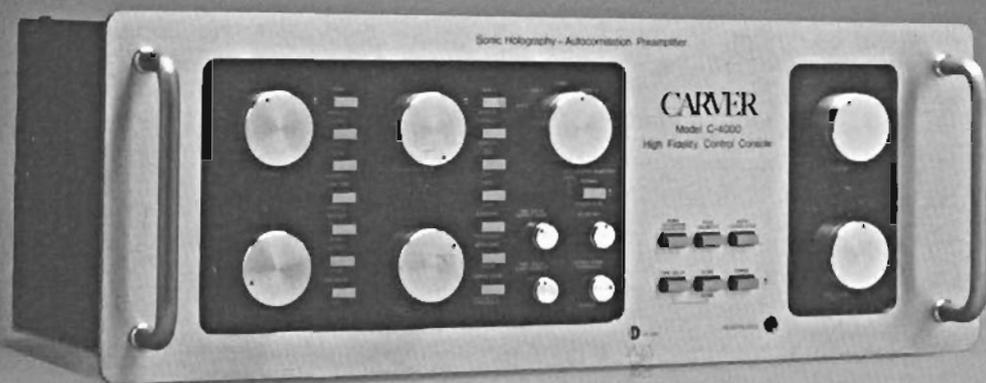


CARVER

MODEL C4000



High Fidelity Control Console

- Wide-range Full-function Preamplifier
- Sonic Hologram Generator
- Autocorrelator Noise Reduction System
- Peak Unlimiter Dynamic Range Expander
- Time Delay

CARVER CORPORATION

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i. INTRODUCTION

Congratulations on your purchase of the Carver C-4000 Sonic Holography Control Console. We believe its sophisticated engineering and meticulous craftsmanship will provide you with many years of listening enjoyment.

At its heart the C-4000 is a high-performance console preamplifier with precise RIAA phono equalization, low distortion, high slew rate, wide dynamic range, and a high degree of control flexibility. But its similarity to other fine preamps ends there. The C-4000 goes far beyond mere accuracy and fidelity: its goal is a dramatically higher degree of realism than has ever been achieved before in audio reproduction. Thus its Peak Unlimiter undoes the dynamic compression which makes many recordings and broadcasts sound "canned". Its Auto-Correlator noise-reduction system strips away the veil of low-level rumble and hiss which form a perpetual psycho-acoustic barrier to realism in the reproduction of musical timbres. And its Time-Delay and revolutionary Sonic Hologram Generator break through the limits of listening-room acoustics and the two-channel stereo medium itself to provide a remarkably three-dimensional you-are-there realism, with a sonic image which subjectively extends far beyond the loudspeakers in both breadth and depth.

ii. NOTICE TO PURCHASER

IT IS O.K. TO BEGIN USING YOUR NEW C-4000 WITHOUT FIRST READING THIS OWNER'S MANUAL. YOU CANNOT HURT IT, AND IT CANNOT HURT YOU. If you have previous experience with stereo components, you probably can complete the input/output connections and make normal use of the basic preamplifier functions without special instructions.

HOWEVER, IT IS NOT POSSIBLE TO MAKE THE SONIC HOLOGRAM GENERATOR WORK PROPERLY WITHOUT READING THIS MANUAL. Because of the special nature of sonic holography, a period of fine-tuning of your stereo system will almost certainly be necessary before its full benefit can be obtained, and this Manual provides essential guidance in that process. For speedy reference, all of the information on sonic holography has been consolidated in one section of the Manual. The same is true for each of the C-4000's other

special signal-processing features: the Peak Unlimiter, the Autocorrelator, and the Time Delay. Subsidiary information is set apart from the main text, either in indented small-type paragraphs or in appendices at the back of the Manual.

This Manual was prepared with unusual thoroughness and care, and we recommend that you read it in its entirety at your earliest convenience. You will find that even some of the "conventional" parts of the C-4000 are in fact not conventional at all.

We also recommend that you record the following information here, for possible future reference:

C-4000 Serial No.: _____

Purchase Date: _____

Store Name and Address: _____

iii. UNPACKING

The C-4000 is not fragile, but it is a precision instrument and deserves to be handled with care. As you unwrap it, we recommend that you return all packing materials to the carton and then store the carton in a convenient place. It has been designed to provide the safest protection for the C-4000 and should be used whenever you ship the unit, e.g. when moving to a new address or returning the unit for servicing.

iv. CAUTION NOTICE

To prevent a fire or shock hazard, do not expose this unit to moisture or rain. If it accidentally becomes wet, disconnect its AC power cord until the unit is thoroughly dry, inside and out.

Before connecting or disconnecting cables, unplug the C-4000's AC power cord or switch off the power to each component in the stereo system – especially the power amplifier. If the power amplifier lacks an on-off switch, unplug its own power cord.

Do not remove the top or bottom cover of the C-4000. There are no user-serviceable parts inside. Refer all servicing to qualified personnel; unauthorized servicing may void the warranty.

1.0 INSTALLATION

The C-4000 is an all-solid-state design and may be operated horizontally, vertically, or at any angle. It has no special ventilation requirements, but it should not be operated in a totally enclosed cabinet since its time-delay amplifiers produce a small quantity of heat (causing the rear panel to become moderately warm to the touch.)

Its location relative to other stereo system components is generally not critical, with one exception: the power transformers of some power amplifiers radiate a severe external hum field, so it may not be practical to place the preamp adjacent to or stacked with the power amp. (Such power amplifiers should also be kept away from turntables, phono signal cables, and the playback heads of tape recorders.)

Typically the preamp must be located close to the turntable in order to keep phono signal cables short; begin by locating the turntable on a stable, vibration-free surface and then arrange the preamp and other system components in convenient locations nearby.

On the other hand, in many systems it would be a worthwhile convenience if the preamp could be located within arm's length of the prime listening seat, while keeping the turntable on a stable vibration-free shelf. Often this can be done simply by using extra-long connecting cables. Such cables increase the risk of hum and radio interference; if neither problem arises, then the remaining concern is the capacitance of the cable as it relates to the "load" requirements of your phono cartridge. If its manufacturer specifies that the cartridge should be used with a relatively low value of load capacitance (e.g. 250 picofarads or less,) then you can't use a long cable unless it is of special low-capacitance design. With cartridges designed for a load capacitance of 400-500 pF, or with low-inductance cartridges, which are insensitive to cable capacitance, you may find that you can use long connecting cables with no adverse effect on the sound. For reference, the lowest-capacitance audio cables typically add about 15 pF per foot of cable length; garden-variety audio signal cables may add as high as 50 or 100 pF per foot of length.

Although all C-4000 instruments are 19 inches wide and have handles, only the charcoal-finished units are intended to be rack-mounted. The decorative handles should not be removed, as the screws which hold them in place are also being used to fasten the front panel to the chassis. If special circumstances require the removal of the handles, bolts and nuts must be used in the mounting holes to fasten the front panel.

2.0 USING YOUR C-4000: A QUICK OVERVIEW

Doubtless you are eager to begin using the C-4000. Therefore this chapter provides a very condensed summary of the essential information that you need to hook up and use the preamp, without waiting to read the entire owner's manual.

NOTE: The C-4000 is different from most pre-amplifiers, not only in the obvious ways (i.e. inclusion of the Sonic Hologram Generator and other special Carver circuits) but in many not-so-obvious ways as well. In order to get full benefit from the original design of the C-4000, you owe it to yourself to read the rest of this manual at your earliest convenience.

Now, to get started. The diagram on the facing page illustrates in typical fashion the connection of stereo system components to the C-4000. Before making connections it is wise to switch off each component's power switch and unplug its AC power cord from the socket. Connect all of the signal cables, then plug in the AC line cords, and finally turn the system components on.

If you have any difficulty deciding how to connect a particular system component, refer to the more detailed discussions in Chapter 3.

The illustration on the following page identifies the many control knobs and pushbuttons on the front panel of the C-4000, with a succinct description of the use of each control. To begin, all of the pushbuttons should be OUT, except the following:

The **POWER** button should be IN in order to turn the preamplifier on.

The **INFRASONIC FILTER** button should be IN.

The **TONE** button should be IN if you want to use the Bass and Treble controls.

To hear the playback from a tape deck, depress the appropriate **TAPE MONITOR** button. Otherwise both **MONITOR** buttons should be OUT.

Initially all of the control knobs (Bass, Treble, Balance, etc.) should be set to the 12 o'clock position, except as follows:

The **VOLUME** knob should be turned down (counter-clockwise) when the **POWER** is first switched on, and then can be adjusted for a pleasing loudness level.

The **TIME DELAY OUTPUT LEVEL** and **ECHO DENSITY** controls should be turned down.

(Continued on page 5)

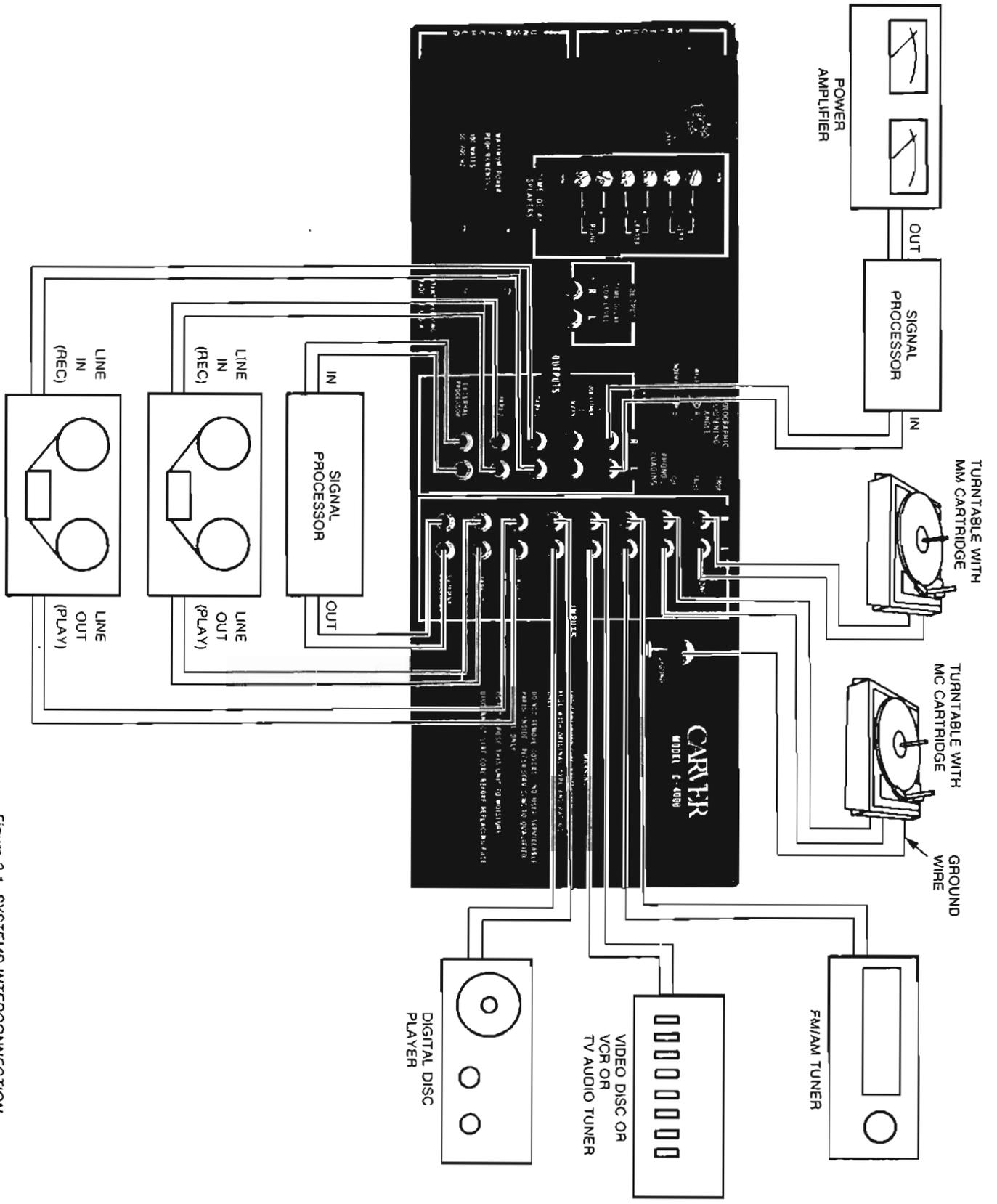


Figure 2-1. SYSTEMS INTERCONNECTION

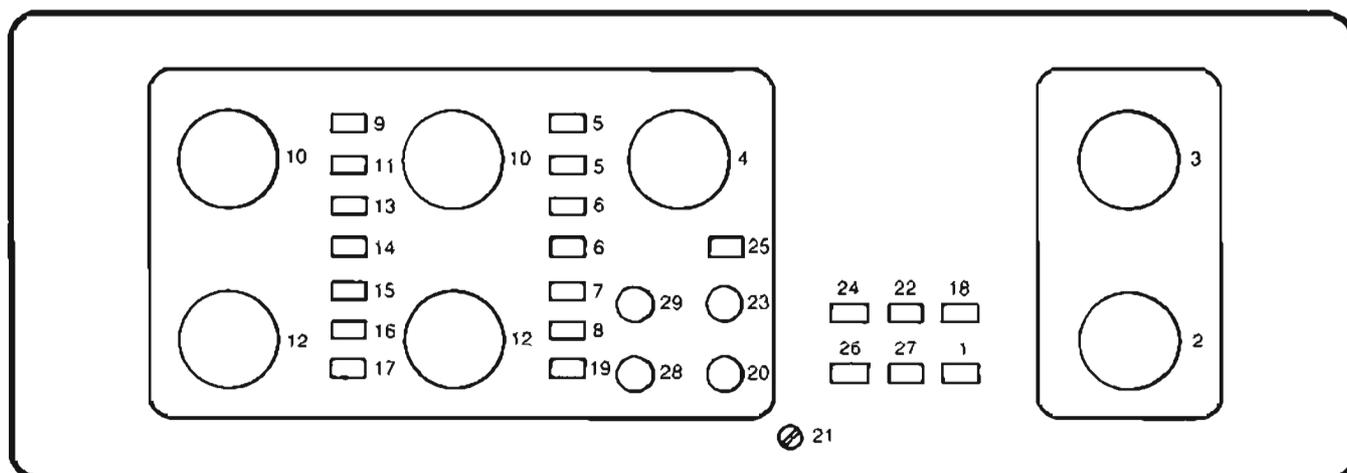


Figure 2-2. FRONT-PANEL CONTROLS

1. **POWER.** Turns on power to the C-4000 and to the Switched AC convenience outlets on the rear panel.
2. **BALANCE.** Adjusts the relative levels of the two stereo channels.
3. **VOLUME.** Adjusts the output levels from the preamplifier.
4. **INPUT SELECTOR.** Selects one of five program sources.
5. **TAPE MONITOR (1 and 2).** Selects the playback output from either tape deck for listening. Allows monitoring of recordings as they are being made.
6. **DUB (1 → 2 and 2 → 1).** Allows copying of recordings from Tape 1 to Tape 2, or vice-versa. When both DUB buttons are out, the signal chosen by the INPUT SELECTOR is fed to the tape decks.
7. **EXTERNAL PROCESSOR.** Engages an outboard signal processor such as an equalizer. The selected processing affects signals being heard, including playback from tapes, but has no effect on recordings being made.
8. **INFRASONIC FILTER.** When engaged, eliminates unwanted signal energy below 20 Hz.
9. **TONE — PUSH ON.** Engages the Bass and Treble tone controls. When out, tone control circuit is bypassed.
10. **LEFT TREBLE, RIGHT TREBLE.** Provides a sloping boost and shelving-type cut for high frequencies.
11. **8 kHz — 2 kHz.** Alters the turnover frequency of the Treble controls; broadens the Treble control range when depressed.
12. **LEFT BASS, RIGHT BASS.** Provides a sloping boost and a shelving-type cut for low frequencies.
13. **40 Hz — LOUDNESS.** Alters the turnover frequency of the Bass controls; broadens the Bass control range when depressed.
14. **-3 dB TRIM.** Alters the overall bass/treble balance of the sound; reduces the bass and lower-midrange energy by 3 dB when depressed.
15. **STEREO — MONO.** Combines the two channels when engaged.
16. **OUTPUT — 15 dB.** Reduces the overall output level by 15 dB.
17. **SPEAKERS OFF.** Mutes the main output signals from the preamplifier, without affecting the Headphone or Time Delay outputs.
18. **AUTOCORRELATOR.** Engages Autocorrelator noise-reduction circuit.
19. **CORRELATION — AUTOMATIC THRESHOLD.** When engaged, autocorrelator self-adjusts to noise level in program. Adjacent LED illuminates in Automatic mode. When button is out, the threshold is set by the CORRELATION THRESHOLD — MANUAL control.
20. **CORRELATION THRESHOLD — MANUAL.** Allows user to set the operating threshold of the Autocorrelator noise-reduction circuit. Adjacent LED illuminates in the Manual mode.
21. **LF CAL.** Screwdriver adjustment to set the threshold of the Low Frequency (hum and rumble) filter in the Autocorrelator system.
22. **PEAK UNLIMITER.** Engages the dynamic range expander.
23. **PK UNLIMIT.** Sets the threshold of the dynamic range expander. Adjust until adjacent LED flashes during peaks.
24. **SONIC HOLOGRAM GENERATOR.** Switches the holographic imaging circuit into the signal path.
25. **HOLOGRAPHIC INJECTION RATIO (NORMAL/THEORETICAL).** Alters the holographic image. The THEORETICAL setting produces a more dramatic wraparound, while the NORMAL setting (button out) is preferred with well-miked recordings.
26. **TIME DELAY.** Activates the time-delay circuits to reproduce ambience via side/rear speakers.
27. **16 MS — 25 MS.** Adjusts the length of the time-delay to vary the spatial quality of the ambience. (Labeled 35/50 MS in pre-1982 production.)
28. **TIME DELAY ECHO DENSITY.** Adjusts the strength of the feedback signal which is recycled through the delay circuit to synthesize reverberation. Its range is from 0 (no reverb) to beyond 100% (producing a sustained oscillation).
29. **TIME DELAY OUTPUT LEVEL.** Adjusts the relative volume level of the delayed ambience signal, i.e. the ratio of ambient to direct sound.

To use the **PEAK UNLIMITER**: depress the **PEAK UNLIMITER** pushbutton to activate the circuit, and then adjust the **PK UNLIMIT** knob until the adjacent **LED** just flashes during the climaxes and high-level transient peaks in the music.

