ASHLY

SC-68 PARAMETRIC NOTCH FILTER

OPERATING INSTRUCTIONS

ASHLY AUDIO INC. 100 FERNWOOD AVE. • ROCHESTER, N.Y. 14621
(716) 544-5191 • TOLL FREE (EXCEPT N.Y. STATE) (800) 828-6308

TABLE OF CONTENTS

control clockwise is equivalent to moving the notch up the keyboard, and counterclockwise moves it down.

The Bandwidth control (Octaves) determines the number of notes, or range of frequencies, which will be included in the notch. This is shown in the illustration as a bracket encompassing roughly 2 1/2 notes on the keyboard, since the Octaves control is set at .3, a fairly broad notch. Turning the control clockwise would make the notch progressively narrower, so that at its narrowest setting of .03 Octaves, less than one note of the scale would be effected.

As noted earlier, each of the 8 filters is identical except for the range of frequencies it is designed to cover. Any filter may be adjusted as desired without regard to the other filters, with one exception: instead of setting two sharp filters to nearly the same center frequencies, it is preferable to use just one filter at the desired frequency and use a slightly broader bandwidth setting. Some people make the mistaken assumption that if one filter set to 3 kHz can produce a 50 dB notch, then 2 filters set to 3 kHz should produce a notch of 100 dB. In fact, it doesn't work that way, but even if it did that would be missing the point. Each filter by itself is capable of producing a notch accurate enough and deep enough to satisfy any real-world requirement. Setting two filters to the same frequency simply isn't necessary.

SETUP SYSTEM CONTROLS

Because a parametric notch filter is particularly useful as an anti-feedback tool in sound reinforcement applications, a special setup system has been incorporated into the SC-68 to make the sound man's job even easier. Essentially, the system consists of a limiter which holds feedback at a listenable low level, switchable listening modes to compare filtered and flat audio, and a meter which indicates, in dB, the improvement in system headroom which has been achieved. As a whole, the SC-68 system is somewhat comparable to having a separate limiter-compressor patched into a parametric equalizer, except that in the SC-68 both the limiter and EQ parameters are optimized for the specific job of feedback control.

The operating mode of the SC-68 is selected by means of push-button switches next to the meter. Only one mode is used at a time. In the SETUP mode, the Setup system is activated. In the OPERATE mode, the entire setup system is disabled and the SC-68 functions as a straightforward notch filter.

A brief description of the operating controls is given in Figure 7. (On the next page)

PEAK LED

The SC-68 peak indicator will flash on whenever signal levels in excess of +14 dBV are present. With an overall system headroom of +20 dBV, this light will warn you when you are within 6 dB of clipping. The peak detection circuit monitors the circuit at 10 individual points, including the input, output, and all 8 filters. Since the SC-68 functions at unity gain or at a loss, a flashing peak light will indicate that high-level signals are occuring somewhere in the audio chain prior to the SC-68.

FILTER CONTROLS

The SC-68 uses 8 filter secions, with the only difference between filters being the range of frequencies spanned by the center frequency (Hz) control. Once you understand one filter section, you will understand them all.

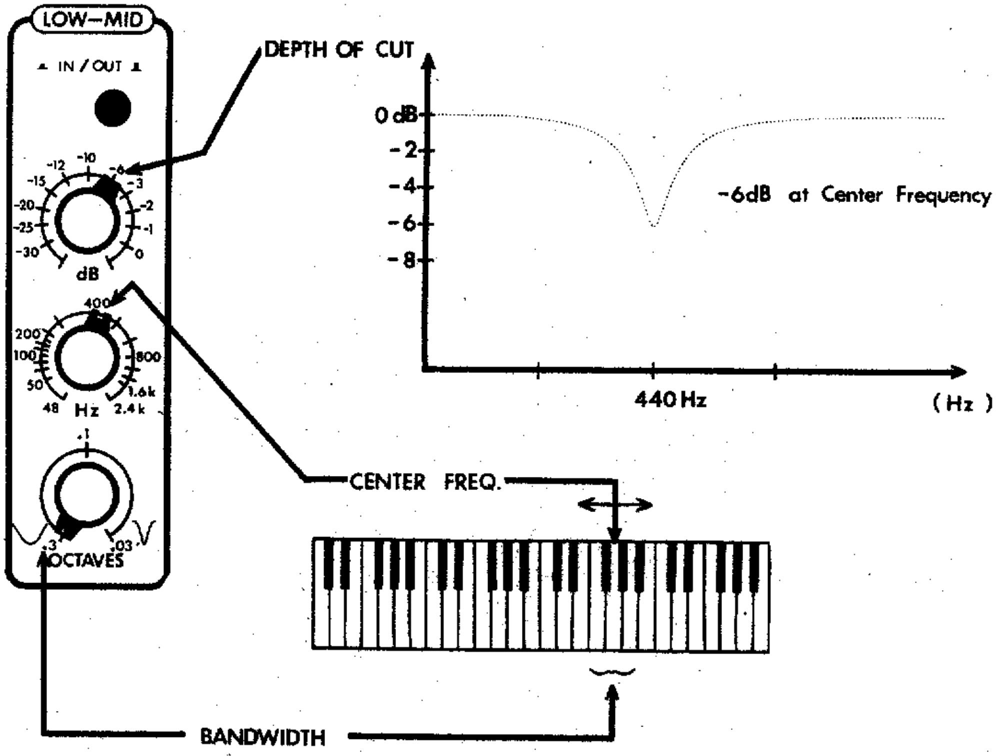
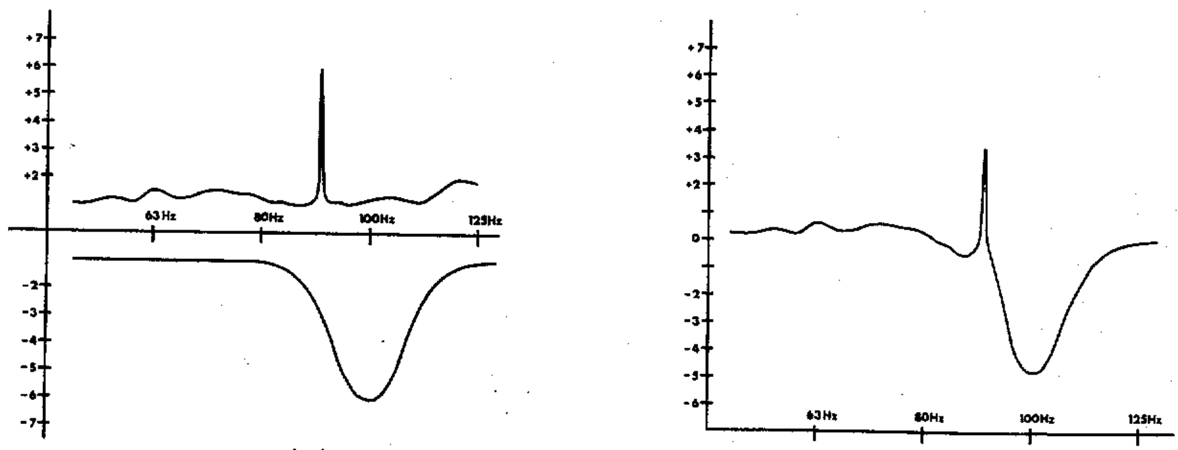


