dB peak @ 4dBV nominal input

Power

50/60Hz available in 100V, 120V, 230V; 25W

Dimensions

2-U rack mount 19 x 3.5 x 9 in. (48.3 x 9 x 22.9 cm); 9 lb. (3.9 Kg)

Options

ADF-4SLU: Blank front panel two-channel slave unit for remote control with your computer and POWER-Q for WIndows
DA-I/O: AES/EBU Digital I/O (add to standard analog)
D-I/O: AES/EBU Digital I/O only (no analog)
SMR-I/O: Serial (RS232) full remote control with POWER-Q for Windows; MIDI for loading presets and output level only
Jensen Balanced Line I/O Isolation Transformers

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 or the maximum undistorted signal by specification (26dBV RMS sinewave) to the noise floor

One-year limited warranty

Section 25: Cautions & Warranty

FCC Statement:
Warning: Changes or modifications to this unit not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.
NOTE: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio TV technician for help.

This digital apparatus does not exceed the Class B limits for radio noise emissions from digital apparatus set out in the Radio Interference Regulations of the Canadian Department of Communications.

Le present appareil numerique n'emet pas de bruits radioelectriques depassant les limites applicables aux appareils numeriques de la class B prescrites dans le Reglement sur le brouillage radioelectrique edicte par le ministere des Communications du Canada.

Safety Information:
Warning! This equipment must be earthed and placed in a rack with adequate ventilation.
Caution! Risk of electric shock. Do not open.
Caution! Shock hazard. Do not remove covers. No user serviceable parts inside. Refer servicing to qualified service personnel.
Warning! To reduce the risk of fire or electric shock, do not expose this product to rain or moisture.

Attention! Cet appareil doit être relié à la terre.
Attention! Risque de choc électrique; ne pas ouvrir.
Attention! Risque de choc; ne pas oter les capots. Aucune pièce accessible à l'intérieur. S'addresser à un technicien qualifié.
Attention! Pour réduire le risque d'incendie ou de choc électrique, ne pas laisser l'appareil sous la plouie ou à l'humidité.

Achtung! Dieses Gerät muss schutzgeerdet sein.
Achtung! Gefar eines elektrischen Stormschlags. Gehause nicht öffnen.
Achtung! Gefar eines elektrischen Stormschlags. Gehäuse nicht öffnen. Keine con Benutzer zu bedienenden Teile im Geräteinneren.
Überlassen Sie das Gerät zu Servicezwecken nur geschultem Fachpersonal.
Um Brandgefar oder das Risiko eines elektrischen Schlags auszuschließen, das Gerät vor Nässe und Feuchtigkeit schützen.

Advertencia! Este equipo debe estar conectado a tierra.
Precaución! Riesgo de descarga eléctrica. No abrir.
Precaución! Riesgo de descarga eléctrica. No desmontar las tapas. Piezas interiores no reparables por el usuario. Reparable sólo por personal cualificado.
Advertencia! Para reducir el riesgo de incendio o de descarga eléctrica no exponga este producto a la lluvia o humedad.

Be sure to read the cautions above the warranty statement on the next page. All unauthorized repairs or modifications void the warranty. In North America, contact Sabine for service. In other countries, contact your authorized Sabine distributor for service.

Made in USA.

CAUTION

EXPOSURE TO EXTREMELY HIGH NOISE LEVELS MAY CAUSE A PERMANENT HEARING LOSS. INDIVIDUALS VARY CONSIDERABLY IN SUSCEPTIBILITY TO NOISE INDUCED HEARING LOSS, BUT NEARLY EVERYONE WILL LOSE SOME HEARING IF EXPOSED TO SUFFICIENTLY INTENSE NOISE FOR A SUFFICIENT TIME. THE U.S. GOVERNMENT'S OCCUPATIONAL SAFETY AND HEALTH ADMINISTRATION (OSHA) HAS SPECIFIED THE FOLLOWING PERMISSIBLE NOISE LEVEL EXPOSURES:

DURATION/DAY IN HOURS	SOUND LEVEL IN dBA, SLOW RESPONSE
8	90
6	92
4	95
3	97
2	100
1-1/2	102
1	105
1/2	110
1/4 or less	115

ACCORDING TO OSHA, ANY EXPOSURE IN EXCESS OF THE ABOVE PERMISSIBLE LIMITS COULD RESULT IN HEARING LOSS. EAR PLUGS OR PROTECTORS IN THE EAR CANALS OR OVER THE EARS MUST BE WORN WHEN OPERATING THIS DEVICE IN ORDER TO PREVENT A PERMANENT HEARING LOSS, IF EXPOSURE IS IN EXCESS OF THE LIMITS AS SET FORTH ABOVE. TO ENSURE AGAINST POTENTIALLY DANGEROUS EXPOSURE TO HIGH SOUND PRESSURE LEVELS, IT IS RECOMMENDED THAT ALL PERSONS EXPOSED TO EQUIPMENT CAPABLE OF PRODUCING HIGH SOUND PRESSURE LEVELS SUCH AS THIS DEVICE BE PROTECTED BY HEARING PROTECTORS WHILE THIS UNIT IS IN OPERATION.