To use the **AUTO-CORRELATOR** noise reduction system: depress the **AUTO-CORRELATOR** pushbutton to activate the circuit, and depress the **CORRELATION — AUTOMATIC THRESHOLD** to engage the automatic setting of threshold sensitivity.

To use the **TIME DELAY**: refer to Chapter 8.

To use the **SONIC HOLOGRAM** generator: follow this step-by-step procedure.

1. Pull your loudspeakers at least two feet away from the wall behind them and at least two to three feet away from the walls on either side of the speakers.

2. If you normally have a very wide spacing between your speakers, move them closer together. They should be not more than a few feet apart.

3. Move the speakers away from any large object or surface that could cause strong sound reflections, or cover that surface with sound-absorbing material.

4. Place your chair approximately on the center-line equidistant from the two speakers, five to ten feet away.

5. Obtain a tape measure or a piece of string. Use it to measure the distance from the middle of your chair to the left-channel speaker. Measure the distance from the middle of your chair to the right-channel loudspeaker.

6. Move either speaker forward or backward as necessary until the two speakers are exactly equidistant from your chair.

7. "Toe in" the speakers to aim them at your chair. Double-check with the string or tape measure to be sure that the two speakers are equally distant from your chair. **DO NOT RELY ON A VISUAL ESTIMATE OF THE CENTER-LINE OF YOUR SYSTEM TO PLACE YOUR CHAIR ON THE STEREO AXIS.**

8. Set the Listening Angle switch (on the rear panel) to **WIDE**.

9. Depress the **SONIC HOLOGRAM** pushbutton to turn the circuit on.

10. Depress the **Injection Ratio (NORMAL/THEORETICAL)** pushbutton, to select the **THEORETICAL** position which yields the most pronounced holographic effect. After you have learned what sonic holography sounds like, you may select either setting of the button, as seems most appropriate for the particular record being played.

11. Enjoy sonic holography.

3.0 SYSTEM WIRING: REAR PANEL CONNECTIONS

3.1 PHONO 1 and PHONO 2 INPUTS

The C-4000 contains two completely independent stereo pairs of phono preamplifier circuits. Connect the signal cables from your turntable to the input which corresponds to the type of pickup cartridge you are using.

Phono 1. This is the standard RIAA input for conventional magnetic cartridges (moving-magnet, induced-magnet, variable-reluctance, moving-flux). It should also be used for "high-output" moving-coil cartridges having a rated output of at least 1mV (e.g. Dynavector 10X, Adcom XC, Satin M117, et al). The Phono 1 input impedance is 47,000 ohms resistance in parallel with a user-selectable capacitance (see below).

Phono 2. This input is for low-output moving-coil pickups. It provides 25 dB more gain than Phono 1, with an input impedance of 39 ohms. (Connect the signal cables from the turntable directly to the Phono 2 jacks; no step-up transformer or pre-preamplifier is needed. With its high gain and extremely low noise, the Phono 2 preamp circuit combines the advantages of pre-preamps and transformers without the faults of either.)

Refer to the instructions supplied with your turntable or tone arm to discover which signal lead carries the Right channel signal and which the Left. When plugging the cables into the preamp's sockets, be sure that each plug is inserted fully into the socket, making a tight fit.

Note that each plug consists of a protruding metal finger which should be a good friction fit as it goes into the socket, plus a metal skirt which should fit tightly on the metal exterior of the socket. If the fit is too loose or too tight, use small ("needle nose") pliers to bend the individual leaves of the skirt slightly in or out. It is also a good practice to rotate or twist the plug on its axis as it goes into the socket; thus any corrosion or oxidation on the plug and socket surfaces will be scraped away, making a good distortion-free metal-to-metal contact.

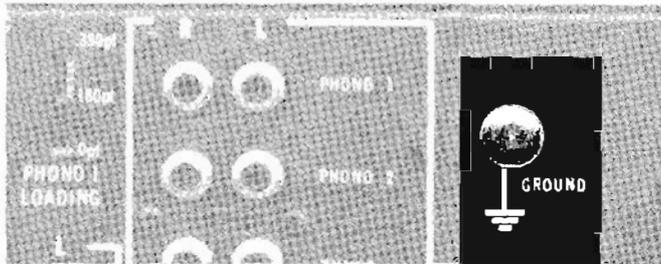
Selecting Input Capacitance. (Phono 1 only). Some phono cartridges are relatively insensitive to the capacitance of connecting cables and the preamp input (these will usually be identified as low-impedance or low-inductance designs). But the majority of high-inductance moving-magnet and induced-magnet pickups will exhibit flattest frequency response only within a specific optimum range of load capacitance values.

In order to select the best value of preamp input capacitance, you first must determine the total capacitance recommended for the cartridge. (See the manufacturer's specifications or refer to magazine reviews.) Then subtract the capacitance which is contributed by the phono signal cables and tonearm wiring of your turntable. (Again, check the maker's specifications, test reports, or as a last resort assume a typical value of 150 pF.)

What remains is the value of input capacitance which should be added by the preamp input; set the PHONO 1 LOADING switch to the nearest value. (There is no need to match the computed value of capacitance exactly: with most cartridges a variation of 50 or 100 pF from the ideal value produces only a very slight change in response.)

Example: the Stanton 881S cartridge is designed for a 275 pF load, and a Pioneer turntable has a cable capacitance of 100 pF; 275 minus 100 equals 175 pF, so set the PHONO 1 LOADING switch to 180 pF.

An alternative approach is to set the capacitance by listening to recordings. Typically, with too low a capacitance the upper-midrange (the soprano voice range) will be "soft" while the response at the highest frequencies will be peaky, leading to increased surface noise and edgy violin tone. Too much capacitance will bring the upper-midrange forward and roll off the extreme highs.



3.2 GROUND

If your turntable is equipped with a separate ground wire, try connecting it to the preamplifier's GROUND post and note whether this connection minimizes hum. If it does, make the connection permanent. If you have two turntables, try connecting both ground wires to the GROUND post.

Normally these are the only connections made to the GROUND post. All other components in the stereo system are grounded through their signal cables, and if you connect additional grounding wires among stereo components the result will be the formation of ground-current loops and an increased hum level.

In some systems it may prove beneficial to connect a wire from the preamplifier's GROUND post to a true electrical earth ground such as a cold-water pipe; see the separate chapter on Hum.

3.2 TUNER INPUTS

Connect the signal cables from an FM or FM/AM tuner to these jacks. If your tuner has both "fixed" and "variable" output jacks, it usually is preferable to use the "variable" jacks. Then you can adjust the tuner's Output Level control so that, when you switch from Phono to Tuner, the volume level of the sound is approximately the same.

3.4 AUX 1 and AUX 2 INPUTS

Connect any other "line-level" signal sources to these jacks. Examples include a TV-sound tuner, a separate AM radio tuner, the output of a spare tape machine used only for playback but not recording, the output of a separate microphone preamp or mixer, etc. Note: if you want to connect the sound signal directly from an operating television set, first check with a TV service technician to be sure that the set's circuitry is isolated from the AC power line via a power transformer. In general, console and projection TVs are safe in this respect, but some portables are not.

3.5 TAPE 1 and TAPE 2 INPUTS

You can connect cables from the line-level playback outputs of two tape recorders to these jacks. The output jacks may be identified as LINE OUT, PLAY, or MONITOR. If you are using a separate tape noise-reduction system (Dolby, DBX, etc.) with a recorder, then the recorder's outputs will be connected to the processor; in this case it is the processor's playback outputs which should be connected to the C-4000's Tape Inputs.

If you are connecting two tape recorders it normally makes no difference which is connected to TAPE 1 and which to TAPE 2.

If your tape recorder is equipped with a DIN five-pin input/output socket as well as conventional phono jacks, make connections to the phono jacks; usually these will provide the best match to the C-4000 in terms of impedance, sensitivity, and minimum noise. If the recorder is equipped only with a DIN socket, then you can obtain an adapter cable with a DIN plug on one end and phono plugs at the other for connection to the C-4000.

3.6 EXTERNAL PROCESSOR INPUTS

These jacks enable the connection of any of a large range of signal processing accessory devices.

Examples include:

- A graphic equalizer or parametric equalizer.
- A dynamic noise filter, impulse noise suppressor ("tick-and-pop filter"), or scratch filter.
- A dynamic-range expander, compressor, or limiter.
- The special equalizer unit supplied with some loudspeakers, notably certain models from Bose, ElectroVoice, KLH, and Infinity. Alternatively, such a

speaker equalizer may be connected between the C-4000's Main Out jacks and the power amplifier's input jacks, leaving the External Processor loop free for other accessories. If the equalizer lacks a Flat or Bypass setting, and if you want to use the power amplifier to drive electrostatic headphones or other loudspeakers not requiring equalization, then the equalizer should be connected to the External Processor jacks where it can be engaged or bypassed at will.

- Any other special-purpose filter or equalizer, e.g. the Cotter audio-bandpass filter or the Allison Electronic Subwoofer.
- The "front" outputs from an external "spatial ambience" matrix decoder or time-delay unit. Usually, however, these signals will be connected directly to the power amplifier inputs so that the preamp's controls can serve as "master" controls governing the sound of all speakers in the system.

In general, devices whose operation depends on the setting of a threshold or sensitivity control (such as an impulse-noise suppressor) should be connected to the External Processor jacks because the signal levels there will not be affected by the settings of the preamp's Volume and Tone controls. But devices such as equalizers, which are not sensitive to signal level, may be connected either in the External Processor loop or at the preamp's Main Output; the choice is a matter of convenience.

The External Processor loop is located after the TAPE input/output circuits in the C-4000's signal path. Therefore signal-processing accessories connected in the External Processor loop can be used to process the playback signal from a tape recorder, but cannot be used to modify or improve the signal being fed to a tape deck for recording.

If you want to pre-process a signal before recording it, there are several ways to arrange it.

- Disconnect the recorder from the normal TAPE input/output jacks and connect it instead to the signal processor's own Tape in/out jacks. (Most equalizers have these jacks.)
- Connect the signal processor to the preamp's TAPE 1 input/output jacks instead of the External Processor jacks. Then when the front-panel DUB 1 → 2 switch is engaged, the processed signal will be fed to the tape recorder connected to the TAPE 2 jacks.
- Disconnect the recorder from the normal TAPE input/output jacks and connect a cable from either of the preamp's sets of MAIN OUT jacks to the recorder's line input. Then the input signal to the recorder will be affected, not only by any signal processor in the External Processor loop, but also by all of the preamplifier's controls and circuits including its Volume, Tone, autocorrelator, peak limiter, and sonic holography functions.

You may connect a spare tape recorder to the External Processor jacks instead of an accessory device. Functionally, the External Processor input/output connections are similar to the two TAPE

connections, except that the "dubbing" function operates only between the two sets of TAPE jacks. So a recorder connected in the External Processor loop can be used for normal recording and playback, but tapes cannot be dubbed (copied) from this deck onto a recorder connected to TAPE 1 or TAPE 2.

There is one major limitation associated with the use of the External Processor Inputs. The Auto-Correlator and Peak Unlimiter circuits operate by sensing the audio signal level at a point in the signal path after the Tape inputs and ahead of the External Processor inputs. Therefore these circuits cannot be used to enhance signals from the External Processor inputs.

3.7 EXTERNAL PROCESSOR OUTPUTS

Connect a signal cable from these jacks to the line-level input jacks of the signal-processing accessory (or other device) whose playback cables have already been connected to the C-4000's External Processor Input jacks.

3.8 TAPE 1 and TAPE 2 OUTPUTS

Connect cables from these jacks to the line-level input jacks of your tape recorders. Be sure that the cable from TAPE 1 OUT goes to the same machine whose playback cable is connected to the C-4000's TAPE 1 INPUTS.

If one of your recorders is connected to an outboard noise-reduction unit (Dolby, DBX, etc.), connect the cable from the C-4000 to the unit's "from amplifier" input jacks rather than the recorder's own input jacks.

3.9 MAIN OUTPUTS

Two pairs of Main output jacks are provided for convenience. Use either pair, or both. Connect a cable from these jacks to the input jacks on your power amplifier. (The power amplifier, of course, should be turned off when making this connection.) The spare set of Main Output jacks can be used to feed signals to a second power amplifier driving additional speakers, to feed processed signals to a tape recorder, etc.

If you are bi-amplifying your speakers with the aid of an electronic crossover, connect the cable from Main Out to the crossover's input jacks; then connect cables from the crossover's High and Low range outputs to the appropriate power amplifiers.

If your speakers require the use of a special equalizer and you have not connected it to the External Processor input/output jacks, then connect the cable from the Main Outputs to the equalizer's inputs, and a second cable from the equalizer's outputs to your power amplifier's input jacks.

The output impedance of the C-4000 is only 910 ohms, so it can drive power amplifiers having virtually any input impedance. And if you are using powered loudspeakers (with built-in amplifiers), or if you choose to locate your

power amplifier close to your speakers in order to avoid any deleterious effects due to long speaker wires, you can safely use long connecting cables from the C-4000 to the power amplifier inputs. Conventional cables as long as 30 feet, or special low-capacitance cables as long as 60 feet, may be connected between the preamplifier and the power amplifier without difficulty.

TIME DELAY SPEAKERS, TIME DELAY LOW LEVEL OUTPUT

These connections are discussed in the chapter on Time Delay.

3.10 FUSE

This fuse protects the preamplifier in case of accident or internal malfunction. If it blows the preamplifier will not function. Replace it with a similar 1¼ ampere fast-blow fuse (e.g. Buss AGC 1¼ or Littlefuse 3AG 1.25 ampere) only. Do not substitute a larger value fuse. If the fuse blows repeatedly after replacement, return the preamp for service.

3.11 AC SOCKETS

The C-4000 is equipped with three "switched" AC convenience outlets whose power is switched on and off by the C-4000's own front-panel power switch. These can be used to supply AC power to all-electronic products such as tuners and power amplifiers. The switch is a heavy-duty unit and can safely be used to switch on power amplifiers of up to 1000 watts.

In addition, three "unswitched" AC convenience outlets are provided; these are always "live" as long as the C-4000's power cord is plugged into a wall socket. The unswitched outlets should be used to supply AC power to mechanical devices (turntables, tape recorders) which should be turned on and off with their own power switches. The unswitched outlets can also be used to supply power to devices which are intended to be left permanently on (e.g. to avoid turn-on transient "thumps").

3.12 AC LINE CORD

After completing connections to the power amplifier and other system components, plug the AC line cord into a convenient wall outlet supplying 110-120 volts AC.

The supplied line cord is a heavy-duty wire capable of carrying the large currents required by power amplifiers which may be plugged into the C-4000's convenience outlets.

The C-4000 itself requires only a modest amount of power, so it can be operated from ordinary AC extension cords if desired. But if you have a substantial power amplifier connected to one of the convenience outlets, a heavy-duty extension cord must be used, one made of wire no higher than 16-gauge, preferably 14-gauge.

4.0 THE FRONT PANEL: PREAMPLIFIER CONTROLS

The front panel of the C-4000 is divided visually into three sections or control groupings:

(1) The group of six switches, the LF CAL control, and the HEADPHONES socket, all located below the CARVER logo.

(2) The master VOLUME and BALANCE controls, conveniently located in the recessed area at the right side.

(3) An array of control knobs and switches located in the recessed area in the left half of the front panel area.

In the first group, five of the six switches and the LF CAL control have to do with the C-4000's special signal processing circuits and will be discussed later.

4.1 POWER

The POWER switch turns the C-4000 on and off, and also switches power to any equipment (e.g., tuner, power amplifier) whose AC line cord is plugged into the SWITCHED AC outlets at the rear.

To avoid any unpleasant surprises, it is always wise to turn down the VOLUME control before switching the power on.

If you prefer, you may leave the C-4000's POWER switch permanently engaged and use an external switch or a clock timer to turn the stereo system on and off. But if you have a substantial power amplifier plugged into the AC convenience outlets on the rear panel, then such an external switch or timer must be a heavy-duty unit rated to handle the amplifier's AC power requirements.

4.2 HEADPHONES

All conventional headphones (i.e., all except electrostatic models) may be plugged in here; the headphones may have any impedance from a few ohms to several thousand ohms. The signals fed to the headphone amplifier are essentially identical to those fed to the Main Output jacks; i.e., they are equally affected by the Volume, Tone, and other controls — including the Peak Unlimiter, Auto-Correlator, and Sonic Hologram Generator. (Of course the effects which the Sonic Hologram Generator are designed to correct for in normal speaker/room listening don't occur in headphone listening, so the subjective consequences of using the circuit with headphones is different than with speakers. Use it or not as you prefer.)

The HEADPHONES socket is a standard 3-contact 1/4" stereo phone jack accepting normal tip/ring/sleeve plugs. The wiring to the jack is correct for most headphone models: the tip of the plug carries the left-channel signal, the ring contact

on the plug carries the right channel, and the sleeve of the plug is the common ground. In a few headphones the wiring is reversed (tip to right and ring to left); with these you can either reverse the phones on the head or have a technician re-wire the plug on the headphone cable. If you are not sure of the plug wiring, you can check it with the aid of the BALANCE control: as the control is turned to the right (clockwise) the sound should go to the right earcup.

You may freely use headphone extension cables. If you want to use a headphone Y-connector to drive two headsets simultaneously, they should be identical models; connecting together two headphones which differ widely in impedance usually will cause a substantial loss of volume in the headset having the higher impedance (or in both).

Normally when you listen to headphones you should depress the SPEAKERS OFF switch (located between the left and right BASS control knobs) in order to cut off the output signal to the power amplifier, thus muting the loudspeakers. The speakers are not automatically silenced when a plug is inserted into the HEADPHONES jack. With the SPEAKERS OFF button engaged, you can freely adjust the Volume control for headphone listening without fear of accidentally overdriving the loudspeakers.

The HEADPHONES socket can also be used as an extra convenience output (instead of the spare pair of Main Output jacks on the rear panel) to feed signals to a tape recorder or a second power amplifier. The output signal, of course, will be affected by the Volume, Tone, and other preamp controls. You will need an adapter cable (Radio Shack #42-2477 or equivalent) to convert from the 3-contact phone plug to a pair of standard phono jacks.

4.3 VOLUME and BALANCE

The VOLUME control is the master gain control for the stereo system. As a general rule you should turn it down when switching between input signals and when cueing the phonograph stylus in the groove; then turn it up to the desired loudness

level. It is normal to find that the preferred setting of the VOLUME control will differ from one program source to another, from one record to another, and from one FM station to another.

The BALANCE control adjusts the relative gains of the two stereo channels. In its detented position (with the pointer dot at 12 o'clock) the two channels have equal gain. Rotating the control toward the right (clockwise) shifts the stereo image toward the right by reducing the output level in the left channel, and conversely, rotating the control toward the left (counter-clockwise) shifts the image toward the left. The control is designed so that small movements of the control off-center produce small shifts of the stereo balance, while large rotations of the control will turn the unwanted channel completely off.