Figure 6 Typical filter section, SC-68

Referring to the above illustration of a typical filter from the SC-68, the IN/OUT switch allows quick A/B comparisons of filtered and flat audio signals for each filter section, making it easy to test the effect of any filter without disturbing the other filters.

Immediately below the switch, the Amplitude (dB) control determines the depth of a notch at the center frequency. Its effect is shown on the accompanying graph, in which a notch is centered at about 440 Hz, and produces a cut of 6 dB at the center freqency, corresponding to the -6 setting of the Amplitude control. Turning the control clockwise will decrease the depth of the notch, eventually obviating the effect of the filter when it is set to "0". Turning the control counter-clockwise will make the notch deeper, producing a more pronounced cut at the center frequency.

The Center Frequency (Hz) control selects the frequency at which the notch will be deepest. Frequencies above and below the center frequency will be proportionately less affected than the center frequency. Frequencies which are outside the range of the filter will be unaffected. In this illustration, the center frequency control is set to approximately 440 Hz, which is shown on the keyboard to be equivalent to the note A above middle C. Turning the Hz

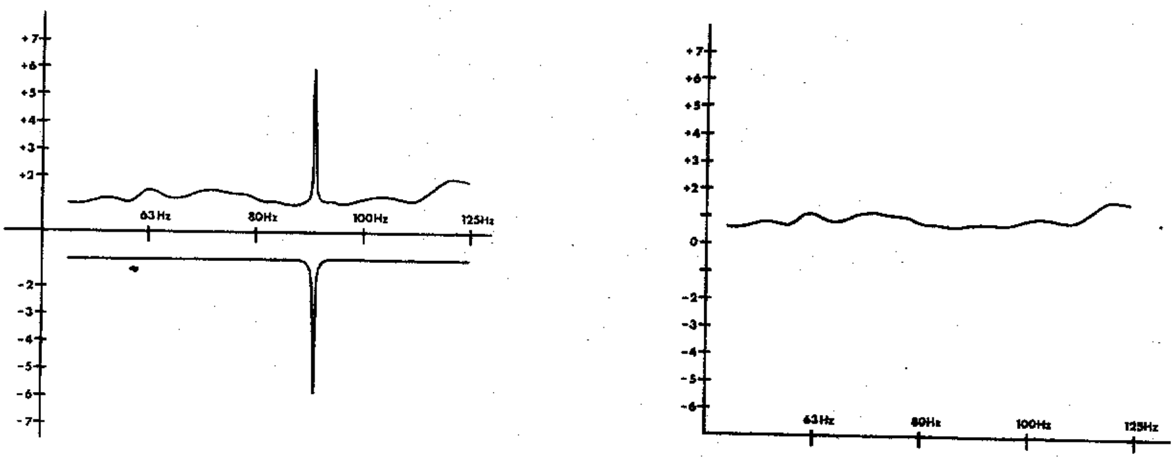


(a) (b)

Figure 4 Graphic Equalizer

In Figure (a), the curve above the axis represents the frequency response of a typical problem room, exhibiting a resonance at 92 Hz. The curve below the line represents a cut that could be made by a typical 1/3 octave graphic EQ to try to null the 92 Hz resonance.

As can be seen from the composite response curve shown in (b), the 1/3 octave graphic has succeeded only partially in removing the 92 Hz resonance, while needlessly inserting a dip in the 100 Hz response of the system.