1. Read all safety and operating instructions before using this product.
2. All safety and operating instructions should be retained for future reference.
3. Obey all cautions in the operating instructions and on the unit.
4. All operating instructions should be followed.
5. This product should not be used near water, i.e a bathtub, sink, swimming pool, wet basement, etc.
6. This product should be located so that its position does not interfere with its proper ventilation. It should not be placed flat against a wall or placed in a built-in enclosure that will impede the flow of cooling air.
7. This product should not be placed near a source of heat such as a stove or radiator.
8. Connect only to a power supply of the type marked on the unit adjacent to the power.
9. Never break off the ground pin on the power supply cord.
10. Power supply cords should always be handled carefully. Never walk or place equipment on power supply cords. Periodically check cords for cuts or signs of stress, especially at the plug and the point where the cord exits the unit.
11. The power supply cord should be unplugged when the unit is to be unused for long periods of time.
12. Care should be taken so that objects do not fall and liquids are not spilled into the unit through the ventilation holes or any other openings.
13. This unit should be checked by a qualified service technician if:
    A. The power supply cord or plug has been damaged.
    B. Anything has fallen or been spilled into the unit.
    C. The unit does not operate correctly.
    D. The unit has been dropped or the enclosure damaged.
14. The user should not attempt to service this equipment. All service work should be done by a qualified service technician.

Limited Warranty

THIS LIMITED WARRANTY VALID ONLY WHEN PURCHASED AND REGISTERED IN THE UNITED STATES OR CANADA. ALL EXPORTED PRODUCTS ARE SUBJECT TO WARRANTY AND SERVICES TO BE SPECIFIED AND PROVIDED BY THE AUTHORIZED DISTRIBUTOR FOR EACH COUNTRY.
Ces clauses de garantie ne sont vaiables qu'aux Etats-Unis et au Canada. Dans tous les autres pays, les clauses de garantie et de maintenance sont fixees par le distributeur national et assuree par lui selon la legislation en vigueur.
Diese Garantie ist nur in den USA and Kanada gultig. Alle Export-Produkte sind der Garantie und dem Service des Importeurs des jewelligen Landes untervorfen.
Esta garantia es valida solamente cuando el producto es comprado en E.U. continentales o en Canada. Todos los productos que sean comprados en el extranjero, estan sujetos a las garantias y servicio que cada distribuidor autorizado determine y otrezca en los diferentes paises.