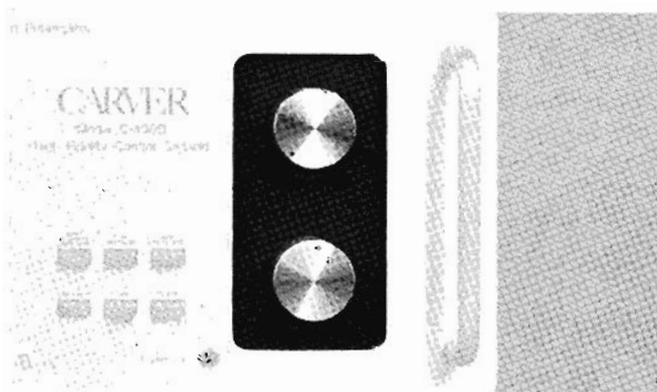
Since obvious left-right balance errors in recordings are relatively rare, it is often assumed that the normal and "correct" setting of the BALANCE control is at its detented center position. But in fact small balance errors, typically 1 to 2 dB in magnitude, are often present in recordings and in stereo components. These are too small to produce an obvious left-right balance shift, but they are still audibly significant because they affect the perceived "depth," detail, and ambience in the stereo image. Feel free to experiment with small rotations of the BALANCE control in either direction from center, to find those settings which yield the best resolution of depth and detail in the stereo sound field.

There are several common causes of small balance errors. One is the phono cartridge: some phono pickups have precisely equal output levels in both channels, but many exhibit channel balance errors of 1 to 2 dB, differing from sample to sample, because of normal construction tolerances.

Tape recorders also commonly have input-to-output balance shifts of 1 or 2 dB because of normal tolerances in the settings of recording bias, internal calibration controls, and the "tracking" of the two sections of ganged Record Level and Output Level controls. Balance errors may even differ from one brand or sample of tape to the next.

Loudspeakers typically differ in sensitivity ("efficiency") by ± 1 dB or more from sample to sample due to normal construction tolerances, and may vary further if you haven't matched the settings of their midrange/tweeter controls. Moreover, if one speaker is located near a wall and the other near an open doorway, their output may be differently reinforced by room acoustics.

In each stage of the stereo system's electronics the two channels can be expected to be well matched. But in an elaborate system with many signal-handling stages (phono preamp, tone-control stage, preamp output stage, peak limiter, equalizer, power ampli-



fier, etc.) the tiny balance errors in each stage can add up to a significant total. Finally, the two channels of the C-4000's own VOLUME control may not track perfectly, especially at low-level settings. (When you find yourself turning the VOLUME control down below 9 o'clock, a better approach is to set the OUTPUT switch to -15 dB instead; this optimizes the system S/N by minimizing residual circuit noise, as well as allowing the VOLUME control to operate in the most linear portion of its range.)

The controls in the left half of the front panel will be discussed in the order in which an incoming signal encounters them, beginning with the Input Selector. (The controls below the Input Selector, which relate to the C-4000's special signal-processing functions, will be discussed in a separate chapter.)

4.4 INPUT SELECTOR

This knob selects the program source which is to be heard; if neither DUB switch is engaged, the same program source is also connected to the tape recording circuits.

When you switch to a position of the INPUT SELECTOR which has no program source connected to it, you may hear some "crosstalk" (leakage of signal from adjacent positions of the switch). This can be eliminated, if desired, by installing "shorting plugs" in the unused input jacks on the rear panel. (Do not install shorting plugs in unused output jacks. This would short-circuit the output of the preamplifier.)

4.5 INFRASONIC FILTER

The selected input signal passes through an infrasonic filter which has no effect on the audible tonal balance of most recordings (even those with prominent deep bass) but rolls off rapidly at frequencies below 20 Hz. When the INFRASONIC FILTER switch is out, this filter is bypassed; when the button is depressed the filter is engaged. It is a good practice to leave the filter switched on permanently.

The filter's "group delay," an unavoidable consequence of its rapid infrasonic attenuation, may produce a just-perceptible alteration of low-frequency transient response. But in general any audible effect of the filter will be beneficial, since reproduction of infrasonic signals yields no benefit. These unwanted signals include switching thumps, turn-on transients of associated components, the thump produced in some FM tuners when tuned past a station, and the transient produced when the phono stylus sets down in the groove. If reproduced at high gain through a powerful amplifier, such signals can damage loudspeakers.

Phonograph output signals are also contaminated with strong infrasonic waveform components due to small and large disc warps, motor rumble (especially in direct-drive units), acoustic feedback and other externally-induced vibrations

sensed by the phono stylus, all of these amplified by the tone arm resonance (due to the stylus compliance interacting with the effective arm mass). These signal components occur primarily in the frequency range from 2 Hz to 15 Hz; if not stripped off the audio signal by an effective infrasonic filter they will tend to waste amplifier power and cause excess woofer cone motion, resulting in audible intermodulation distortion (muddy bass).

Since the INFRASONIC FILTER is located in the preamp circuit ahead of the tape outputs, it will also prevent low-frequency overload of recordings by infrasonic contamination. However, it cannot remove any low-frequency switching thumps produced by the tape recorder itself; so if you have a recorder which produces such thumps when going in or out of the playback mode, reduce the setting of the VOLUME control until the recorder's output has stabilized.

In early-production C-4000 units the input impedance for line-level signals was 10K ohms with the infrasonic filter switched out, increasing to 47K when the filter was switched in. That difference may produce a slight change in volume level as the filter is switched in and out. One advantage of keeping the infrasonic filter engaged is that its higher impedance prevents the low-frequency rolloff that could occur if a program source had a high output impedance and a too-small output coupling capacitor. (In later C-4000 units the input impedance is 47K ohms at all times.)

4.6 TAPE 1 MONITOR, TAPE 2 MONITOR

There are no "tape" settings on the Input Selector switch. To listen to the playback output from a tape recorder, simply depress the appropriate TAPE MONITOR button. These monitor switches permit the output from the tape machine to be heard regardless of the setting of the Input Selector. (If no tape machine is connected to the input jacks, or if one is connected but not operating, then depressing the corresponding TAPE MONITOR button will produce silence.)

The TAPE MONITOR buttons govern only the selection of what signal is heard. They have no effect on the signal which is fed to the preamp's Tape Out jacks for recording; that selection is controlled exclusively by the Input Selector and the two Dub switches. If the Dub switches are disengaged, then the program source selected by the Input Selector is connected to the Tape Out jacks for recording, regardless of the positions of the TAPE MONITOR switches; i.e., depressing TAPE 1 MONITOR to hear the playback from recorder #1 does not automatically permit that playback signal to be copied onto recorder #2.

The principal purpose of the TAPE MONITOR buttons, of course, is to enable you to listen to tape playbacks and to "monitor" the quality of new recordings while they are being made.

In all cases, depressing TAPE 1 MONITOR lets you hear the output from the recorder connected to

the TAPE 1 jacks on the rear panel, and depressing TAPE 2 MONITOR lets you hear the output from recorder #2.

The identity of that output signal depends on the recorder's own controls. In the playback mode, of course, the signal is reproduced from the tape. With a two-head recorder, when you put it into the recording mode, normally the recorder's input signal will also be present at the output jacks after passing through the recorder's Recording Level control and input circuitry. In the case of a three-head recorder, when it is in the recording mode its output signal will either be the input signal or the freshly-recorded playback signal from the tape, depending on the setting of the recorder's own monitor (Tape/Source) switch. So, with a three-head recorder you can instantly compare the recorded signal with the incoming signal and detect any recording faluts.

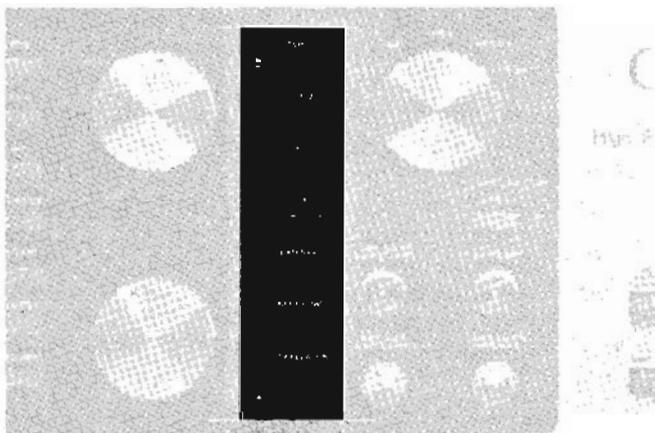
Two special situations deserve mention. (1) If you are using an external "single-pass" noise-reduction system with the recorder (e.g., a Dolby or DBX unit of the type which must be switched between recording and playback modes), then the monitor output of the system during recording may be an "encoded" version of the input signal with a substantially altered tonal balance. (2) If a Dolby-encoded FM broadcast is fed through a recorder having a built-in FM Copy mode, then the recorder's output may be a decoded version of the broadcast. In this case the TAPE MONITOR switch will enable you to hear the decoded sound. Thus the uses of the MONITOR function are varied, but the operating rules are simple:

To listen to phonograph records, radio broadcasts, or program sources connected to the pre-amp's AUXiliary inputs, leave both TAPE MONITOR switches disengaged (out). You will hear the program source chosen by the Input Selector.

To hear the output from tape recorder #1, depress TAPE 1 MONITOR.

To hear the output from tape recorder #2, depress TAPE 2 MONITOR.

If both TAPE MONITOR buttons are depressed, the output from recorder #2 will be heard.



4.7 DUB 1→2 and DUB 2→1

As described above, the TAPE MONITOR switches (in conjunction with the Input Selector) choose what signal you will listen to.

The DUB switches, in conjunction with the Input Selector, choose what signal will be fed to the preamp's Tape Output jacks for recording. Again, the operating rules are simple:

To record from a phonograph record, radio broadcast, or program source connected to the AUX inputs, set the Input Selector to the appropriate input and leave both DUB switches disengaged (out). The selected input signal will be fed to both sets of Tape Output jacks and may be recorded on either tape machine or on both simultaneously.

To copy a tape recording from recorder #1 onto recorder #2, depress the DUB 1→2 button.

To copy a tape from recorder #2 onto recorder #1, disengage the DUB 1→2 button and depress the DUB 2→1 button.

When you finish copying a tape in either direction, be sure to disengage the DUB button so as to permit resumption of normal recording from the Input Selector.

WARNING: Do not depress both DUB buttons at the same time. If you do so while both recorders are in the recording mode (or while both have their own monitor switches set to SOURCE), you may set up a feedback loop in which a signal would circulate at a very high level. Depending on its frequency and volume level, such a feedback signal could damage your amplifier or loudspeakers, and in any event could produce a startlingly loud howl. (If the feedback signal were ultrasonic, as it will be with some recorders, it could burn out your tweeters without even being heard.) Fortunately, there is no situation in which you would ever want or need to engage both DUB buttons at the same time.

As described earlier, when recording from a phonograph record or FM broadcast via the Input Selector, you can hear the program source (with the Tape Monitor buttons out) or you can hear the output of the tape machine while the recording is being made (by depressing the appropriate Tape Monitor button).

Similarly, when using one of the DUB buttons to copy a tape, you can depress either Tape Monitor button to listen to the output of either the source recorder or the copying recorder; i.e., if you are copying a tape from recorder #2 to recorder #1, you can depress TAPE 2 MONITOR to hear the source program; if you disengage TAPE 2 MONITOR and depress TAPE 1 MONITOR you will hear the output of the copying recorder.

While dubbing from one recorder to another, if you leave both Tape Monitor switches disengaged, you will hear the program source chosen by the

Input Selector. Thus you can listen to phonograph records or an FM broadcast while copying tapes; whenever you want to check on the progress of the copying, simply depress the appropriate Tape Monitor button to hear the output of the copying recorder.

We have discussed the use of the dubbing switches with two tape recorders. If you have one tape recorder and a signal-processing accessory (e.g., a graphic equalizer) which you would like to use to process signals before they are recorded, the dubbing switches can be used for that purpose. Connect the tape recorder to the TAPE 2 input/output jacks on the rear panel, and connect the signal processor to the TAPE 1 input/output jacks (instead of the External Processor jacks where it could be used only to process playback signals).

Now, when you press DUB 1→2, the processed signal will be fed to the tape recorder for recording. With the TAPE MONITOR switches disengaged you will hear the unprocessed program signal. By depressing TAPE 1 MONITOR you will hear the processed signal. And by depressing TAPE 2 MONITOR you will hear the processed signal at the output of the recorder. If you decide that you would prefer to record the unprocessed signal, disengage the DUB 1→2 button and the unprocessed signal from the Input Selector will be fed directly to the recorder.

If you want to process a tape recording during playback rather than during recording, disengage DUB 1→2 and depress the DUB 2→1 button. The playback output of the recorder will be fed directly to the signal processor. If you then depress TAPE 2 MONITOR you can hear the unprocessed playback signal. If you disengage TAPE 2 MONITOR and depress TAPE 1 MONITOR you will hear the processed playback signal.

4.8 EXTERNAL PROCESSOR

Any external signal-processing accessory, such as a graphic equalizer or DBX disc decoder (for processing playback signals only), can be connected to the EXTERNAL PROCESSOR input/output jacks on the rear panel, and can be engaged or bypassed at will by means of the EXTERNAL PROCESSOR switch.

Thus if you have a special loudspeaker equalizer connected to these jacks, you should depress the EXTERNAL PROCESSOR button to engage the equalizer when listening to the speakers which require the equalization. When listening to headphones or to other speakers which don't require the equalization, switch out the EXTERNAL PROCESSOR.

If you have no equipment connected to the External Processor jacks, or have equipment connected but not operating, then depressing the EXTERNAL PROCESSOR switch will produce silence.

You may connect an extra tape recorder to the External Processor jacks and use the EXTERNAL PROCESSOR switch as another Tape Monitor switch. A recorder connected to these jacks can be used to record from the program sources chosen by the Input Selector switch, and can also copy tapes from recorders connected to either set of Tape input/output jacks (by engaging the corresponding Tape Monitor switch). In this case the DUB switches are not used. Remember, however, that the Autocorrelator and Peak Unlimiter circuits cannot be used with signals from the External Processor inputs.

4.9 STEREO/MONO

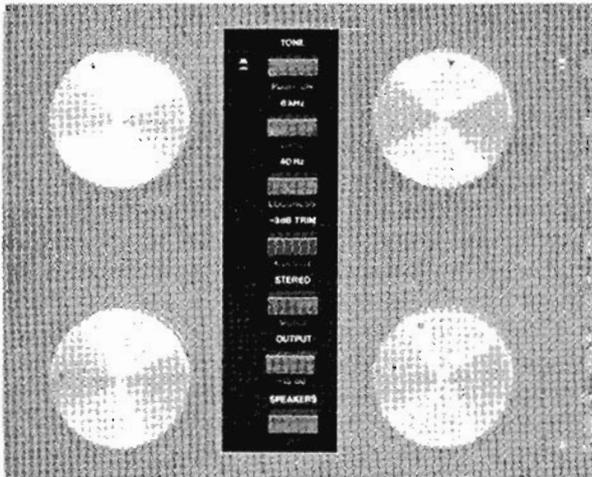
Depressing this switch combines the two stereo channels together into mono. This has two principal applications. (1) When playing old monophonic records, switching the preamp to mono will cancel any vertical rumble and vertically-modulated groove noise, yielding quieter playback. (2) When listening to a monophonic program source (e.g., TV audio) which is connected to only one input jack, switching to mono will couple the signal into both output channels and thus to both loudspeakers for listening.

For all normal listening to conventional stereo program sources, the STEREO/MONO switch should be disengaged (out).

4.10 TONE CONTROLS

The C-4000 is equipped with separate Left and Right controls for adjusting the Bass and Treble tonal balance in each channel. Normally, of course, the tone controls should be adjusted identically in both channels, and the knobs are detented to make it easy to achieve equal settings. However there are some circumstances in which different settings in the two channels are useful — such as a phono cartridge or tape recorder with different response in the two channels (this problem is not as rare as it should be), or an asymmetry in the acoustics of the listening room. The best procedure in such situations is to start with identical signals in the two channels (e.g., a monophonic record or FM interstation hiss) and adjust the relative settings of the Left and Right tone controls, listening alternately to one channel and then the other, until the two channels sound as nearly alike as possible.

In general, moderate rotations of the Bass and Treble controls away from center will prove to be most useful in correcting the tonal balance of recordings. Maximum boost of either Bass or Treble should be used with caution and only at modest loudness levels. Sustained operation at high volume levels with full Bass or Treble boost may call for more energy than your power amplifier can deliver and more than your loudspeakers can safely accommodate.



TONE on-off. The top three switches between the Treble knobs affect the operation of the tone controls. When the TONE button is disengaged (out), the tone control circuits are completely bypassed, and adjustment of the Bass and Treble controls has no effect. The TONE button must be depressed in order for the tone controls to affect the sound. Thus you may set the Bass and Treble controls to a desired setting and then, by switching the TONE button in and out, make instant comparisons of the corrected sound versus the "flat," uncorrected sound to evaluate the effect of the tone controls.