(a) (b)

Figure 5 Parametric Notch Filter

In Figure (a), the curve above the axis again represents the frequency response of a room with a 92 Hz resonance. The curve below the axis represents the notch which can be created with an SC-68, which is an exact reciprocal of the room resonance, and which will have the desired effect of nulling the resonance.

The result of the correctly tuned notch on the room response is shown above. The problem has been fixed, and no unwanted side effects have been created.

EXPLANATION OF PARAMETRIC NOTCH FILTERS AND THE SC-68'S CONTROLS

Types Of Equalizers Used In Audio

An equalizer is a filter or a group of filters working together to shape the frequency response of an audio system. The simplest filter is one that will produce a fixed amount of gain or loss (boost/cut) over a fixed range of frequencies (bandwidth). The filter will generally have the greatest effect in the middle of that band of frequencies which are being affected (center frequency). Since all of these circuit parameters are pre-determined and fixed, the filter's applications will obviously be limited to performing specific functions.

Graphic Equalizers were the first universal tone controls, providing a piece-wise approximation of total frequency response. All of these equalizers have a common limitation: some of the characteristics of equalization are fixed. For example, the center frequency and sharpness for each band of a graphic are pre-determined. This leads to an immediate frustration because these characteristics are never exactly appropriate. (What do you do when you need a fader right between two sliders on a graphic?)

In the late sixtys, the first parametric equalizers were developed by George Massenburg at ITI. Parametrics provide independent and continuous adjustment of all three possible characteristics: amplitude, center frequency, and bandwidth. As a result, virtually any desired frequency response may be obtained with no restrictions imposed by the equalizer itself.

A parametric notch filter is further distinguished from other equalizers in that its amplitude adjustment is a cut-only type, and it is capable of much more attenuation at the center frequency (greater than 30 dB) than is a typical parametric equalizer (12-15 dB). The SC-68 consists of 8 individual filter circuits which are applied to a common buss. Each filter acts as a frequency dependent resistance with the resistance dropping to zero at the center frequency. In effect, the audio signal is shorted to ground at the center frequency. This enables each band to operate as a nearly infinite cut. Each filter can be individually switched on and off the buss to test its effect, and the entire unit can be bypassed to compare the final setup to a flat setting.

WHAT CAN THE SC-68 DO THAT A 1/3 OCTAVE GRAPHIC CAN'T?

The adjustable center frequency and extremely narrow bandwidth offered by the SC-68 make it more useful than a 1/3 octave graphic equalizer for removing feedback and room resonances. Each band of a graphic is set to a fixed center frequency, which may or may not be close to the feedback frequency you are trying to correct. A parametric notch filter, on the other hand, lets you tune to precisely the frequency in question, without influencing any more of the frequency spectrum than necessary.

This is represented in Figures 4 and 5 on the next page.