ONE-YEAR LIMITED WARRANTY/REMEDY
SABINE, INC. ("SABINE") warrants this product to be free from defects in material and workmanship for a period of one (1) year from date of purchase PROVIDED, however, that this limited warranty is extended only to the original retail purchaser and is subject to the conditions, exclusions and limitations hereinafter set forth:
CONDITIONS, EXCLUSIONS AND LIMITATIONS OF LIMITED WARRANTIES
These limited warranties shall be void and of no effect if:
a. The first purchase of the product is for the purpose of resale; or
b. The original retail purchase is not made from an AUTHORIZED SABINE DEALER; or
c. The product has been damaged by accident or unreasonable use, neglect, improper service or maintenance, or other causes not arising out of defects in material or workmanship; or
d. The serial number affixed to the product is altered, defaced or removed; or
e. The power supply grounding pin is removed or otherwise defeated. In the event of a defect in material and/or workmanship covered by this limited warranty, Sabine will repair the defect in material or workmanship or replace the product, at Sabine's option; and provided, however, that, in any case, all costs of shipping, if necessary, are paid by you, the purchaser.
THE WARRANTY REGISTRATION CARD SHOULD BE ACCURATELY COMPLETED, MAILED TO AND RECEIVED BY SABINE WITHIN FOURTEEN (14) DAYS FROM THE DATE OF YOUR PURCHASE.
In order to obtain service under these warranties, you must:
a. Bring the defective item to any Authorized SABINE DEALER and present therewith the ORIGINAL PROOF OF PURCHASE supplied to you by the AUTHORIZED SABINE DEALER in connection with your purchase from him of this product. If the DEALER is unable to provide the necessary warranty service, you will be directed to the nearest other SABINE AUTHORIZED DEALER which can provide such service.
OR
b. Ship the defective item, prepaid, to:
SABINE, INC.
13301 HIGHWAY 441
ALACHUA, FL 32615-8544
including therewith a complete, detailed description of the problem, together with a legible copy of the original PROOF OF PURCHASE and a complete return address. Upon Sabine's receipt of these items:
If the defect is remedial under the limited warranties and the other terms and conditions expressed have been complied with, Sabine will provide the necessary warranty service to repair or replace the product and will return it, FREIGHT COLLECT, to you, the purchaser.
Sabine's liability to the purchaser for damages from any cause whatsoever and regardless of the form of action, including negligence, is limited to the actual damages up to the greater of $500.00 or an amount equal to the purchase price of the product that caused the damage or that is the subject of or is directly related to the cause of action. Such purchase price will be that in effect for the specific product when the cause of action arose. This limitation of liability will not apply to claims for personal injury or damage to real property or tangible personal property allegedly caused by Sabine's negligence. Sabine does not assume liability for personal injury or property damage arising out of or caused by a non-Sabine alteration or attachment, nor does Sabine assume any responsibility for damage to interconnected non-Sabine equipment that may result from the normal functioning and maintenance of the Sabine equipment.
UNDER NO CIRCUMSTANCES WILL SABINE BE LIABLE FOR ANY LOST PROFITS, LOST SAVINGS, ANY INCIDENTAL DAMAGES OR ANY CONSEQUENTIAL DAMAGES ARISING OUT OF THE USE OR INABILITY TO USE THE PRODUCT, EVEN IF SABINE HAS BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES.
THESE LIMITED WARRANTIES ARE IN LIEU OF ANY AND ALL WARRANTIES, EXPRESS OR IMPLIED, INCLUDING BUT NOT LIMITED TO, THE IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR USE; PROVIDED, HOWEVER, THAT IF THE OTHER TERMS AND CONDITIONS NECESSARY TO THE EXISTENCE OF THE EXPRESS LIMITED WARRANTIES, AS HEREINABOVE STATED, HAVE BEEN COMPLIED WITH, IMPLIED WARRANTIES ARE NOT DISCLAIMED DURING THE APPLICABLE ONE-YEAR PERIOD FROM DATE OF PURCHASE OF THIS PRODUCT.
SOME STATES DO NOT ALLOW LIMITATION ON HOW LONG AN IMPLIED WARRANTY LASTS, OR THE EXCLUSION OR LIMITATION OF INCIDENTAL OR CONSEQUENTIAL DAMAGES, SO THE ABOVE LIMITATIONS OR EXCLUSIONS MAY NOT APPLY TO YOU. THESE LIMITED WARRANTIES GIVE YOU SPECIFIC LEGAL RIGHTS, AND YOU MAY ALSO HAVE OTHER RIGHTS WHICH MAY VARY FROM STATE TO STATE.
THESE LIMITED WARRANTIES ARE THE ONLY EXPRESS WARRANTIES ON THIS PRODUCT, AND NO OTHER STATEMENT, REPRESENTATION, WARRANTY OR AGREEMENT BY ANY PERSON SHALL BE VALID OR BINDING UPON SABINE.
In the event of any modification or disclaimer of express or implied warranties, or any limitation of remedies, contained herein conflicts with applicable law, then such modification, disclaimer or limitation, as the case may be, shall be deemed to be modified to the extent necessary to comply with such law.
Your remedies for breach of these warranties are limited to those remedies provided herein, and Sabine gives this limited warranty only with respect to equipment purchased in the United States of America.

INSTRUCTIONS-WARRANTY REGISTRATION CARD
1. Mail the completed WARRANTY REGISTRATION CARD to:
SABINE, INC.
13301 HIGHWAY 441
ALACHUA, FL 32615-8544
a. Keep the PROOF OF PURCHASE. In the event warranty service is required during the warranty period, you will need this document. **There will be no identification card issued by Sabine, Inc.**
2. IMPORTANCE OF WARRANTY REGISTRATION CARDS AND NOTIFICATION OF CHANGES OF ADDRESS:
a. Completion and mailing of WARRANTY REGISTRATION CARDS - Should notification become necessary for any condition that may require correction, the REGISTRATION CARD will help ensure that you are contacted and properly notified.
b. Notice of address changes - If you move from the address shown on the WARRANTY REGISTRATION CARD, you should notify Sabine of the change of address so as to facilitate your receipt of any bulletins or other forms of notification which may become necessary in connection with any condition that may require dissemination of information or correction.
3. You may contact Sabine directly by telephoning (904) 418-2000.
4. Please have the Sabine product name and serial number available when communicating with Sabine Customer Service.

SABINE™ ADAPTIVE AUDIO

www.Sabine.com

Manufactured by: Sabine, Inc. • 13301 Highway 441 • Alachua, FL 32615-8544 USA • Phone: (904)418-2000 • Fax: (904)418-2001