8 kHz/2 kHz. The 8 kHz/2 kHz switch selects the "turnover frequency" of the Treble controls. In the 8 kHz position (out) the Treble controls

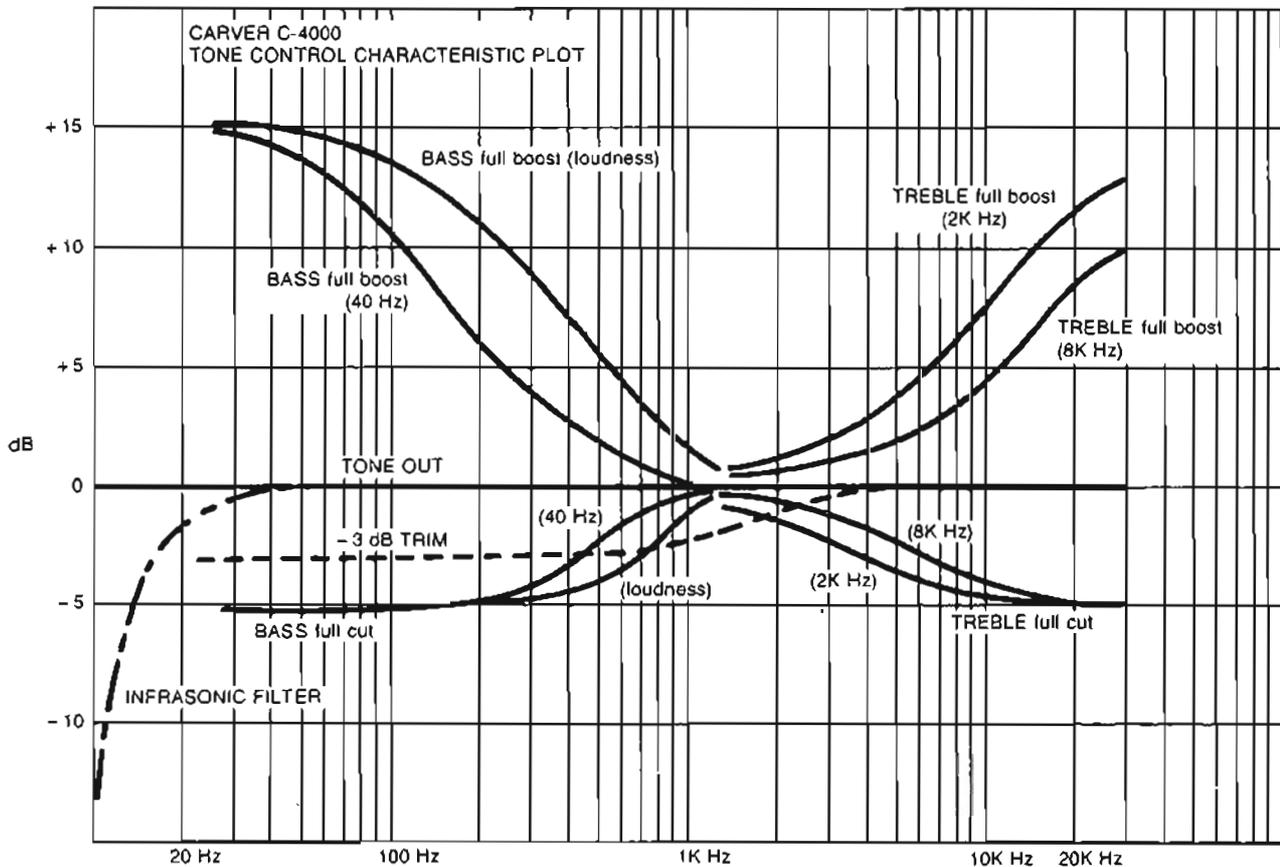


Figure 4-1. TONE CONTROL CURVES

These curves illustrate the maximum boost and cut provided by the Bass and Treble controls at each setting of their respective turnover switches, together with the response of the infrasonic filter and -3 dB TRIM circuit.

The Bass and Treble controls behave conventionally in the boost position, but they change their characteristics altogether in the cut position. Rather than rolling off the Bass or Treble, they merely reduce the Bass or Treble energy level (at the rate of about 1 dB for each click-stop) while maintaining a relatively flat frequency response within their assigned ranges.

This characteristic is very useful, for example, if a recording sounds too bright; you can reduce the relative high-frequency energy in the sound without rolling off the extreme highs. By maintaining a level Treble response all the way to 20 kHz when the Treble control is in the cut position, the "air" and sheen in the sound may be retained in a recording while excessive brightness is reduced.

The tone controls have been designed to be musically useful, and we encourage their routine use to enhance your enjoyment of musical sound.

affect only the highest audible frequencies, particularly the subtle overtones and "air" in the sound. Most music has relatively little energy at the frequencies in the top octave (indeed, in some recordings, the top octave contains mainly noise and distortion with virtually no musical energy), so at this setting of the turnover switch, adjustments of the Treble control have a subtle effect on the perceived tonal balance. In the 2 kHz setting of the switch the effect of the Treble control is broadened and a boost or cut of the Treble controls has a more obvious effect on the overall tonal balance.

40 Hz/LOUDNESS. This switch selects the turnover frequency of the Bass controls. In the 40 Hz setting (out), most of the action of the Bass controls occurs at the lowest audible frequencies, i.e., in the deep bass. Since most recordings have little energy in the deep bass, the effect of Bass control adjustments will usually be rather subtle. But with those recordings which contain significant deep-bass energy, and with loudspeakers having authentic low-end response, the 40 Hz setting of the Bass control can be used to bring up the solid foundations of the musical frequency spectrum to a satisfyingly audible level without encountering boominess due to excessive mid-bass boost.

When the 40 Hz/LOUDNESS switch is depressed, the effect of the Bass controls is broadened by two octaves to include the mid-bass range as well as the deep bass. Then the Bass controls have an obvious effect on the tonal balance of most recordings, and the Bass boost contour follows that of the traditional Fletcher-Munson "loudness compensation" curves. The impact of instruments such as bass guitar, kick drum, and tympani is due primarily to energy in the 60-150 Hz range, so the LOUDNESS setting of the bass contour switch is the preferred setting when you want to accent the sound of these instruments.

4.11 - 3 dB TRIM

This switch is intended primarily for use with the Sonic Hologram Generator, but it also serves as a useful general-purpose extra tone control. When depressed it introduces a "shelf" in frequency response, reducing the level uniformly by 3 dB at all frequencies below approximately 800 Hz. This can be helpful to improve the tonal balance of a muddy, thick-sounding recording. It also serves to correct the tonal balance of speakers which were not designed to take into account the effects of bass reinforcement when placed near a wall or floor.

This shelving equalization is switched in or out of the circuit by the -3 dB TRIM button. It is separate from, and is unaffected by the settings of, the tone control circuits. It is not affected by the TONE on/off bypass switch.

4.12 OUTPUT

When this switch is disengaged (out) the preamp operates with its full normal gain. Depressing this switch reduces the output level of the preamp by 15 dB in both channels.

This switch provides a convenient method of temporarily reducing the volume (e.g., when answering the telephone, or cueing the phono stylus in the groove) and then returning precisely to the pre-set level.

It also provides a way of extending the useful range of the Volume control. If you have a high-gain power amplifier and relatively sensitive (efficient) loudspeakers, you may find that, for the loudness levels you prefer, the Volume control is always confined to the lowest portion of its range, and turning it up past 12 o'clock produces excessively high sound levels. In this case, by depressing the OUTPUT switch to reduce levels by 15 dB, you can comfortably operate the Volume control through the middle portion of its range.

A good rule is to engage the -15 dB switch whenever you find yourself operating the Volume control below about 9 o'clock — for example, when listening to quiet music late at night, or when using headphones. The -15 dB setting helps to maintain the best signal-to-noise ratio by reducing the residual circuit noise levels in the preamp itself.

If you find yourself using the -15 dB setting virtually all the time, you may be interested in a simple modification which your dealer can make to reduce the output level of the C-4000. Its output stage gain is "programmable" and can be decreased by 3, 6, or 8 dB by changing a resistor in each channel.

4.13 SPEAKERS OFF

This switch silences the output of the C-4000 at the Main Output jacks, without affecting the signal fed to the HEADPHONES jack or to the TIME DELAY circuits. By depressing the SPEAKERS OFF button and also switching off the TIME DELAY you can mute the loudspeakers in order to listen to headphones.

This completes the description of the "conventional" controls and functions of the C-4000 preamplifier. The remaining front-panel controls are for the special circuits developed by Carver: the "Peak Unlimiter" dynamic range expander, the "Auto-Correlation" noise-reduction system, the "Sonic Hologram" stereo image processor, and the Time-Delay ambience reproduction system. These circuits go beyond the conventional hi-fi goal of simply providing an accurate reproduction of the signal in the recording. They approach the original and true goal of high fidelity: the life-like re-creation of the experience of hearing live music itself.

5.0 PEAK UNLIMITER Dynamic Range Expander

5.1 WHAT IT DOES

Much of the energy and vitality of live music has to do with its dynamic range, its soft/loud contrasts, from the poetic pianissimos and dramatic quiet pauses in which an entire audience collectively holds its breath, to the exciting wall-shaking fortissimos when an entire orchestra strokes, bangs, and blows at full blast. But when music is recorded its dynamic range is usually constricted to fit into the limited modulation range of record grooves and within the limited signal-to-noise ratio of tape recorders. And when recordings are broadcast, additional dynamic compression and limiting are usually applied at the transmitter.

Of the various kinds of dynamic-range processing which are applied in recording and broadcasting, the most common is "limiting," in which the median volume levels are left unaltered but the crescendos and bursts of high energy are squashed. In effect a low-headroom ceiling is put on the music, flattening its peaks.

The Carver Peak Unlimiter reverses this process, expanding the compressed dynamics of recordings back to their full range, effectively restoring the dynamic contrasts which the composer or performer originally intended.

The Peak Unlimiter increases the dynamic range of the signal by 4 to 5 dB, which means that the peak power in transients is potentially multiplied by 2½ to 3 times. In order to minimize the likelihood that this boost would overdrive your power amplifier into distortion, the overall volume level is slightly reduced when the PEAK UNLIMITER pushbutton is engaged; thus, while the peak levels in the music are boosted, the softer levels in the music are made slightly quieter.

5.2 THE CONTROLS

The dynamic range expander is easy to use. Simply depress the PEAK UNLIMITER pushbutton to activate the circuit, and then adjust the PK UNLIMIT threshold control so that the adjacent LED just flashes during the climaxes and high-level transient peaks in the music. Since the highest-level

peaks in music often occur during brief transients, you will find that the Peak Unlimiter often restores the missing vitality, punch, and impact of the music without actually making it seem louder. (If you adjust the PK UNLIMIT threshold control so that the LED stays illuminated for prolonged periods instead of just flashing on peaks, then you are causing the circuit to boost the dynamic range of sustained portions of the music, which probably will make them sound louder. But you may be expanding the dynamics of portions of the music which were never compressed.)

6.0 AUTO-CORRELATION

Noise Reduction System

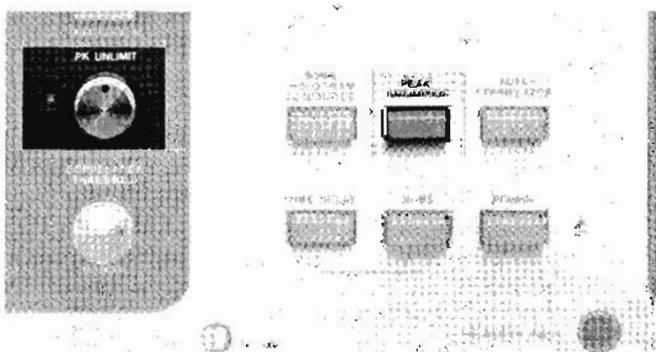
6.1 WHAT IT DOES

One of the important barriers to realism in reproduced music is the veil of low-level noise in all recordings — the rumble and surface noise of records, the background hiss in tapes, the residual hum and hiss in an FM tuner. Subconsciously we try to acclimate ourselves to such noises, mentally tuning them out so as to focus on the sound of the music itself; but the need for such continuous subliminal effort may be one of the causes of "listener fatigue." It is difficult to recognize what a psychoacoustic barrier surface noise is, until you try to return to normal listening after experiencing how remarkably relaxing and pleasurable music can be without such noise.

Unlike the Dolby and DBX noise-reduction systems, which can work only by specially encoding the recording at its source and decoding it in playback, the patented Carver Auto-Correlator noise-reduction system operates with all recordings and broadcasts, mono or stereo, disc or tape. Like the brain, it discriminates between noise and music by recognizing their differing natures. Surface noise and tape hiss are random, while music is a mathematically "correlated" signal, i.e., a patterned signal in which, for instance, high-frequency overtones are mathematically related to each note's fundamental frequency.

The Auto-Correlator noise-reduction system contains a sophisticated bank of comb filters (notch filters spaced at multiples of a common frequency), controlled by an analog computer which continually tracks the musical signal and computes the frequencies at which overtones can occur. At each moment in time it "opens" those filters where musical overtones can occur and "closes" those filters in which only noise is found.

Simultaneously a separate, independently adjustable low-frequency filter blocks any turntable rumble and hum below its "LF CAL" threshold. This dynamic filter has flat response to 20 Hz when music is present, but sharply attenuates the bass below 200 Hz when no music is being heard.

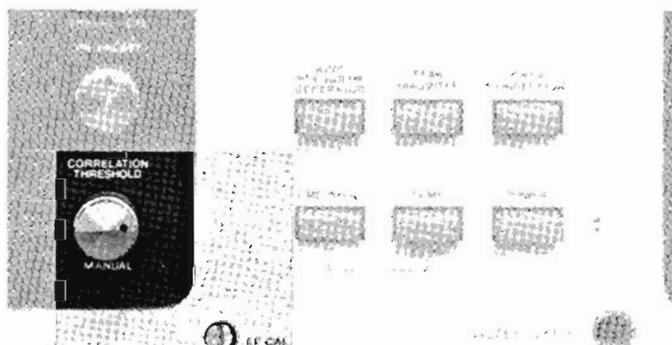


6.2 THE CONTROLS

The noise-reduction system has four operating controls.

AUTO-CORRELATOR. This pushbutton switch engages the noise-reduction system (button IN) or bypasses it completely (button OUT).

CORRELATION — AUTOMATIC THRESHOLD. This pushbutton selects automatic (button IN) or manual (button OUT) adjustment of the sensitivity threshold for the auto-correlator hiss-reduction circuit. The system contains an analog computer which continuously tracks the level of high-frequency noise in the signal and automatically adjusts the auto-correlation threshold to suppress that noise. You probably will find it convenient to leave it in this automatic mode for most listening. Sometimes, however, especially when listening critically, you may find that you can make a slightly "smarter" choice of threshold than the automatic circuit can; for those occasions, switch the button OUT and use the manual threshold control.



CORRELATION THRESHOLD — MANUAL.

This control is engaged when the Correlation — Automatic Threshold pushbutton is switched OUT. To use it, find a quiet passage in the recording where you can clearly hear the surface noise or tape hiss. Begin by rotating the Manual threshold knob fully clockwise (to the right), and depress the AUTO-CORRELATOR pushbutton to activate the circuit, if you haven't already done so. Listen to the hiss and slowly rotate the knob counterclockwise until you hear a sudden decrease in the noise level. If you rotate the knob further to the left the high-frequency overtones in the music will rapidly become muffled also. Turn the knob back to the right a little to restore the high frequencies in the music, but not so far as to restore the unwanted hiss.

The setting of this threshold may vary from one record to the next, and will probably vary from one program source to another (e.g., from record to tape).

At first you may find the performance of the auto-correlator difficult to accept; you may sense

an apparent loss of highs even when it is correctly adjusted. This is due to a psycho-acoustic illusion in which the ear equates the presence of high-frequency noise with an extended high-frequency bandwidth; our hearing mechanism even interprets random-phase noise content as added "air", depth, and detail in the recording itself. So while the auto-correlator may at first seem to be dulling the highs, you should give yourself a chance to acclimate to the true tonal balance of music uncontaminated by noise. (Incidentally, many loudspeakers have a tweeter level control which is usually set for a mild high-end rolloff in order to reduce the annoyance value of noise; you may find that the auto-correlator enables you to enjoy loudspeakers with a more nearly flat high end.) After you have lived with the auto-correlator for a while, when you switch it out you may be astonished to learn how severely "normal" amounts of noise detract from musical enjoyment.

LF CAL. This screwdriver-adjust control separately sets the operating threshold for the low-frequency noise-reduction circuit, independent of the auto-correlator hiss threshold. Its purpose is to suppress the low-frequency hum and rumble of your turntable. There are two methods of adjusting the LF CAL:

(1) Switch to PHONO, place the stylus in a blank groove (the lead-in groove, blank groove between selections, or a blank groove on a test record), turn up the Volume and Bass controls so that you can hear the low-frequency hum and rumble clearly, and adjust the LF CAL control with a screwdriver. Begin with the control turned fully clockwise (to the right), and slowly turn it counterclockwise (to the left) until you hear a decrease in the low-frequency noise level. Stop at that point; if you turn the control further to the left, you may begin to attenuate low-frequency musical energy and authentic hall ambience in the recording. Normally you only need to make this adjustment once — until you install a different cartridge, alter the tone-arm damping, or change to a different turntable.

(2) Many good turntable/cartridge combinations yield so little audible hum or rumble that the preceding method of adjustment is difficult to carry out. In that case, use this alternate procedure. Set the tone controls for maximum bass boost, switch on the auto-correlator, and set the LF CAL control fully clockwise (to the right). Play a record that has deep-bass musical energy. Rotate the LF CAL control fully counterclockwise and verify that you hear an audible decrease in deep-bass level. If not, select another record which does have authentic deep-bass signals (e.g., bass drum, pipe organ pedal notes, or the low-frequency portions of a frequency-response test record), in

which you can identify the decrease in bass caused by rotation of the LF CAL control over its full range. Now rotate the LF CAL control slowly clockwise (to the right) until the bass response is restored to its original level. Continue to rotate the LF CAL control about 1/8 turn more to the right, and stop. This ensures that the noise-reduction will not alter the musical bass content of recordings.

7.0 SONIC HOLOGRAM Stereo Image Processor

7.1 WHAT IT DOES

“Sonic Holography” is a method of processing stereo signals so as to correct a basic imaging flaw which is inherent in two-channel stereophonic recording and reproduction via loudspeakers. What that flaw is, and why it can't be corrected by any conventional recording technique, are discussed in detail in Appendix A (Chapter 9 of this manual), “The Theory of Sonic Holography.”

You don't need to study that theory in order to make it work for you; this chapter will explain fully how to adjust your loudspeakers and set the controls of the Carver Sonic Hologram Generator. We suggest that you peruse the Appendix at your convenience, in order to understand the logical underpinnings of the Hologram Generator controls and the instructions which follow.

Briefly, the problem is that in stereo listening, both ears hear the outputs of both loudspeakers. When a sonic event such as a musical transient is reproduced by the Left-channel loudspeaker, the sound travels in a straight line from the speaker to your left ear. A tiny fraction of a second later the same Left-speaker sound arrives at your right ear, somewhat filtered by the obstruction of your head. If the same sonic event was recorded in both stereo channels, as is normally the case, then some version of it will also be reproduced by your Right-channel speaker, whose sound will arrive first at your right ear and then, a tiny fraction of a second later, at your left ear. Thus the single original sonic event is represented by a total of four sound arrivals at your two ears.

In real life a single sonic event can never cause more than two sonic arrivals: one at your left ear and one at your right ear. (Which ear gets the sound first depends on which direction you are facing, relative to where the sound is coming from. If you are facing the sound, it will arrive at both ears simultaneously.)

The goal of the Carver Sonic Hologram Generator is to eliminate the “extra” two sonic arrivals that occur in stereophonic playback but do not occur in real life. With these eliminated, the ear/brain system of the listener will receive unambiguous timing and phase information about the original sounds as they struck the recording micro-

phones. Without extra sonic arrivals to confuse it, the ear/brain system will be able to perceive the true location of each sound source in the stereo recording — not only from left to right but also from near to far in space.