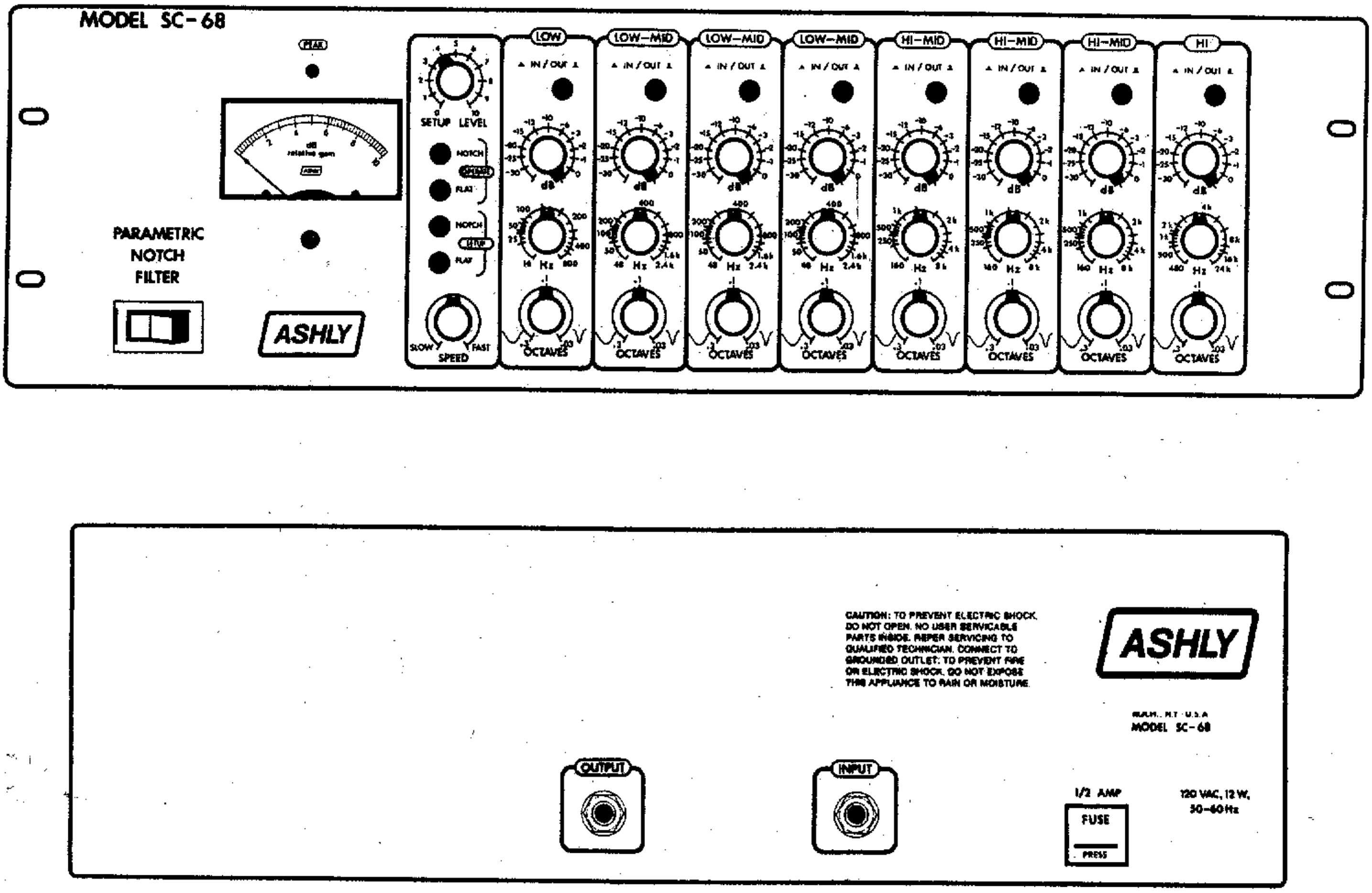


Figure 1. SC-68 Front & Back Panels

INPUT, OUTPUT, AND POWER CONNECTIONS

This equalizer should be connected to a 3 wire grounded outlet supplying 120 Volts, 50-60 Hz. Power consumption is 12 watts.

The INPUT is a 10K ohm active balanced type on a standard stereo phone plug. The (+) or in-phase connection is on the tip and the (-) or out-of-phase connection is on the ring. When feeding the equalizer from unbalanced sources, connect the signal hot to the tip (+) and the signal ground to the ring ( ). To use the input as a common unbalanced type, simply use a mono phone plug in the usual way. (See Definition Of Terms, "Wiring", page 13.)

The OUTPUT is a low impedance type (typically about 50 ohms). This should be terminated in a load of 600 ohms or higher to realize maximum headroom. The output can be used as a balanced type by using a stereo phone plug wired in the same manner as the input or as an unbalanced type by using a mono phone plug.

The SC-68 is a unity-gain device (except for the notches in the frequency response) and has a headroom of +20 dBV. The nominal operating level is 0 dBV. It can be inserted anywhere in the signal path between the mixer and the crossover (or power amp in passive crossover or full-range systems).

INTRODUCTION

The SC-68 Parametric Notch Filter is a cut-only equalizer which is capable of producing relatively sharp dips in the frequency response of an audio system. It is useful for notching out narrow segments of the audio frequency spectrum, thereby reducing problems such as feedback, cabinet resonances, hum, video sync buzz, and single hot notes on musical instruments. The SC-68 is useful in professional sound reinforcement, recording, and broadcast applications, with audio performance to meet the most exacting requirements.

The SC-68 includes a setup system to aid an operator in eliminating feedback in sound reinforcement. Standard procedure in "tuning" a room is to purposely induce system feedback and then equalize the system at the predominant feedback frequency. When the SC-68 is used in the SETUP mode, it automatically increases system gain until feedback occurs, and then holds the feedback at a controlled low level while the operator identifies and notches out the offending frequency. When that has been done, the limiter again increases system gain, causing controlled feedback at new frequencies. These can then be notched out in the same manner. As this process continues, the built-in gain meter indicates directly the improvement in gain-before-feedback (headroom). The meter can also be used to show the effect of such factors as mic and speaker placement or movement of objects on the stage. As a result, tuning for minimum feedback becomes a precise process without the speaker damage and ear abuse which often occur with conventional means.

The Ashly Parametric Notch Filter is capable of much narrower and deeper notches in the audio frequency spectrum than can be achieved with a typical parametric or graphic equalizer. Because it exerts its maximum cut at precisely the problem frequency, its net effect on the perceived audio response of a system is negligible; A/B comparison of "flat" and "notched" signals sounds identical except for the absence of feedback and musical "hot spots". Low noise and distortion add to the audio transparency of the SC-68.

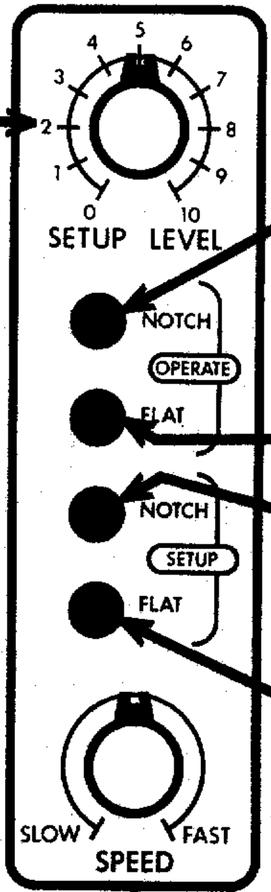
As powerful and flexible a tool as the SC-68 is, it is actually quite easy to use once its functions are understood. Please read this manual carefully in order to take full advantage of the SC-68's capabilities.

UNPACKING

As a part of our system of quality control every Ashly product is carefully inspected before leaving the factory to ensure flawless appearance. After unpacking, please inspect for any physical damage. Save the shipping carton and all packing materials, as they were carefully designed to reduce to a minimum the possibility of transportation damage should the unit again require packing and shipping. In the event that damage has occurred, immediately notify your dealer so that a written claim to cover the damages can be initiated.

THE RIGHT TO ANY CLAIM AGAINST A PUBLIC CARRIER CAN BE FORFEITED IF THE CARRIER IS NOT NOTIFIED PROMPTLY AND IF THE SHIPPING CARTON AND PACKING MATERIALS ARE NOT AVAILABLE FOR INSPECTION BY THE CARRIER. SAVE ALL PACKING MATERIALS UNTIL THE CLAIM HAS BEEN SETTLED.

SETUP LEVEL CONTROL. When the SC-68 is used in the SETUP mode, the system will be induced to feed back, and the built in limiter will hold the feedback at a controlled low level while the operator determines the problem frequency and works on notching it out. The SETUP LEVEL control allows you to control the actual listening level of the feedback in the room, from moderately loud to very quiet. Adjust it to suit your taste and then leave it at that setting.



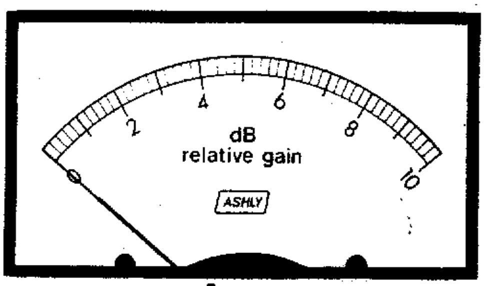
OPERATING MODE SWITCHES

Normal operating mode, with filters active.

Filters bypassed, SC-68 functions as a unity-gain device. Use this mode to compare noched and flat sound.

Setup mode. Filters active, limiter active, all setup controls active.

Setup mode, filters bypassed. Setup controls still active. Use this mode for A/B comparisons of system gain with and without EQ during setup procedure.



METER. Gives a direct reading, in dB, of the relative improvement in system headroom as you notch out feedback frequencies. For ease of reading, the meter should be "zereoed" at the beginning of the setup procedure. See application notes which follow.

SPEED control. Adjusts the attack and release characteristics of the limiter when the SC-68 is used in the SETUP mode. Different feedback frequencies, in different size halls, take varying amounts of time to establish themselves. This control prevents the limiter from "hunting" and false tracking.

Figure 7 SC-68 Setup System Controls

USING THE SC-68 TO CORRECT FEEDBACK PROBLEMS

Most sound system operators are familiar with a method of "tuning" a room that involves purposely causing feedback to occur by turning up system gain, and then equalizing at the feedback frequency to stop the feedback. Usually, after the feedback has been stopped, the operator will again increase system gain until another feedback frequency occurs. That frequency is EQ'd, and the whold process is repeated until it becomes impossible to identify individual, predominant trouble spots. The problem with this approach is that it's very easy for the feedback to get out of control, which can be annoying at best or destructive at worst.

The setup system in the SC-68 essentially uses the same procedure described above to identify and remove feedback problems; as usual, you will turn up your mixer's volume level to purposely induce feedback and then tune to remove it. The difference is that the SC-68 has a built in limiter which will keep the feedback from running away, and maintain it at a listenable level until you have identified the feedback frequency. As soon as you have notched out that frequency, the SC-68 will automatically increase its gain and find the next trouble spot for you. In this way, the equalization process can take place at safe listening levels and a relaxed pace. As an added feature, the meter on the SC-68 will let you see exactly how much you've improved system headroom, leaving no doubt about the effectiveness of the EQ you've applied. In order for the meter reading to be accurate, you'll be asked to "zero" the meter at the beginning of the EQ procedure.

The step-by-step procedure for room equalization with the SC-68 is given below.

SET-UP PROCEDURE

1) "Normal" the SC-68: all 8 filters switched OUT, Mode switch set to "OPERATE/FLAT". Patch the SC-68 into your system between the mixer and your electronic crossover, or between the mixer and power amp if you're not using an active crossover.

2) Set up your system for normal operation, just as if you weren't using the SC-68; ie., play some music through the system and verify that everything is working and sounds reasonable. Turn up whatever mic inputs you normally use, and adjust each mic's sensitivity and EQ as usual. In this way, any room EQ you do will be based on normal system operation, and shouldn't change radically once the show starts.

3) Switch the SC-68 to the "SETUP/NOTCH" mode, adjust the SETUP LEVEL control to 5 and the SPEED control midway. The meter will be pegged to the right at this point.

4) Increase your mixer's master gain control until feedback is heard, and then continue to increase it another 10 dB or so. Adjust the SETUP LEVEL control to obtain a feedback level which is above room noise but not louder than necessary. Finally, re-adjust the mixer's master gain to obtain a 0 dB reading on the setup meter. Turning the gain up will move the needle to the left, and turning the gain down moves the needle to the right. After the meter has been zeroed, do not adjust SETUP LEVEL or the mixer's gain, since they will throw off the meter calibration.

5) Estimate the feedback frequency and punch in an appropriate filter band. Set the notch depth to 6 dB, bandwidth to .1 octaves, and slowly tune the frequency control until you find the first feedback point. When you think you've nulled it, try narrowing the bandwidth as much as possible and adjust the cut so that it is just deep enough to cause the feedback to change to another frequency. As a new frequency establishes itself, the meter reading should increase slightly. Notch out the new frequency, and keep repeating the process until no further improvement can be obtained. At this point, the meter will show the improvement in gain-before-feedback in decibels. If two frequencies occur close together, widen the bandwidth of one filter to suppress both frequencies with one notch.