This is accomplished by cancelling the unwanted second arrival of the sound from each speaker to the opposite-side ear, so that each ear is free to concentrate its attention on the signal from the speaker on the same side; i.e., the left ear will hear mainly the Left speaker, and the right ear will hear mainly the Right speaker, without the confusing acoustic crosstalk which normally occurs in stereo playback.

How it works. The circuit produces electronic crosstalk signals from each channel into the opposite-side speaker. These are virtually identical to the unwanted acoustic crosstalk second-arrival signals which flatten the image in stereo. They have essentially the same delays as the second-arrival signals, and about the same filtering as that caused by the blockage of your head, but they are inverted in phase. Consequently, when these mirror-image crosstalk signals are reproduced by your speakers they cancel the acoustic crosstalk signals arriving at each ear from the opposite speaker.

In order for this process to work well, the second arrival at each ear from the opposite-side speaker must be synchronized with the delays in the circuit. This requires that the speakers be located at **exactly equal** distances from your chair. Moreover, the effect may be largely nullified if strong reflections off walls or other surfaces generate additional “second arrivals” to confuse the perceived stereo image. Thus you must devote greater than usual care to locating the speakers and dealing with the acoustical environment around them.

This may seem like a lot of bother — and it is. But this special care in setting up will be well rewarded when you hear the full benefit of sonic holography — the wide stereo stage extending far beyond the speaker positions, the precise localization of sounds in depth, the almost palpable feeling of three-dimensional ambience filling the space around you.

7.2 MAKING HOLOGRAPHY WORK: CONTROLLING REFLECTIONS

In any listening room much of the output of the loudspeakers reflects off walls and furnishings and eventually arrives at the ear. Now the essential object of sonic holography is to give each ear a clear opportunity to lock onto the primary direct sound from the same-side speaker, without the image confusion produced by acoustic crosstalk from the opposite-side speaker. Understandably, the ear's perception of the first-arrival image can also be confused by any other early arrivals which follow too

quickly after the direct sound — such as reflections off surfaces close to either loudspeaker.

Therefore it is strongly recommended that both loudspeakers be located at least two feet out from walls in any direction — more if possible — in order to maximize the delay time of any reflections. The advantage of such placement, in dramatically easing the task of obtaining the full benefit of sonic holography, can hardly be over-stressed.

Speaker Placement

Even if your loudspeakers were specifically designed for placement against a wall or in corners, we urge you to experiment with moving them out into the room to get them away from reflecting surfaces. In most cases the speaker manufacturer's recommendation of wall or corner placement is made only for the sake of obtaining the strongest and smoothest low-frequency output; but you will find that with many recordings the Carver Sonic Hologram Generator yields a powerful, room-filling quality of bass sound which overcomes any bass loss produced by moving the speakers away from walls or corners. Try it and see. Hint: to preserve the smoothest bass response, avoid any speaker location which will cause the woofer to be equally distant from the side wall, rear wall, and floor. For instance, rather than placing it 2 feet diagonally out from a corner, try staggered distances such as 1.5 feet above the floor, 2.5 feet from the side wall, and 3.5 feet from the rear wall.

If you are like many audiophiles, you already have installed your loudspeakers in locations which you consider optimum for your room, either with respect to stereo imaging or to solve the practical problems of integrating speakers with the decor and furnishings of a living room, and you may be reluctant to move them. But many arrangements which are satisfactory for stereo are problematic for sonic holography; with just a little experimenting you may discover a setup which works well for holographic imaging and satisfies your other requirements too.

For example in rectangular rooms loudspeakers are very often located along one of the shorter walls, firing down the length of the room. In order to obtain a suitably wide stereo image the speakers then must be placed rather close to the corners and adjacent side walls, generating strong early reflections. But sonic holography produces an image which extends substantially beyond the angular spread of the speakers, so you can move the speakers closer together without sacrificing spaciousness; indeed, a closer-than-usual spacing of the speakers generally improves the performance of the Sonic Hologram Generator, stabilizing the 3-D holographic image and enlarging the listening area in which it can be fully enjoyed. Thus if you have your speakers nine

feet apart in a 12-foot wide room (each only 1½ feet from the adjacent side wall), try spacing them only six feet apart instead. You may find that getting them a full three feet away from corners improves their tonal balance in stereo as well as helping the hologram generator to work better.

In many rooms an even better arrangement is to place the speakers along the longer wall of the room, firing across the shorter dimension. This automatically tends to place the listening area closer to the speakers and thus more in their direct sound field rather than in the reverberant field formed by the room's many reflections, and of course the corners and adjacent side walls are far away at the ends of the room. This room set-up is also optimum for the use of time-delay since the ambience speakers can be placed at the ends of the room far from the main listening area.

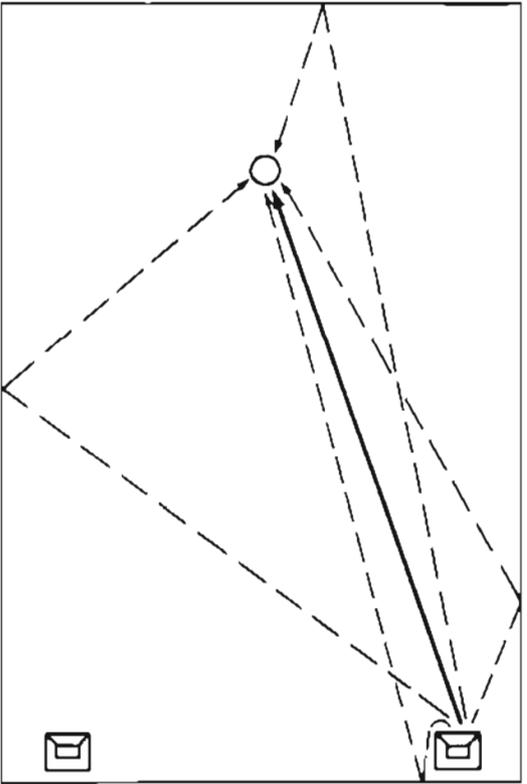
In many homes it is admittedly impractical to have the speakers freestanding in the middle of the room, because they may block traffic paths as well as being unsightly. But don't ignore the substantial benefits which may be gained by moving the speakers out from the wall as little as a foot or two instead of having them flush against the wall. A modest forward movement of the speakers often elicits a significantly more "open" sound in conventional stereo operation as well as better holographic imaging.

Reflections off the floor, especially at midrange frequencies, can be as deleterious to stereo and holographic imaging as reflections off walls are. Unless your speakers were specifically designed to be floor-standing, they probably shouldn't be on the floor. (This is especially critical with two-way speakers, in which the woofer reproduces the midrange as well as the bass.) With rare exceptions loudspeakers in cabinets perform best when raised off the floor on stands or shelves so that the midrange and tweeter are approximately at ear level.

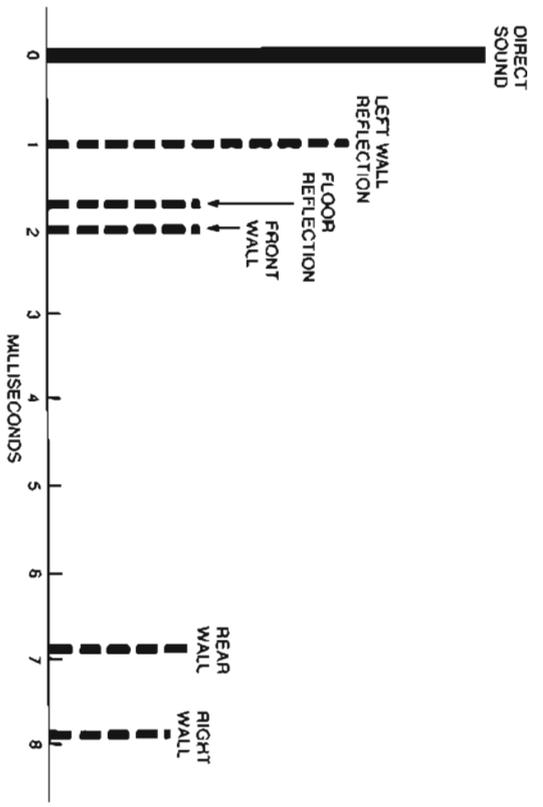
Dealing with Reflections

We very strongly recommend that you locate the speakers at least two or three feet away from all walls, as discussed in the preceding paragraphs. If this is simply impractical for you, or if you have to settle for a placement compromise (for example, with the speakers well away from side walls but still placed on shelves with their backs against a wall), then you should mount a direct attack on the resulting early reflections. Until you do, you will not obtain the full benefit of sonic holography.

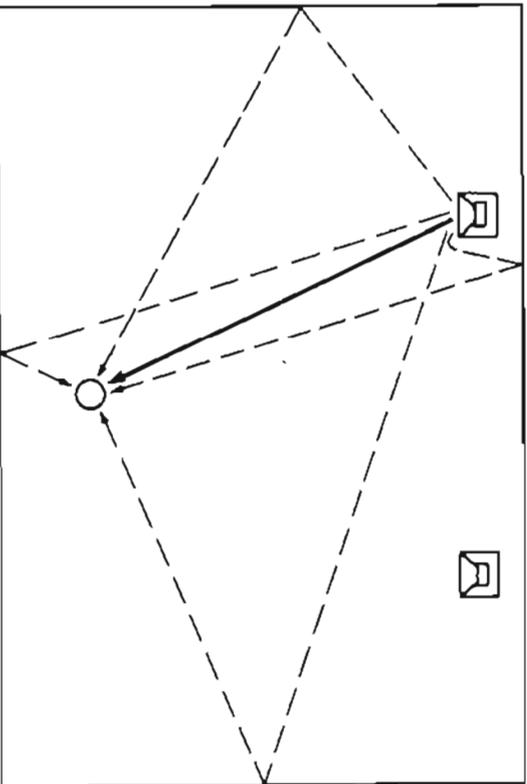
The basic idea is called "dead-end, live-end" room design: the area around the loudspeakers is made acoustically "dead" to suppress early reflections, while the area around the listener is kept



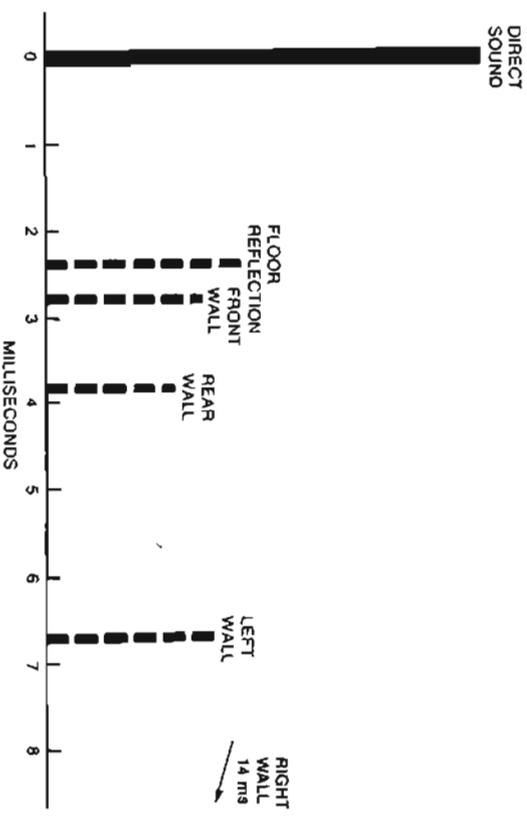
The sketch above shows a speaker installed in a room in a conventional location with its back against a wall and not far from a side wall, i.e. close to a corner (1 1/2 feet away). The sketch shows the path of the direct sound (solid line) and the principal early reflections. The chart to the right shows the resulting pattern of early



sound arrivals at the listener's ear (in terms of relative intensity and time), showing that each direct sound is closely followed by a cluster of early reflections which muddy the holographic imaging.



This sketch shows a better arrangement of the speaker in the same 12 x 18 foot room; the speakers are now five feet away from the side walls, substantially weakening the mirror-image wall reflection and moving it six milliseconds later in time where it becomes innocuous. The speaker has also been pulled forward from



the wall a few inches, delaying the reflection off the front wall by nearly a millisecond. As the chart in Part B illustrates, the ear now has a much better chance to lock onto the holographic image formed by the direct sounds, before room reflections start arriving.

Figure 7-1. SOUND RAYS AND REFLECTION TIMES IN TWO ROOMS

acoustically "live" so that random reflections (arriving long after the direct sound) will establish a desirably uniform sound field.

For instance if your loudspeakers are placed on either side of a large window, drapes must be used to suppress reflections off the glass — but not just any drapes. Heavy foam-backed drapes provide much more absorption than plain cloth or fiberglass weave, and provide useful thermal insulation as well. A single layer of cloth stretched across the glass absorbs only at high frequencies, but deeply pleated draperies are effective absorbers over a broad frequency range.

In general, if your speakers are mounted against a wall, the area of the wall around — and especially between — the speakers should be treated to minimize reflections. Drapes, pleated and hung a few inches away from the wall, are effective, as is a large cork panel (which can also serve as a bulletin board for tacking up notes, clippings, and photos). Another excellent approach is to surround the loudspeakers with bookshelves and open record cabinets whose front surface is approximately flush with the front panel of the speaker. (Of course this only works if they are reasonably well-filled with books or records; empty shelves simply produce an undesirable resonant cavity for the sound.)

The reflection which usually is most deleterious to stereo and holographic imaging is the strong, mirror-like first reflection off the side wall near each speaker. So, while you may profitably apply sound-absorbing treatment to the entire side wall near each speaker, it is especially important to identify and treat the specific area on each wall where the primary early reflection occurs. Usually this will not be directly adjacent to the speaker but two or three feet in front of it.

You can locate this area easily with the aid of a helper and a flat mirror: sit in your normal listening chair (on the system's stereo axis, equidistant from the two speakers), have your helper hold the mirror flat against the wall and slide it along the wall until you can see the loudspeaker's image reflected in the mirror. Repeat the process on the opposite wall to find the image of the other speaker.

Apply sound absorbing treatment to an area a foot or two in diameter around each mirror-reflection location on the walls. (The treatment need not span a vertical dimension as large as the full height of your speakers if they are tall floor-standing models; just be sure that the treatment covers wall areas at the same height as the midrange and tweeter.) If you are reluctant to add drapes or sound-absorbing panels to those wall areas, an effective compromise is to cover those wall areas with bookshelves filled with books of varying depth; this will produce random, diffuse scattering of the sound rather than the pronounced mirror-like reflection off the bare wall.

As noted earlier it is also possible to get strong reflections off a bare floor, coloring the midrange

sound and diminishing the depth imaging of the stereo system. These reflections should be treated by covering the appropriate floor areas with carpeting (preferably a deep-pile plush carpet with rubber or foam backing, because a thin scatter rug will not provide much absorption at midrange frequencies). Or you may obstruct the path of the floor reflection by placing an upholstered hassock or other piece of low furniture on the floor area where the primary reflection will occur.

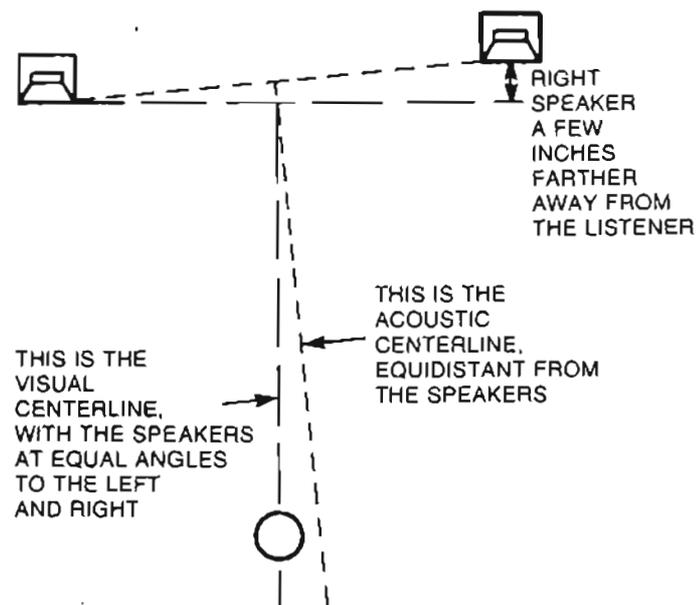
You can locate the spot on the floor in front of each speaker where the strong primary reflection will occur by again sitting in your normal listening seat; have a helper lay a mirror flat on the floor and slide it along until you see the speaker's reflection in the mirror. You may want to remove the speaker's grille, or tape a ribbon on the grille in front of the midrange, so that you can identify in the mirror the spot on the floor where midrange reflections occur — unless you have two-way speakers, in which case the woofer's reflection is equally important.

One more general rule: whenever possible it is a good idea to "toe in" the speakers, placing the listening area on-axis for each speaker rather than 30 degrees off-axis, and weakening the off-axis sound radiated toward the side walls.

7.3 MAKING HOLOGRAPHY WORK: FINDING THE STEREO AXIS

Setting up a stereo system for sonic holography basically involves just two steps. Both of these steps are beneficial in stereo listening and crucial for successful holography. One of them is the control of early reflections, already discussed. The other consists simply of placing the listener on the stereo axis equidistant from both loudspeakers.

As described in the "Theory of Operation" section, sonic holography operates basically by feeding to each loudspeaker a phase-inverted sample of the



signal in the opposite-channel speaker, delayed in time by a small fraction of a millisecond; when this signal arrives at your head it cancels out the acoustic crosstalk from each speaker to your opposite-side ear. Obviously, in order for this cancellation to take place, the timing of signal arrivals at your head is crucial. If one speaker is a few inches farther from you than the other speaker is, the arrival timings will not be correct and the holographic effect will be substantially diminished.

It is NOT sufficient just to visually estimate the centerline of the system. When attempting to align your chair on the visual centerline midway between the speakers it is impossible to tell whether you are a couple of inches closer to one speaker than the other — but a two-inch error in distance can make a dramatic difference in holographic imaging. The only way to be sure is to measure the distance from your chair to each speaker.

Note that the goal is not to optimize the lateral (left-right) centering of your chair but to set it equally distant from the two speakers. Ideally the speakers should be equidistant from your chair to within an accuracy of about 1/4 inch. But this does not mean that your head must also be laterally centered to the same accuracy.