It is possible that you'll have to increase the depth of some notches when frequencies nulled early in the tuning process re-establish themselves. It's best to go around the procedure a few times until you are convinced that all notches are producing their desired effect without being any deeper than necessary. At any time in the tuning process, you can switch between SETUP/NOTCH and SETUP/FLAT to re-establish your reference. Also, switching each filter band in and out will verify the effect of each notch.

It may be necessary to adjust the SPEED control of the setup limiter to obtain stable readings. The response time of the feedback modes in some rooms is slow and it is easily possible to tune right by a feedback point and miss it if you go too fast.

6) When you're finished tuning, reduce the master gain control of your mixer to a normal operating range, and punch the "OPERATE/NOTCH" switch. The system is now ready for use. Switch between OPERATE/NOTCH and OPERATE/FLAT to compare gain and sound quality, but remember that the limiter is now disabled; don't let the feedback "run away".

ADDITIONAL NOTES AND HELPFUL HINTS:

System headroom may be improved even further by re-positioning mics and speakers. The SC-68's meter will indicate any improvement. Don't attempt to actually operate the system in the SETUP mode, as it will sound terrible and operate at a very low level. The maximum gain before feedback of your system should be determined if the steps on page 9 are followed. If you are still not satisfied, here are some suggestions that might help:

Place all main system speakers and monitors such that they do not "see" any microphones that are to be fed to them.

Keep all guitar and keyboard stage amps away from vocal microphones.

Keep your stage volume as low as possible so your sound person has something to work with out front.

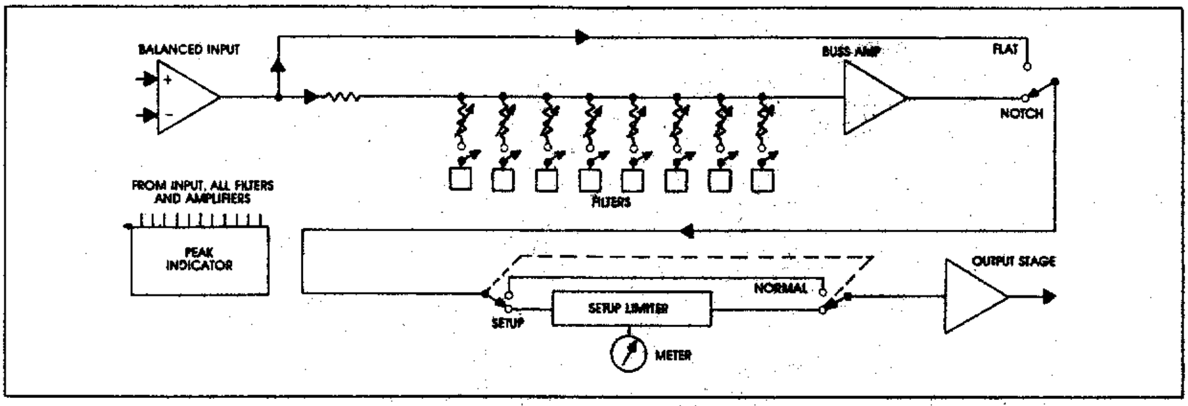
Work vocal microphones as close as possible.

When you can't hear yourself, have everyone else turn down rather than you turning up. You have already determined the maximum volume of your PA system, and if your fellow band members feel they can't turn down any more, collect donations to pay for upgrading your present system, ie; separate mixes for monitor speakers, more directional microphones, better speakers, etc.

SPECIFICATIONS

AMPLITUDE	0 to -30dB
FREQUENCY	(low) 16Hz-800Hz, (3/low-mid) 48Hz-2.4kHz, (3/hi-mid) 160Hz-8kHz, (high) 480Hz-24kHz
BANDWIDTH	.3 octave - .03 octave
INPUT IMPEDANCE	10k Ω active balanced bridging
MAX. IN-OUT LEVEL	+20dB (+5dB at max. cut, full sharp)
FREQUENCY RESPONSE	±.5dB 20Hz-20kHz
OUTPUT IMPEDANCE	50 Ω , term with 600 Ω or more
HUM AND NOISE	-87dBV (eq in), -95dBV (eq out)
DISTORTION	< .05% THD, +10dBV, 20Hz-20kHz
POWER	120 VAC, 50-60Hz, 5W.
SHIPPING WEIGHT	12 lbs.

BLOCK DIAGRAM



TROUBLE SHOOTING TIPS

NO OUTPUT

Check AC power - is the pilot light on? Check in/out connections, are they reversed? Are you sure you have an input signal?

PEAK LIGHT FLASHES OR STAYS ON ALL THE TIME

If the peak light flashes, the signal level to the equalizer is too high. Turn down the gain. If it is on all the time, disconnect the input and output cables. If it is still on, the unit must be returned for service.

DISTORTED SOUND

This will only be caused by too much signal (which will show on the "peak" light. If the light is not flashing, there is an overload somewhere else in the chain. Adjust the relative gain of each component in your chain to keep everything at a comfortable level. Also, make sure you are not trying to operate the SC-68 in the SETUP mode.