Speaker alignment can easily be established with the aid of a tape measure, a string, or a spare length of speaker wire. Fasten one end of the string to the center of your chair, or have a helper hold it firmly in place (either at the center of the seat or the center of the chair's back). Pull the string taut (but don't pull so hard that you stretch the string and alter its length), mark it where it touches the center of either speaker's front panel, and then note whether the same point on the string just reaches the other speaker's front panel. If not, you

have two options: either move one of the speakers slightly forward or back until both are equally distant from your chair (easy to do if they are freestanding in the room, not as practical if they are shelf-mounted against a wall); or move your chair laterally toward the more distant speaker to compensate for the difference in distance. The required lateral shift will be larger than the difference in distance — typically two or three times larger. So if your right speaker is three inches farther from your chair than your left speaker is, move the chair at least six inches to the right and then repeat the string test, until you have the distances to both speakers equal.

If you have odd-shaped speakers and are not sure where to measure to, measure to the center of the midrange driver. If your speakers are very wide, the best method is to "toe in" the speakers toward your listening chair; measure to both the inside and outside corners of each speaker from your chair, and adjust to make both sets of measurements equal.

7.4 MAKING HOLOGRAPHY WORK: THE CONTROLS

All of the work in sonic holography is in the initial set-up that you have now completed — controlling early reflections and matching the speaker-to-listener distances. The routine operating controls are quite simple and seldom need adjustment.

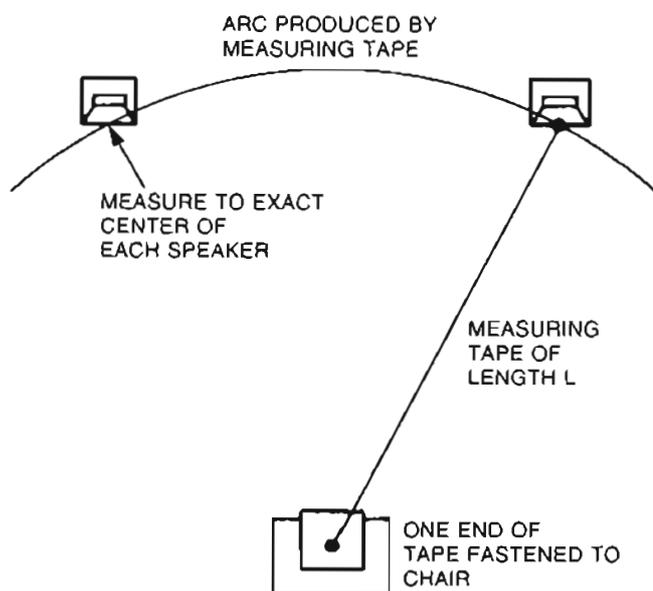
Sonic Hologram Generator (on/off). This push-button switches the entire holographic imaging circuit into the preamplifier's signal path (button IN) or bypasses it (button OUT).

Holographic Listening Angle. This switch is located on the rear panel. We suggest that you leave it permanently set at "Wide."

The "Normal" setting provides precise matching of the electrical and acoustical crosstalk signals, but it requires extremely careful adjustment of speaker distances, and a precisely maintained location of the listener's head on the stereo axis, in order to provide the full holographic effect. In the "Wide" setting the circuit parameters have been adjusted to provide a wider listening zone which is more convenient and practical for day-to-day listening, with no diminution in the quality of the holographic image.

Holographic Injection Ratio (Normal/Theoretical). This pushbutton varies the strength of the electronic crosstalk signal and thus alters the apparent character of the holographic image. Use either setting, as you prefer; you may find that the preferred setting varies from one record to the next, or from one kind of music to another.

In the "Normal" setting a conservative level of the phase-inverted electrical crosstalk is injected.



This tends to produce the most natural imaging with well-made recordings in which a minimum number of microphones were used in order to capture a natural stereo perspective. In this setting, there is little or no change in the overall volume and tonal balance of the sound when the SONIC HOLOGRAM generator is switched in and out of the system.

In the "Theoretical" setting the crosstalk cancellation signal is stronger, producing a more obvious change in the imaging. This setting is recommended for recordings which need (or can benefit from) more dramatic enhancing. It tends to produce more "wrap-around" of images to the sides of the listener, creating a stereo stage much wider than the angle between the speakers. With more energy at the sides, the overall volume of the sound may appear to increase somewhat, and its tonal balance may be altered (depending on what sounds are being strongly imaged at the sides of the room).

7.5 LEARNING TO HEAR THE HOLOGRAPHIC IMAGE

Sonic holography frees the sound from the constraints of the speaker cabinets and from the confines of the listening room. It produces a palpably three-dimensional sound image that differs both in character and in size when compared to the conventional stereo image.

In breadth, for example, conventional stereo typically paints an image stretching across the wall from one speaker to the other — and no further. Sonic holography, even with unusually closely-spaced loudspeakers, can fill the entire 180-degree hemisphere in front of the listener with sound — with images localized not only between the speakers but beyond them, and occasionally even all the way around to the left or right of the listener.

Even more dramatic is the depth of the holographic image. Stereo tends to paint an image on the wall, or at best project an image slightly behind the wall, between the speakers. But the holographic image seems to energize the air. Depending on the miking used in the making of the recording, images may be localized in space well behind the speakers, while others are projected dramatically in front of the speakers. The instruments which provide live music with its power and impact, e.g., the string bass and drums, seem to fill the air around the listener with their energy. And the ambience of the recording site, which in conventional stereo playback may have been completely hidden in the sound, is heard in its true perspective above and around the performers — even when the Time Delay ambience circuits are switched off.

If these characteristics are not obvious to you when you first switch on the Sonic Hologram Generator, don't worry. Years of listening to the

perspective of conventional stereo often create a perceptual mind-set, causing the ear to filter out and reject impressions which contradict its expectations. Listen for a while to a variety of recordings with the hologram generator on, and let yourself gradually acclimate to the new perspective which it provides. It may take several hours of listening — either in one session or spread out over a few days, as you prefer. But after you have learned how to "hear" the holographic image, switching out the Sonic Hologram Generator will produce a bit of a shock as the three-dimensional sound field collapses suddenly into the space between the speakers and ordinary stereo is discovered to be flat, thin, and confined by comparison.

One of the difficulties in acclimating to the experience of holographic imaging is the sensory conflict between your eyes and ears. Your eyes are telling you that you are in your living room looking at a couple of speaker boxes which are the sources of sound, while your ears are trying to give you the message that you are in a large sound field with a breadth, depth, and ambience which could only exist in a much larger environment. So one way to accelerate the process of acclimatization is to switch off the lights and listen in the dark for a while; focus your attention on the sounds of the various instruments or voices and try to hear where they are localized, floating in space at some distance away. If you have a good live-concert recording, or any recording made in an acoustically "live" concert hall with a minimum number of microphones, listen for the sound of the space itself; the ambience fills the air with an almost palpable "feel" which disappears instantly when the Sonic Hologram Generator is switched off.

You should, of course, experiment with a variety of recordings while acclimating your ears to the effects of sonic holography. Not surprisingly, the holographic image will vary somewhat with the miking of the recording.

If continued listening with good recordings fails to develop a holographic image having dramatically greater width and depth than the conventional stereo image, check to see whether you have given sufficient weight to the set-up advice earlier in this chapter. For example:

(1) Are the two speakers positioned at precisely equal distances from your chair, as measured with an alignment string or tape measure?

(2) Are the speakers placed at least two feet away from all walls? Even if you are convinced that such placement is not practical for you, you still should try it for one day in order to assure yourself that the hologram generator is working correctly (and to learn what it should sound like).

(3) Have you devoted enough effort and attention to the control of early reflections, especially

those off the wall behind the speakers and the primary mirror-image reflections off the floor and side walls?

(4) Are the speakers wired in phase? (To check, switch off the hologram generator and the time-delay, and depress the STEREO/MONO button to produce a monophonic image, if the system is correctly phased you should hear a well-defined phantom mono image floating midway between the speakers.) If the system does not produce a well-defined mono image it cannot be expected to produce good images in either stereo or holographic operation.

7.6 EFFECTS OF SPEAKER DESIGN ON SONIC HOLOGRAPHY

Early reflections are the nemesis of the Sonic Hologram. The design of your loudspeakers can influence this problem in two ways.

(1) The wider the radiation angle (dispersion pattern) of the loudspeaker, the greater is the proportion of its output which is launched toward walls and other reflecting objects — rather than toward you. So if your loudspeakers have an omnidirectional or multidirectional radiation pattern (with drivers or reflecting panels aiming the sound to the sides, rear, or upward from the speaker as well as forward toward the listening area), good holographic imaging will be harder to achieve than with a simple front-firing design. This is not to say that it can't be done; but it means that with such speakers it is especially important that they be placed well away from all walls, and even with optimum placement it will still be necessary to apply some well-planned sound-absorbing treatment to those wall areas where the primary reflections occur.

If your speakers have a dipole radiation pattern (e.g., large electrostatic and planar magnetic designs), they radiate equally strongly to the front and rear. So there is a particularly strong primary reflection off the rear wall behind the speaker which must be dealt with. The first and best treatment is simply to move the speakers several feet forward into the room, so that their rear-wall reflection is delayed by several milliseconds. Sound-absorbing treatment of the wall behind the speakers is optional.

Even if your speakers are conventional front-firing designs but have better-than-average dispersion at midrange and treble frequencies, it is not unusual to find that a plainly audible improvement in holographic imaging is obtained by adding sound-absorbing panels to the side-wall areas where the primary mirror-image reflections occur on each side.

On the other hand, if your speakers employ horn midrange and tweeter elements, they may have a sufficiently narrow and well-controlled radiation pattern that they can even be installed against walls and near corners without incurring severe dilutions in imaging due to reflections.

(2) Sonic Holography employs signal delays of a fraction of a millisecond. In some speakers of older design, reflections with similar delays can occur on the speaker itself due to projecting edge moldings and other irregularities on the front surface of the cabinet. In most modern speakers such reflections (together with cabinet diffraction effects) are minimized by the use of sound-absorbing felt, rounded corners, smooth-fronted baffles, and unconventional cabinet shapes (slim columns, pyramids, or the use of large cabinets for woofers plus separate small cabinets for midrange/tweeter elements). If your speaker does not have visible evidence of design effort to minimize early reflections, this does not necessarily imply trouble; but if you are ambitious you might want to try installing on the front baffle board a 1/2 inch or 3/4 inch layer of sound-absorbing felt with holes cut to expose the drivers.

7.7 PATENT NOTICE

The circuitry and application of the Carver Sonic Hologram Generator are protected by patents. Purchase of the C-4000 gives you an implied license to use it to play recordings, but **not to make recordings.**

Of course, by connecting a tape recorder to the spare set of Main Output jacks it is possible to make hologram-encoded tapes which will exhibit holographic imaging in normal stereo playback. You may make a few recordings of this type, to demonstrate holography to friends or to experiment with the playback of hologram-encoded cassettes in a car. (Note: Of course any copying of recordings may be a violation of the copyright of the performing artist, producer, or manufacturer of the recording.)

You are **not** permitted to make hologram-encoded recordings for commercial sale. If you wish to do so, you must first obtain a license from, and arrange for payment of royalties to, Robert W. Carver, P.O. Box 664, Woodinville, Washington 98072.

8.0 TIME DELAY

Ambience Reproduction System

8.1 WHAT IT DOES

Live music is enhanced by the acoustic "ambience" of the environment in which it is heard. While the best two-channel stereo systems may provide an accurate reproduction of the sound engraved in the record grooves, they are limited in the realism of their result. They cannot re-create the sensation of being there in the same acoustic environment with the musicians, because that experience depends in part on being "enveloped" in reflected sounds from all directions.

The unique Carver Sonic Hologram Generator broadens and deepens the perspective of stereo recordings, realistically re-creating the frontal hemisphere (i.e.g. the half-circle in front of the listener) as it was "heard" by the recording microphones. The Time Delay circuitry in the C-4000 is intended to complete the circle, surrounding the listener with ambience — not the ambience of your listening room, but the ambience of the larger night club, concert hall or cathedral where the music was meant to be heard. The circuit works by presenting the ear with psycho-acoustic cues which it is sensitive to — cues which exist because of the relatively slow speed of sound in air.

In any acoustic environment, sound reaches a listener's ears via a direct straight-line path from the source and also via reflections off the walls and other surfaces. Since the reflected sounds travel a longer air-path than the direct sound, they take longer to arrive, and the delay time of reflected sounds is the major cue which the ear/brain subconsciously uses to judge the size of the environment. In a living room path lengths are only a few feet long, so reflection delays are short, while in a concert hall reflections may travel dozens of feet longer and so arrive dozens of milliseconds later than the direct sound.

The time-delay circuitry delays the audio signal by either 16 or 25 milliseconds (switch-selectable), by electronically storing it for that interval in a special integrated circuit. This circuit is a "charge-coupled device" (CCD) containing thousands of microscopic cells in which slices of the audio signal are stored as packets of electronic charge; the signal is sampled, stored, and shifted from cell to cell at a precisely determined rate until it emerges from the circuit after the desired delay. The Time Delay Echo Density control then feeds a selected fraction of the signal back to the input of the circuit to produce a repetitive delay, i.e., a string of many reflections back-and-forth between the walls of the electronic concert hall. The high-frequency response of the delayed signal is also contoured to simulate the high-frequency attenuation that occurs due to absorption at reflecting surfaces and in long-air-paths in a large acoustic space.

The delayed sound is then amplified by a pair of 20-watt power amplifiers included in the C-4000 and

is reproduced via two, three, or more small loudspeakers scattered around the walls of your listening room.

The time delay has two effects. (1) The presentation of the sound from the sides of the room after a slight delay reveals much of the original concert hall ambience which is hidden ("masked") by the direct sound in simple stereo playback; this effect was discovered by European acoustics researcher E.R. Madsen. (2) The delays directly simulate the reflections from the walls of an environment far larger than your listening room. As a result the time-delay circuitry, especially when used in conjunction with the Sonic Hologram Generator, produces the impression that your living room walls have vanished, yielding to a sense of space similar to the larger environment where the recording was made.

8.2 CHOOSING LOUDSPEAKERS FOR AMBIENCE

When you upgrade your stereo system it is tempting to buy a new pair of loudspeakers and simply relegate the older pair of speakers to the secondary time-delayed channels. This may work satisfactorily, but a better procedure is to use the old pair as extension speakers in a separate room or simply sell them, and to purchase new speakers for the time-delayed channels. There are two reasons for this preference.

(1) The walls of a concert hall have a major influence on the sound of a symphony orchestra, yet the walls are never audible as distinct sources of sound; their delayed reflected sounds simply blend in with the direct sound from the stage. Similarly the sound produced by the time-delay speakers in your listening room should blend unobtrusively with the direct sound from your main speakers. This result is most easily achieved if your secondary speakers have (as nearly as possible) the same tonal balance as your main speakers. A smooth midrange response, without major peaks and valleys, is particularly important; if the delayed-channel speakers have a peak at a particular frequency, musical notes at that frequency may appear to jump toward the delay speakers instead of having a stable location in the main stereo image.

(2) You probably will want to experiment with various locations for the delayed-channel speakers. Often the best placement turns out to be high on the walls, or lying on their backs on a shelf firing upwards. Such placements are easier to achieve with small speakers than with full-size models. You may also find that the best ambience reproduction is obtained with several channels of delayed sound instead of just two; and it is much more practical to scatter several small speakers around the room than to clutter it up with an equal number of full-sized speakers.

You may find that the manufacturer of your main stereo speakers also makes smaller models having a

similar sonic character, or you may select any other compact model having smooth response and a generally similar tonal balance.

When choosing speakers for time delay, the following characteristics are NOT important:

- Strong deep-bass output.
- High power-handling (the delayed channels are supposed to be significantly lower in volume than the main stereo speakers).
- Extended treble (the high frequencies are rolled off in the time-delay circuitry, as they are in the ambient sound field of a concert hall).

The following characteristics are desirable (together with smooth response and compact size) in speakers for time delay:

- Wide dispersion (the delayed sound should be scattered uniformly around the listening room, not beamed directly at the listener).
- High efficiency (to make the best use of the 20-watt time-delay amplifiers built into the C-4000).

"Mini" speakers, with 3-inch or 4-inch woofers in cabinets 0.5 cubic foot or less in volume, are an obvious and popular choice. But they typically have rather low efficiency and a response that rolls off rapidly below 100 Hz. A better choice would be one of the many speakers employing 5-inch or 6-inch woofers in 1 cubic foot cabinets for improved efficiency and useful bass output down to 60 Hz or better.

8.3 PSYCHOACOUSTICS AND SPEAKER PLACEMENT

Ambience in a concert hall or other large acoustic environment is omnidirectional and diffuse, arriving randomly at the ear from all angles. Similarly, in the home, the time-delay speakers are not intended to be heard as sources of sound. Their output should be scattered randomly about in the room so that it arrives at the ear after being reflected off the walls and ceiling. Here are some guidelines for placement and aiming:

(1) Psychoacoustic studies have shown that the reflected sounds that contribute most to the perceived ambience of a listening environment are those arriving at the listener from the ceiling and side walls. Surprisingly, sounds from the rear are less important. So the ambience speakers should be located on the side walls (roughly 90 degrees away from the main stereo speakers) or should be aimed to reflect much of their sound off the side walls and ceiling.

One practical arrangement is to mount the speakers on the rear wall (behind the listener), oriented back-to-back, i.e., firing toward the left and right so that most of their sound reflects off the side walls before reaching the listener. Or locate them near the rear corners of the room, but aimed diagonally outward toward the side walls rather than toward the center of the room. If the speakers are mounted on the side walls on shelves, they should be placed high (near the ceiling) or aimed upward, so as to avoid beaming their sound directly into your ears.

(2) Typically home environments are acoustically absorptive below waist-level, because of carpets, furniture, etc. So if a loudspeaker is placed on the floor or on a low stand, some of its output will be absorbed rather than being scattered uniformly about the room. Elevating the speaker above head level ensures that a greater proportion of its sound will be reflected off ceiling and wall surfaces a few times before reaching your ears. (Incidentally this may not be true in store showrooms, which often have false ceilings employing sound-absorbing acoustical tiles.)

In normally reflective home environments time-delay speakers often work best when placed high on the walls on shelves or hanging from picture hooks. If this is impractical, a sensible alternative is to mount them at a modest height, lying on their backs to fire much of their sound up toward the ceiling, or mount the speakers on tilted stands a foot or two away from the wall, aimed toward the wall and up so that their sound is bounced off the wall and ceiling.

(3) Place the ambience speakers as far away from your chair as possible. Simply moving them away increases the ratio of diffuse sound to direct sound. If your room is long and narrow, the preferred arrangement is to have your main stereo speakers located along the longer wall, firing across the shorter dimension of the room, with the time-delay speakers located at the far ends of the room. This is better than having the main speakers at one end of the room and the ambience speakers on the side walls only a few feet to the left and right from your listening chair.