EXCESSIVE HUM OR NOISE

Hum will usually be caused by a "ground loop" between components. Try using the suggested balanced input and output hook-ups if the other pieces of equipment used in conjunction with your equalizer have balanced inputs and outputs. Noise can be caused by insufficient drive signal. Make sure you are sending a nominal 0 dBV line level signal to the equalizer.

NOTE:

UN-SHIELDED CABLES, IMPROPERLY WIRED CONNECTIONS, AND CABLE WITH BROKEN STRANDS (SHORTS ETC.) ARE THE MOST COMMON PROBLEMS. MAKE SURE YOU USE GOOD QUALITY CABLE.

WHEN IN DOUBT, GET IN TOUCH WITH YOUR ASHLY DEALER, OR CALL THE FACTORY DIRECT - (800)828-6308. In New York State dial (716)544-5191.

OHM
The unit of electrical resistance or impedance.

PHASE
Describes how well two signals are in step. In-phase means that positive and negative peaks in two signals occur together, while out-of-phase means they do not occur together. Variations in signal timing as well as polarity can make two signals in or out of phase, or anywhere in between. Phase is usually measured in degrees where 0 degrees is in-phase, 180 degrees is out-of-phase, and 90 degrees is in between (sometimes called quadrature).

PREAMPLIFIER
The first stage of amplification, designed to boost very low level signals to line level.

"Q"
A measurement describing the sharpness or broadness of a filter.

SHELVING
Describes an equalization action where all frequencies above or below a particular frequency are boost or cut.

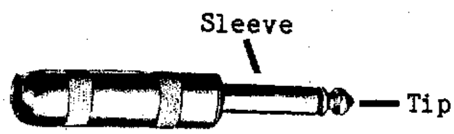
TRANSIENT
A sudden burst of energy in an audio signal, such as a breath blast in a microphone, the sound of a snare drum, or a deep scratch in a record. Transients frequently reach peak levels of 10 to 30 dB above standard operating level, and may cause distortion or even damage to equipment.

UNITY GAIN
Output level = Input level.

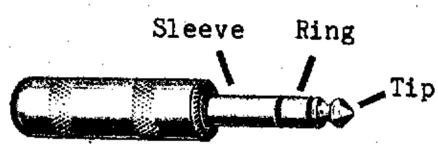
WIRING, PHONE PLUG AND XLR
A stereo phone plug is wired + to the tip, - to the ring, and shield to the sleeve. For a mono phone plug, combine - and shield, and connect both to the sleeve.

An XLR (3 Pin) connector is wired + to pin 3, - to pin 2, and shield to pin 1.

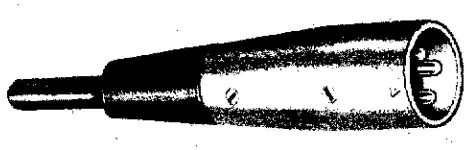
Mono Phone Plug:
(for unbalanced inputs and outputs)



Stereo Phone Plug:
(for balanced inputs and outputs)



XLR Type Connector:
(Male Shown)



DISTORTION

Generally refers to ANY modification of an audio signal which produces new frequencies which were not in the original. Examples are harmonic distortion, where a circuit adds overtones to a fundamental signal, and intermodulation or IM distortion, where two frequencies beat together to produce sum and difference frequencies.

FEEDBACK

Generally refers to any process where an output is in some form routed back to an input to establish a loop. Negative feedback tends to be be self stabilizing, while positive feedback causes instability.

FILTER

A circuit designed to pass some frequencies, but not others. There are three general categories of filters: High-pass, band-pass, and low-pass. The high-pass filter passes frequencies above a certain limit, the low-pass passes frequencies below a limit, and the band-pass passes one group of frequencies without passing those above or below. Our equalizer uses band-pass filters, crossovers use high and low-pass filters.

FREQUENCY

The repetition rate of a waveform. Frequency is measured in Hertz. One cycle per second (cps) is one Hertz (Hz). The higher a note on a musical scale, the higher its frequency.

FREQUENCY RESPONSE

Refers to relative gain and loss at various frequencies across the audio band. May be illustrated by a graph called a frequency response plot, usually graphing decibels vs. Hertz or octaves.

HERTZ (Hz)

The unit of frequency measurement. (Formerly called Cycles-per-Second: this explains it perfectly)

HEADROOM

Refers to the increase in level above normal operating level that can be obtained without clipping. Usually expressed in dB.

IMPEDANCE

Essentially the AC equivalent of resistance. It describes the drive capability of an output, or the amount of drive required for an input at any given signal level.

KHz

Kilohertz. 1,000 Hertz.

LEVEL

The magnitude of a signal, expressed in decibels or volts.

LINE LEVEL

Meaning "somewhere around 0dBV" as opposed to MIC level of around -40dBV.

OCTAVE

A logarithmic unit to compare frequencies. +1 Octave means double frequency, -1 Octave means half frequency.

DEFINITION OF TERMS AS USED IN THIS MANUAL

ACTIVE

Electronic circuits which use devices such as transistors and integrated circuits, and which are capable of voltage and power gain as well as loss. Circuits using only resistors, capacitors, transformers, etc., are referred to as passive.

AMPLITUDE

The voltage level of a signal. May be measured in volts or decibels. Generally corresponds to the volume or intensity of an audio signal.