8.4 USING MORE THAN TWO AMBIENCE SPEAKERS

With omnidirectional ambience speakers, and with highly reflective room surfaces, one could in principle create a truly diffuse, omnidirectional ambient sound field using just one pair of ambience speakers. But a room with a well-balanced mixture of absorption and reflection is more comfortable to live in (and better suited to reproduce music in) than a totally reflective environment. And few speakers are omnidirectional. So, while a pair of time-delayed speakers can go a long way toward establishing a satisfying illusion of concert-hall ambience, a still closer approach to realism can be obtained by using more than two ambience speakers. Here are two options worth considering.

(1) Depending on the quality (and quantity) of ambience in the recording, the use of a pair of side-located or side-firing ambience speakers may leave you with a feeling of an ambient "hole in the middle" — ambience all around you but missing from in front. To remedy this we suggest that you add an ambience speaker at the front of the room. It could be located between your main stereo speakers and aimed toward the listening area in the

conventional manner, or you may prefer to try aiming it away from you so that its sound scatters off the wall behind the stereo speakers, or aiming it upward to scatter off the ceiling and upper wall surfaces at the front of the room. Or, especially in a room with a high or "cathedral" peaked ceiling, you may mount the third ambience speaker on the ceiling itself. Another option that works surprisingly well is to place the third speaker under the couch or chair that comprises your main listening position.

Connections for this "center" ambience speaker are provided in the C-4000, so no extra amplifier is required.

(2) An additional pair of speakers may be added to the first time-delay pair at the sides or rear of the listening room. Spreading out the sources of time-delayed sound makes it more diffuse and permits each delayed speaker to be operated at a lower volume level, minimizing any chance of it being heard as a distinct sound source. For best results, all of the delay speakers should be of the same brand and model, in order that their efficiency and tonal quality be uniform.

In addition to its time-delay circuitry the C-4000 contains a pair of 20-watt amplifiers to drive the time-delay speakers. The amplifiers are designed to drive impedances as low as 4 ohms. So if you use one pair of time-delay speakers (i.e., one speaker per channel), they may have an impedance of either 4 or 8 ohms. If you use two pairs of speakers wired in parallel on each channel, they should be 8-ohm models. If you want to use two pairs of 4-ohm speakers in each channel, they should be wired in series. (If they are identical models, series wiring will not degrade their performance.)

The "center" time-delay speaker terminals are effectively in parallel with the side speakers, so if you use the suggested arrangement of three ambience speakers (one on each side plus one in front), you should consider the combined impedance of the front speaker together with the side speakers. If all three speakers are 8-ohm models, then the combined impedance of the front speaker together with either side speaker is 4 ohms, which is fine. But if all three are 4-ohm models, the front and left ambience speakers would present a combined impedance of only 2 ohms to the left-channel amplifier; this would probably cause the amplifier's current-limiting protection circuit to activate, preventing you from fully exploiting the amplifier's rated output. What about combining speakers with dissimilar impedances? If each side speaker is 8 ohms and the front/center speaker is rated at 4 ohms, the combined impedance is:

$$\frac{1}{\frac{1}{8} + \frac{1}{4}} = 2.7 \text{ ohms.}$$

This is marginal, but it will generally work OK as long as the speakers really do have their advertised impedances. (Some so-called 8-ohm speakers really are 4 ohms at the frequencies where the highest power demands in music occur, although their impedance may average 8 ohms across the entire audible frequency range.) Try it and see.

8.5 CONNECTIONS

To use the time delay, simply connect the ambience speakers to the TIME DELAY SPEAKERS terminals on the rear panel of the C-4000. Observe the marking of the terminals to ensure correct phasing of the ambience speakers.

Phasing of ambience speakers is not as important as correct phasing of your primary stereo speakers. But in case of doubt it is easily checked, as follows. Place the left and right ambience speakers face-to-face not more than a foot apart, depress the SPEAKERS OFF button to silence the main stereo speakers and let you hear the time-delay speakers alone, depress the STEREO/MONO button to blend the two channels into mono, depress the TIME DELAY button to switch on the delayed outputs, and play some bass-heavy music. If the sound of the ambience speakers is rich in bass output, the speakers are correctly phased. If the sound is thin, reverse the connections at the back of the right speaker only.

If you are using three ambience speakers as suggested, disconnect the center (front) speaker while checking the left/right phasing. After that phasing test is completed, mark the right-channel ambience speaker's wires and disconnect them while you check the front speaker's phasing by placing it face-to-face with the left ambience speaker. Again, if the sound is thin instead of rich in bass, reverse the wiring at the back of the center speaker only. Finally reconnect the right ambience speaker as marked.

In most systems the built-in time-delay power amplifiers will be completely satisfactory. But if you are using very inefficient ambience speakers, or like to use lots of time-delay level, or want to use two or more pairs of 4-ohm ambience speakers, then you may find it preferable to use an external power amplifier to drive the ambience speakers. Simply connect a stereo patch cord from the TIME DELAY LOW LEVEL OUTPUT jacks to the power amplifier's input jacks, and connect the ambience speakers to the external power amplifier's speaker terminals instead of those on the C-4000.

8.6 THE TIME DELAY CONTROLS

Operation of the time delay is governed by four controls. The TIME DELAY switch engages the delay circuits when depressed and disengages them when in the "out" position. The 16/25 ms switch* selects the length of the primary delay. The TIME DELAY ECHO DENSITY knob adjusts the

recycling of signals to create multiple delays simulating reverberation. And the TIME DELAY OUTPUT LEVEL knob adjusts the volume of the time-delayed outputs (relative to the volume of your front stereo speakers).

*In C-4000 units manufactured before 1982 this switch is labeled 35/50 ms. But the delay times were shortened to 16 and 25 ms early in 1981 production for better sonic performance.

To activate the delay, first set the TIME DELAY OUTPUT LEVEL and ECHO DENSITY controls to minimum (fully counterclockwise) to avoid any unpleasant surprises; then depress the TIME DELAY switch, and set the 16/25 ms switch to its "in" (25 ms) position. If you want to hear only the delayed sound depress the SPEAKERS OFF switch to silence the main stereo speakers. Turn up the TIME DELAY OUTPUT LEVEL control to hear the delayed sound. Then, as you gradually turn up the ECHO DENSITY control you can hear the sound begin to take on a hollow quality as successive delays overlap. (This is especially obvious with speaking voices, such as radio announcers.) If you turn up the ECHO DENSITY too far the recirculation circuit will oscillate, howling at specific frequencies. In general you will find that low-to-moderate settings of the ECHO DENSITY control (below about 12 o'clock) sound best with most music.

To evaluate the effect of the time delay properly, it must be heard together with the direct sound from the main stereo speakers. Release the SPEAKERS OFF button, turn down the TIME DELAY OUTPUT LEVEL, adjust the master VOLUME control for a comfortable loudness level, and turn the TIME DELAY OUTPUT LEVEL back up until you can just hear the contribution being made by the ambience speakers to the total sound.

Then turn the TIME DELAY OUTPUT LEVEL back down just slightly so that you are no longer explicitly conscious of the sound coming from the ambience speakers. Thus adjusted, the ambience speakers will contribute a realistic sense of spaciousness to the sound but — like the walls of a concert hall — they will not call attention to themselves as discrete sources of sound. To test the effect of their contribution, use the TIME DELAY pushbutton to switch the ambience speakers off and on.

Ideally, switching on the time delay should produce a relatively undramatic, natural sense of spatial freedom as the sound is heard in its natural environment rather than in the spatially constricted confines of a small living room. Switching off the time delay should produce a rather more obvious ef-

fect, with a noticeable feeling of loss as the three-dimensional sound field collapses into the front wall between the main stereo speakers.

If you are not accustomed to using a time-delay system you may be tempted to operate the ambience speakers at exaggerated levels at first in order to hear them produce an obvious expansion of the perceived listening space. As you grow accustomed to it, you will find that more subtle ambience levels are satisfying — and more natural. If the TIME DELAY OUTPUT LEVEL is set too high you not only run the risk of stereo images being pulled toward the ambience speakers, but you also may hear noise or distortion in the ambience channels. Today's time-delay circuits are not yet capable of the ultra-low noise and distortion of amplifying circuits. The dynamic range of the delay circuits in the C-4000 is ample for virtually any normal requirement, so audible noise from the ambience speakers is usually indicative of non-optimum control settings.

If you hear distortion in the ambience channels when playing music loud, you could be overdriving either the delay circuits or (more likely) the built-in 20-watt power amplifiers. To find out which, turn down the TIME DELAY OUTPUT LEVEL slightly. This control is at the output of the delay circuits and governs the signal level fed to the power amplifiers; if turning it down slightly eliminates the distortion, then it was the amplifiers which were being overdriven.

It will probably take a period of experimenting for you to discover the control settings which allow you to best exploit the dynamic range of the time delay circuits and associated power amplifiers, and to discover the settings of the delay controls which sound best with various kinds of music. The 16 ms delay with a low setting of ECHO DENSITY provides the most subtle and natural effect for many recordings, while the 25 ms delay and medium-to-high settings of ECHO DENSITY provide the option of more dramatic expansion for special spatial effects, at the risk of some twangy coloration of the sound.

As noted earlier, the VOLUME control raises and lowers the overall sound level in both the main and ambience channels. The TIME DELAY OUTPUT LEVEL then adjusts the relative level of the ambience outputs, thus fixing the ratio of ambient to direct sound. The level of the delayed sound is not affected by the SPEAKERS OFF button; nor is the delayed sound affected by the BASS and TREBLE tone controls.

9.0 APPENDIX A: THE THEORY OF SONIC HOLOGRAPHY

9.1 THE ILLUSION OF STEREO PERSPECTIVE

Today's microphones, recording systems, amplifiers, and loudspeakers are, at their best, very good indeed. The sounds of singing voices and musical instruments can be reproduced with a remarkably high degree of accuracy, faithfulness, and freedom from falsifying distortions or aberrations. But this is true only in terms of re-creating the direct sounds of individual voices and instruments, recorded with closely-placed microphones. When an attempt is made to reproduce a complex sound field (e.g., the total sound of an ensemble of voices or instruments in a three-dimensional acoustic environment such as a night club, church, or concert hall), then the impression of realism starts fraying at the edges.

Stereo sound is an illusion, and for some listeners it is not a particularly successful or convincing one. Stereo reproduction is subject to fundamental distortions of spatial perspective, sufficiently severe that no six-year old with normal hearing will be fooled into confusing a stereo playback with a real, live sonic event. The imaging of stereo is an acquired taste which audiophiles learn to be sensitive to — acclimating to its unnatural perspective in order to enjoy the portrait of sound which the stereo system paints upon the wall between the loudspeakers.

Consider, by analogy, the illusion of depth perspective that is provided in photographs and paintings by converging straight lines and the hazy reduction of contrast in "distant" objects. The geometry of perspective is part of the perceived real world, and rendering it is an essential requirement for any landscape painter. Certainly the historic discovery of optical perspective a few hundred years ago resulted in paintings that are generally more pleasing to view than, for instance, the flat two-dimensional figures in Egyptian paintings from the tombs of the Pharaohs. Still, few people viewing paintings (or even photographs) have ever been fooled into believing they were looking through a window at a real three-dimensional scene. And while stereo sound is both more realistic and more pleasing than monophonic reproduction, it is still only an attractive illusion.

Many listeners don't care about its limitations. For most people, a stereo system is a pleasant vehicle for listening to recorded music, and — like a Renaissance painting — it is not judged on its ability to deceive our senses. Most listeners accept the stereo illusion on its own terms, imperfect as it is, and don't let it stand in the way of enjoying the

rest of the sonic experience: the rhythms, melodies, harmonies, orchestration, song lyrics, the power of pipe organ pedals and the crisp impact of percussion.

But some of us who are audiophiles want more. For decades "high fidelity" has been billed as providing either they-are-here or you-are-there realism, and this is what we look for. Historically the strongest push for the continued improvement of recording and playback systems has come from the desire to recapture the elusive sense of "being there" — in the night club with the jazz trio, in the concert hall with the symphony orchestra, in the cathedral with the choir. Realism is the criterion. And by this criterion stereo sound is flawed.

It may not be obvious why this is so. It is widely assumed that two-channel stereo is theoretically "correct," because we have two ears: if we could eliminate all distortions, the two information channels in a recording should be sufficient for a convincing reproduction of a sonic experience, since the listener has only two aural channels through which all sounds are processed. But the ears are not just two simple information channels for auditory signals.

The two ears are mounted on one head, a semi-dense object whose dimensions are comparable to the wavelengths of mid-frequency sounds. And the ears are not simply holes leading to microphone-like sound detectors: each eardrum and ear canal is accompanied by the pinna, that convoluted flap of skin and cartilage which protrudes from each side of the head and, incidentally, serves as a support for eyeglasses and earrings. The pinnae produce complex patterns of reflections and frequency-response alterations that vary with the direction of arrival of each sound wave. Finally, the connections from the ears to the brain involve a great deal of complex signal processing that has been programmed by a lifetime of experience and by millennia of evolutionary adaptation.

Now, the problem with stereo is simple: both ears hear both speakers.

To see why this is important, consider the process of recording and reproducing a sound — one musical note played by one instrument, located several feet to the left of the center of the stage. What would you hear as a listener if you were located in an ideal front-and-center seat? The sound spreads out in all directions at a speed of approximately 1100 feet per second. If you are facing the center of the stage, the sound arrives at your left ear first and at your right ear very shortly afterward — how long afterward depends on its angle of arrival.

If the sound source is exactly in front of you, identical signals arrive at both ears at the same time. If the signal source were located directly to your left, 90 degrees away from the frontal direction, it would arrive first at your left ear, in unaltered form; in order to get to your right ear it has to travel the additional distance between your ears and so is delayed by about 0.5 millisecond. And since your head blocks high frequencies but isn't large enough to be an effective barrier for lows, your right ear receives a filtered version of the sound. Since the instrument in our example is only a few feet left of stage center and so is only slightly to the left of front (rather than 90 degrees around to the left), the arrival of the sound at your right ear is delayed by only a small fraction of a millisecond and its frequency spectrum is only modestly filtered by the acoustic shadow of your head.

9.2 STEREO RECORDING AND PLAYBACK

If the sound is recorded and later played back via loudspeakers, the result will depend on the microphone technique employed. Consider the simplest and most common: the sound is recorded via a single close-up microphone whose signal is "panpotted," i.e., split and recorded in both stereo channels but slightly stronger in the left channel in order to place its image slightly to the left of center. In playback the sound emerges simultaneously from both speakers (a little louder in the left).

Assume that you are sitting equally distant from the two speakers, facing the midpoint between them. The sound from the left speaker arrives at your left ear, and at the same time the sound from the right speaker arrives at your right ear. A fraction of a millisecond later the sound from the left speaker, after filtering by the acoustic shadow of your head, arrives at your right ear; and similarly the sound from the right speaker arrives at your left ear.

In the "live" listening experience the single sonic event produces two arrivals at the ear. The delay and frequency-spectrum differences between the arrivals at the two ears are the primary cues which the brain uses to determine the direction of the sound source. In the "panpotted" stereo recording and playback, the sonic event has produced a total of four arrivals at the ears, the first two being simultaneous and identical in frequency spectrum — a very different set of cues.

In an effort at greater realism, some recording engineers attempt to record the musical performance with a "coincident pair" of crossed cardioid or figure-8 microphones. The sound from the instrument, regardless of where it is located on the

stage, arrives simultaneously at the two mikes and is recorded in both channels, with a difference in intensity which is proportional to the source's angular displacement away from stage center. Thus in playback the sound emerges simultaneously from both loudspeakers, with some difference in strength; but just as with panpotting, the original sonic event generates a total of four sound arrivals at the ears.

The other common technique for recording large ensembles such as symphony orchestras and choruses is to hang two microphones in front of the stage, separated by about eight feet. Now, if the instrument is located several feet left of stage center, its sound will reach the left microphone first and will get to the right microphone after an extra air-path delay of, say, three milliseconds. As with the previous examples the sound of each instrument is present in both channels of the recording, but in this case with a time-delay as well as an intensity difference between the channels.

In playback the sound emerges from the left speaker, is heard by your left ear, and arrives at your right ear with some head-shadow loss after a small fraction of a millisecond. Meanwhile, three milliseconds after its appearance in the left speaker, the sound emerges from the right speaker and arrives successively at the right and left ears in turn. Not only do four sonic arrivals at the ears arise from the single sonic event — this time they are spread out by several milliseconds in time because of the spacing of the recording microphones. (In life a single event cannot generate arrivals at the ears spaced more than one millisecond apart, since no one's ears are spaced more than a foot apart.)

There are additional stereo miking techniques in common use, but all share the characteristic that every sound is present to some degree in both channels. Therefore every sonic event always produces four sonic arrivals at the ears in stereo playback — instead of the two which in life provide the brain's primary cue for localizing the direction of a sound. Of course this problem is avoided in a "ping-pong" recording, in which the sound emerges only from the left speaker or only from the right; but that's not stereo and cannot present a panoramic image spanning the space between the speakers.

One successful approach to lifelike sound reproduction is binaural recording, using microphones buried in a dummy head so that the recorded signals already contain the inter-aural delays and head-shadow losses which a live listener would experience. The recording is played through headphones, so that each ear hears only what the

same-side microphone in the dummy head picked up. This method is not without technical flaws, but its most important limitation is economic: most listeners don't like to be confined to headphone listening, so binaural recordings have very limited sales potential. As a practical matter most recordings must be engineered for loudspeaker playback.

Therefore we need a direct solution to the problem of acoustic crosstalk in the listening environment. This is the goal of the Carver Sonic Hologram Generator.

9.3 SONIC HOLOGRAPHY: CANCELLING ACOUSTIC CROSSTALK

As noted earlier, the essence of the problem is that both ears hear the sounds from both loudspeakers. The sound from each speaker reaches the same-side ear directly and then, after a brief delay and some loss due to the acoustic blockage of the listener's head, reaches the opposite-side ear. Conceptually, the object of sonic holography is simply to cancel out that delayed, attenuated signal reaching the opposite-side ear, so that each ear will be exposed mainly to just the signal from the speaker on the same side.

In principle it is quite straightforward. We know that the signal from the left-channel speaker arrives first at the left ear, then arrives in slightly weaker form at the right ear after an added delay of about 0.2 millisecond with its highs rolled off. All we have to do is feed to the right speaker a sample of the left-channel sound that is delayed by 0.2 ms and rolled off in highs. This signal from the right speaker will get to the right ear simultaneously with the arrival of the unwanted acoustic crosstalk signal from the left speaker. So if we phase-invert our specially-delayed right-speaker sample of the left-channel signal, this electrical crosstalk will cancel the acoustic crosstalk as the two signals arrive at the right ear. A complementary process is used to cancel the acoustic crosstalk from the right speaker into the left ear.