BALANCED

A 3-wire circuit arrangement in which two conductors are designated as signal lines (+ and -), and the third is a shield and chassis ground. The signal lines are of opposite polarity at any given moment, and are of equal potential with respect to ground. Balanced input amplifiers are used on all Ashly SC series products to improve hum and noise rejection. Jumpering signal minus (-) to ground provides an unbalanced input.

CENTER FREQUENCY

The frequency (or pitch) at which a filter is most effective. In a parametric equalizer, it refers to the frequency where a particular boost/cut control has maximum effect.

dB

A unit by which audio levels can be COMPARED. Often thoroughly misunderstood are the concepts that decibels represent the level of a signal compared to some reference level (15 dB cut means a certain level less than a previous level --- the absolute level of the signal need not be known), and that decibels are a logarithmic unit.

- Some handy numbers to remember when dealing with decibels:

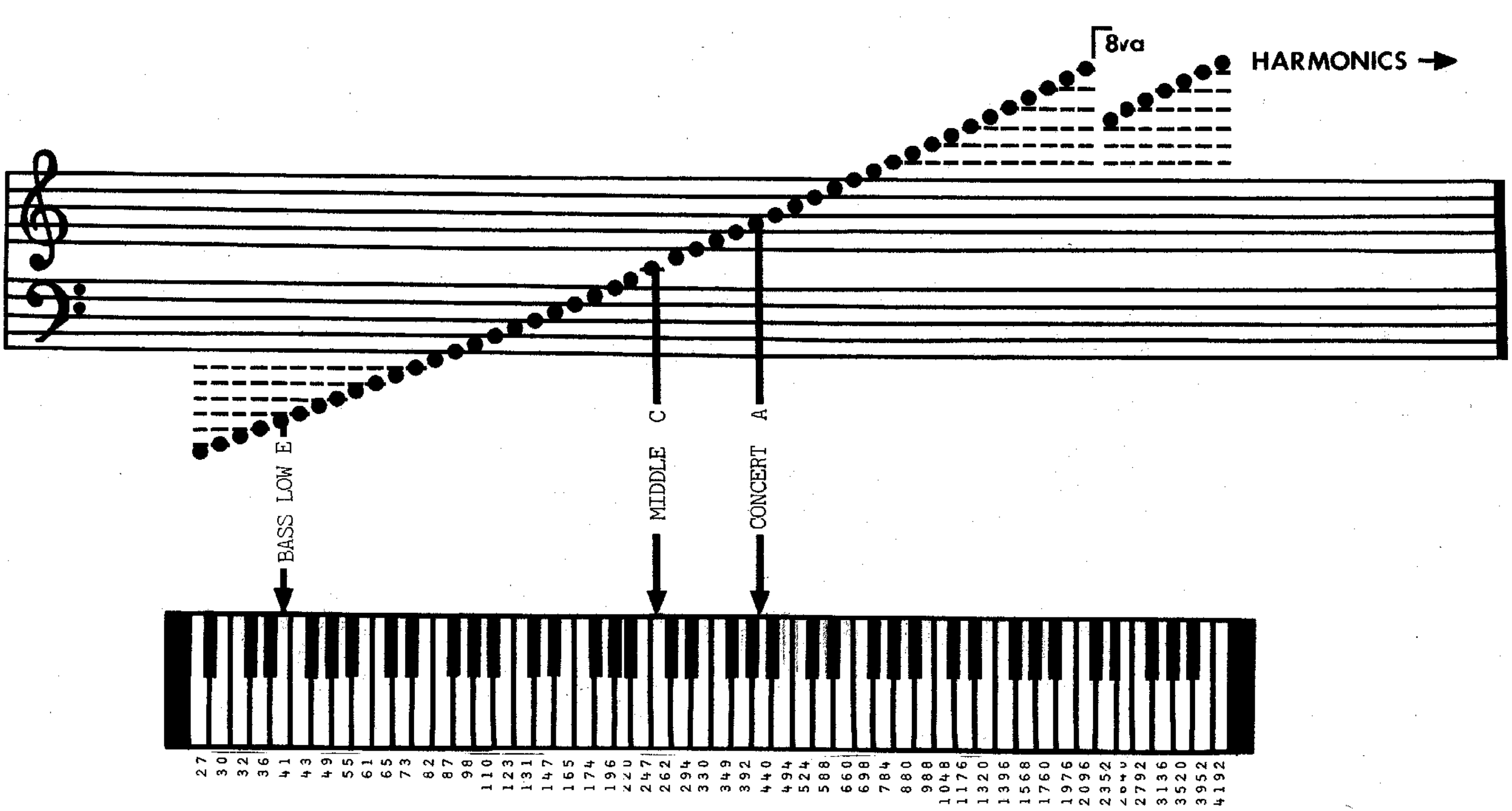
  +3 dB = Double Power
  +6 dB = Double Amplitude, Quadruple Power
  +10 dB = 10X Power
  +20 dB = 10X Amplitude, 100X Power

dBm

A unit of measurement in decibels where 0 dBm = a power level of 1 milliwatt into a 600 ohm load. Originally defined by the telephone company to measure line levels.

dBV

Decibel Volts, an update of the dBm definition where 0 dBV = the same voltage level as 0 dBm, but with no regard to power or impedance. 0 dBV = 0.778 Volts. This unit is much more appropriate for modern audio equipment with high impedance inputs and low impedance outputs.



Actual frequency numbers (Hz) may vary from the above chart.