The actual operation of the Carver Sonic Hologram Generator circuit is rather more complex than this, of course, but that is the basic operating principle.

The result is an unparalleled ability to hear the true depth and breadth of the stereo image contained in the recording. For instance, when instruments or voices are recorded in stereo, each direct sound reaching the mikes from the sound source is accompanied by slightly delayed reflections from nearby surfaces such as the floor and walls of the stage. In live listening these "early" delays, while not resolved as distinct sounds, contribute to the ear's perception of the distance of the sound source and the character of its acoustic environment. In stereo playback the oppo-

site-ear early delays produced by acoustic crosstalk confuse the ear's perception of these early delays in the recording. By cancelling acoustic crosstalk sonic holography restores perception of differences in depth and ambience in the stereo image which are "masked" in ordinary stereo playback.

Why is this process called sonic holography? An optical hologram is a photograph made with a laser whose coherent beam of light is split into two beams and used to illuminate an object; the two beams are recombined, forming alternating rings of constructive and destructive interference, and the interference pattern is photographed. When the picture is developed and another laser is used to project it, a three-dimensional image of the photographed object is projected in space.

By analogy, a sonic hologram generator takes the beam of sound produced by each loudspeaker and splits it so that a related beam of sound is produced by the opposite speaker (after delay and filtering) in such a way that acoustic interference of the sounds occurs in the air near each ear, revealing the true three-dimensional sound image that was hidden in the stereo recording. Recall that "stereo," in the original Greek, means "solid" or three-dimensional, not just wide. Ideally stereo is intended not only to paint a sonic image onto the wall between the loudspeakers but to yield realistic perception of depth as well.

10.0 APPENDIX B: IN CASE OF DIFFICULTY

10.1 HUM

Under normal operating conditions you will not hear any hum originating in the circuitry of the C-4000. There is one exception to this rule: if you have a high-gain power amplifier and unusually sensitive (i.e., efficient) loudspeakers, normal listening levels will involve using abnormally low output levels from the preamp, and those small signals might then pick a bit of hum or hiss in the preamp's circuits. Such situations usually are characterized by a need to use only low settings of the Volume control, with settings above 12 o'clock yielding excessively loud sound. In this case the solution is to turn down the power amplifier's input level controls about halfway, so that you can turn up the preamplifier's signal levels to normal. If your power amplifier lacks input level controls, an alternative procedure is to depress the OUTPUT -15 dB pushbutton, which attenuates the preamp output and enables the use of higher Volume control settings for better signal-to-noise ratio in the circuitry.

Except for the condition described above, audible hum will nearly always be found to be due to problems external to the C-4000 — usually in the

signal source, i.e., the turntable or tape deck. Many turntables, for example, have a hum field in the vicinity of the platter (due to the turntable's motor or internal power transformer) which is acceptably low with moving-magnet cartridges but audibly bothersome with moving-coil cartridges. You should experiment with reversing the AC power cord of the turntable (preferably with the motor running and the cartridge suspended midway over the platter, held up via the cueing control) to see which orientation of the plug minimizes the audible hum. The hum may also vary with the location and orientation of the turntable with respect to AC wiring in the walls, making it necessary to move the turntable to another part of the room. Both turntables and tape decks, of course, are sensitive to the external hum fields created by many power amplifiers, and sometimes to the hum fields of other household appliances (such as a refrigerator on the other side of the wall). It is important that signal cables in general, and the turntable signal leads in particular, should not run close to and parallel with AC power cords, nor close to a power transformer or motor (including that in the base of the turntable).

In some cases hum may be minimized by connecting a heavy stranded wire from the preamp's Ground post to a true earth ground – which may turn out to be any, all, or none of the following: the third (round) hole in an electrical wall socket in modern U.S. homes, a steam radiator, or a cold water pipe. However, if your power amplifier employs a three-wire power cord, the stereo system may already be grounded through that, in which case another grounded wire from the preamp will create a "ground loop" and make the hum worse. As for the various components within the stereo system, they are mutually grounded via the shields of the signal cables and should not be interconnected with additional grounding wires, for the same reason (except, of course, the turntable, whose grounding wire usually – but not always – should be connected to the preamp Ground post).

Finally we come to the other common source of hum problems, the signal cables and their associated plugs. Inexpensive cables often have mediocre wrapped shielding, so better cables may make an audible difference. And it is important that the plug at each end of every cable makes a good, tight fit in its mating socket. (In this, don't neglect the "source" end of the turntable signal cables, which in many tables are plugged into sockets underneath rather than being soldered to terminals within the turntable's base.) Crimp the leaves of the phono plug's skirt slightly inward, if necessary, to ensure that it has a tight friction fit on the exterior of the phono jack. When plugging in each cable, use a

rotary twisting motion as the plug goes into the jack, in order to scrape away any invisible surface corrosion and make a clean metal-to-metal contact. Finally, in many inexpensive molded cables, the wire breaks where it makes contact with the plug; this problem can be identified by wiggling the cable and listening for an intermittent signal connection or intermittent hum.

10.2 RADIO-FREQUENCY INTERFERENCE

Radio Frequency Interference from CB, TV, AM, and other radio transmitters is a common problem, and like hum it usually can be traced to a condition external to the C-4000. If you encounter RFI the first step is to switch off the Time Delay and depress the SPEAKERS OFF button to mute the preamp's output. If you still hear the interference it is being detected and amplified in the power amplifier. Sometimes RFI gets into the power amp via the signal cables running from the preamp, and may be cured by substituting cables with better braided or foil shielding. More commonly RFI enters the power amp through its output terminals, with the speaker wires acting as receiving antennas. In this case it might be cured by connecting a 0.01 to 0.1 microfarad disc capacitor across the speaker terminals in each channel, but be sure to check with the amplifier manufacturer first: some power amps become unstable and burn out when certain values of capacitance are connected at their output terminals. A simpler cure is to place the power amp near the speakers and use short wires; then use extra-long well-shielded signal cables from the preamp to the power amp, which shouldn't cause any problems.

If the interference disappears when you mute the output of the preamp, then the interference is part of the signal and probably is entering the preamp from one or more of your signal sources. Use the Input Selector and Tape Monitor controls to identify which signal sources are picking up the interference. Usually turntables and tape decks are most vulnerable to RFI. If the RFI is coming in through the phono signal leads, cables with better shielding might help. Other options include wrapping the signal cables with aluminum foil which is then connected to the Ground post; or forming a loop in the cables, adjusting the size of the loop to tune out the interference, and taping it in place. And as with hum, try tightening all phono plugs and twist them in their sockets to get good metal-to-metal contact.

RFI in tape decks may enter via signal cables, but more commonly the interference is picked up directly in the playback head and its associated internal wiring, so a cure is likely to involve a trip to the factory or service shop for approved modifications. Or you might be able to reduce the

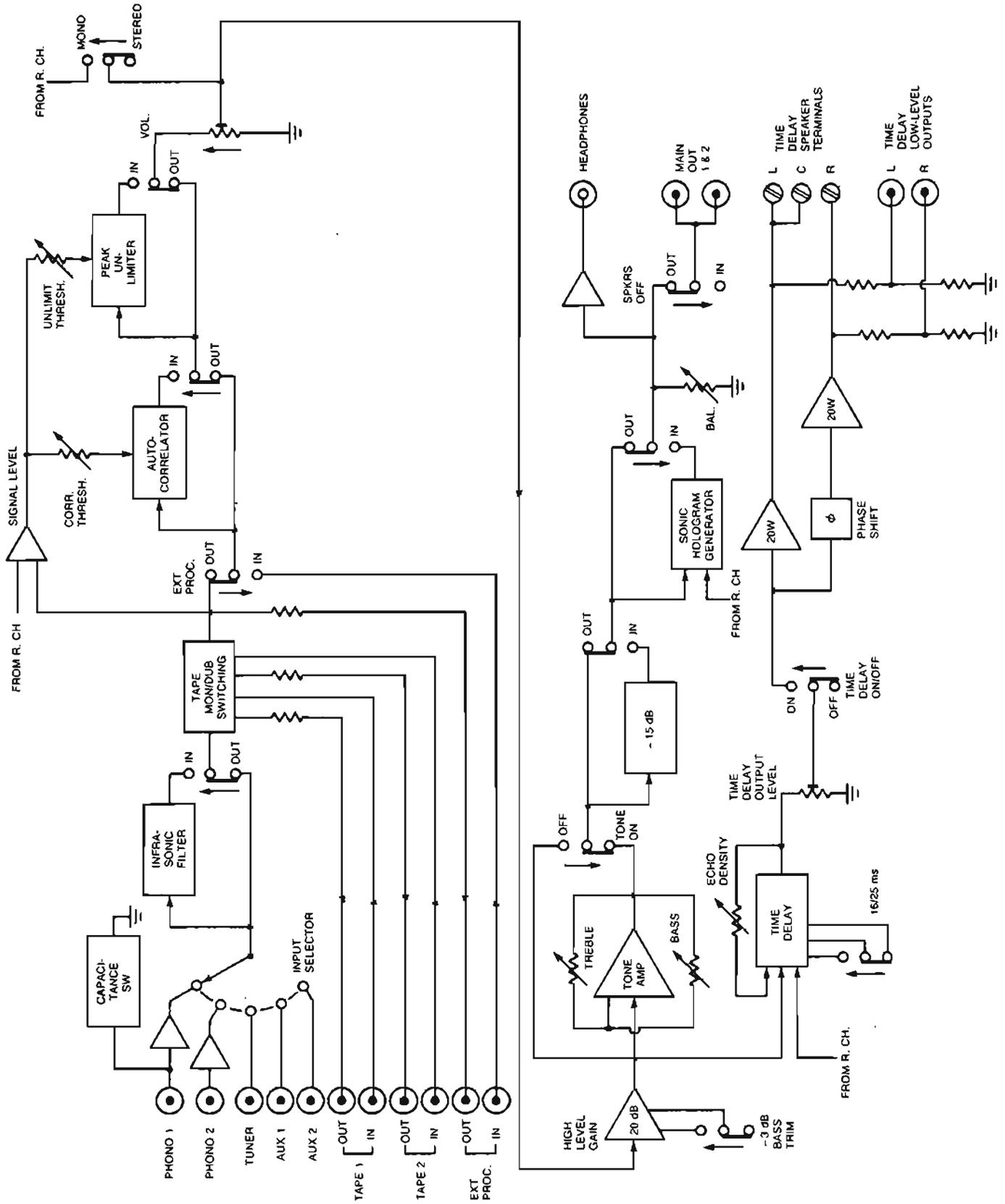


Figure 10-1. BLOCK DIAGRAM C-4000 LEFT CHANNEL

interference to tolerable levels simply by turning the tape deck 90 degrees or moving it to another location in the room.

10.3 PROBLEM SOLVING: DIAGNOSTIC HINTS

In view of the C-4000's input/output flexibility, the variety of its signal-processing functions, the many connecting cables to and from the components which may be connected to it, and the large number of possibilities for mis-set switches both on the C-4000 and on the various ancillary components operating with it, obviously it is impossible to offer a complete troubleshooting guide to all of the problems which could, in principle, occur. Most such potential problems will be avoided simply by following the instructions in this owner's manual and the instructions supplied with associated products; and many other possible problems will be prevented simply by the excellent reliability of modern solid-state components.

So in this section we will provide a guide to some of the most basic and common difficulties which may arise from time to time, and suggestions as to their probable cause. To illustrate the sort of thought process which is useful in tracking down problems, we begin with the most basic: no sound because the preamp's power is off. Did you accidentally hit the Power button when reaching for the Time Delay button? Was the preamp's AC line cord accidentally pulled partially out of its wall socket during housecleaning earlier in the day? Has the preamp's fuse blown? Did something else on that same household branch circuit (including the power amp or other component plugged into the preamp's AC convenience outlets) cause a current surge which blew the fuse or circuit breaker protecting that entire branch circuit? In some houses having duplex AC wall sockets, the lower one is permanently live while the upper one (intended for lamps) is controlled by a wall switch near a doorway; was the preamp's AC cord accidentally plugged into the upper socket? Is the preamp's AC cord plugged into a clock timer which is presently off or unplugged from the wall?

Incidentally, how can you tell when the preamp's power is on? The auto-correlator is equipped with two red LEDs which also serve as pilot lights: when the CORRELATION — AUTOMATIC THRESHOLD pushbutton is depressed, the adjacent LED is illuminated. When that button is out, the LED adjacent to the MANUAL correlation threshold knob is illuminated. If **neither** LED is glowing, either the preamp's power is off or an LED has failed.

PROBLEM: NO SOUND

SUGGESTED CAUSES

1. Preamp power off, power amp off.
2. Line cord unplugged (preamp or power amp).
3. Fuse blown (preamp or power amp).
4. Power off at wall socket (check with lamp).

PROBLEM: NO SOUND (POWER ON).

SUGGESTED CAUSES

1. Input Selector set to inactive input.
2. Either Tape Monitor button depressed with no tape machine running.
3. SPEAKERS OFF button depressed.
4. External Processor button depressed with no processor connected, or with a signal processor connected but not operating.
5. Input level controls turned down on power amplifier.
6. Input or output signal cables disconnected.
7. Selected program source not operating.
8. Output level control turned down at program source.
9. Program source misadjusted (for example, FM tuner tuned between stations with Muting circuit engaged).

PROBLEM: NO SOUND IN ONE CHANNEL

SUGGESTED CAUSES

1. Defective cable from preamp to power amp or from program source to preamp.
2. Speaker wire loose or disconnected.
3. Balance control fully clockwise or counter-clockwise.
4. Imperfect contact in switch (especially any level or slide switch in a program source or signal processor, as well as the various signal-routing switches in the preamp).

PROBLEM: LOUD HOWL, SQUEAL, OR WHISTLE

SUGGESTED CAUSES

1. Both DUB switches depressed at the same time, with recorders in recording or source monitor mode.
2. Tape Monitor engaged while microphones (in the same room as the speakers) are connected to tape deck for recording.
3. Time Delay Echo Density knob turned up too far.

PROBLEM: SOLO VOICES OR INSTRUMENTS SOUND THIN, SHRILL, OR DISTORTED

SUGGESTED CAUSES

1. Treble controls set to maximum boost.
2. Phono cartridge wired out of phase.
3. Speakers wired out of phase.

PROBLEM: ASYMMETRIC HOLOGRAPHIC PROJECTION (IMAGE STRETCHES BEYOND SPEAKERS ON ONLY ONE SIDE, OR WRAPS AROUND FURTHER ON ONE SIDE THAN THE OTHER).

SUGGESTED CAUSES

1. Asymmetric arrangement of the speakers in the listening room. Center the speakers.
2. Reflective surface near opposite-side speaker. (i.e. if right-side holographic image is weak, move the left speaker away from the wall.)

PROBLEM: HOLOGRAM DOES NOT PRODUCE A WELL-DEFINED OPTIMUM CENTER POINT FOR LISTENING. (LEFT-RIGHT HEAD MOVEMENT DURING SET-UP RESULTS IN SEVERAL PREFERRED AXES.)

SUGGESTED CAUSES

1. One speaker farther away from listener than the other. Use tape measure to establish precisely equal distances from chair.
2. Listening room too "live", especially in the area around the speakers.
3. Loudspeakers:
 - (a). are too close to reflective wall surfaces.
 - (b). are too far apart (the listening angle is too wide).
 - (c). have excessively wide dispersion.
 - (d). have front-panel diffraction problems.

If these suggestions do not help, or the problem you encounter falls outside the scope of this manual, please write:*

Carver Corporation Technical Services
P.O. Box 664
Woodinville, WA 98072

or call (206) 483-1202. Your inquiry will be promptly answered. You may be directed to a Carver-authorized service center, or asked to return the unit to the factory. We must have the serial number of your unit before we can authorize its return. Your dealer, if convenient, may also offer assistance and may be consulted.

*Applies to U.S. customers only.

10.4 SPECIFICATIONS

PHONO INPUTS

Frequency Response	Within ± 0.25 dB of extended RIAA curve.
Gain at 1 kHz (phono stage only)	Phono 1 (moving magnet): 35 dB Phono 2 (moving coil): 60 dB
IHF Input Sensitivity (re 0.5V out)	Phono 1: 0.9 mV Phono 2: 0.05 mV
Input Impedance	Phono 1: R = 47K ohms C = 0, 180, or 390 pF Phono 2: R = 39 ohms
S/N Ratio, IHF A	Phono 1: 81 dB re 5 mV Phono 2: 75 dB re 0.5 mV
Overload (Phono 1)	100 mV at 1 kHz 1V at 20 kHz

HIGH-LEVEL INPUTS

Frequency Response	20 Hz to 20 kHz +0, -0.25 dB 1.5 Hz to 60 kHz +0, -3 dB
Gain	20 dB
IHF Input Sensitivity	50 mV in for 0.5V out
Input Impedance	47K ohms
Output Impedance	Main Outputs: 910 ohms Tape and External Processor Outputs: 2K ohms
S/N Ratio, IHF A	Hologram out: 86 dB re 0.5V 98 dB re 2V Hologram in: 82 dB re 0.5V 94 dB re 2V
Distortion	THD, SMPTE 0.05% or less; IM, IHF IM typically 0.003% T.I.M.: unmeasurable
Infrasonic Filter	-3 dB at 15 Hz, 12 dB/octave
Autocorrelator (Patented)	Noise reduction approximately 3 dB at 1.5 kHz, increasing to 8 dB from 2.5 kHz to 20 kHz. Low-frequency noise reduction -3 dB at 200 Hz, increasing to 10 dB from 100 Hz to 20 Hz
Peak Unlimiter	Dynamic range expansion 5.5 dB
Time Delay	Delay: 16 or 25 ms (35 or 50 ms in early units). Recirculation: variable from 0 to 100% Bandwidth: 8 kHz Distortion: 0.25% or less S/N Ratio, IHF A: 85 dB re 1 volt, measured at Low Level outputs Power Output: 20 watts/channel from 20 Hz to 12 kHz at no more than 0.25% THD Outputs: Left back, Right back, Center; outputs are amplitude and phase randomized
Dimensions	19 x 6 $\frac{3}{4}$ x 8 $\frac{1}{2}$ "
Weight	11 lbs. (5 kg)

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P.O. Box 664 • 14304 N.E. 193rd Place, Woodinville, Washington 98072 Tel. (206) 483-1202