CARVERaudio.com

CARVER

MODEL C-1

SONIC HOLOGRAPHY PREAMPLIFIER OWNER'S MANUAL

NOTICE

IT IS O.K. TO BEGIN USING YOUR NEW C-1 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 on 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-1 are in fact not conventional at all.

We also recommend that you record the following information here, for possible future reference. You should also save your purchase invoice, in case service is required.

C-1 Serial No.:________________________

Purchase Date:________________________

Store Name and Address:____________________

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Please take a minute to fill out and mail the Carver Warranty Registration card.

WARNINGS

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-1'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-1. There are no user-serviceable parts inside. Refer all servicing to qualified personnel; unauthorized servicing may void the warranty.

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INTRODUCTION

Congratulations on your purchase of the Carver C-1 Sonic Holograhy Preamplifier. We believe its sophisticated engineering and meticulous craftsmanship will provide you with many years of listening enjoyment.

At the heart of the C-1 is a high-performance console preamplifier with precise RIAA phono equalization for moving-magnet and moving coil cartrides, extremely low noise and distortion, high slew rate, wide dynamic range, and a high degree of control flexibility. The built-in Sonic Hologram generator distinguishes the C-1 from all other fine preamplifiers, breaking through the limitations of listening-room acoustics and the conventional two-channel stereo medium itself to provide remarkable three-dimensional you-are-there realism, with a sonic image which extends far beyond the loudspeakers in breadth and depth.

UNPACKING

Remove the C-1 from its packing materials with care so as to avoid damage to or loss of any packing materials. The C-1 is a precision instrument and deservices to be treated with care. It has been designed to provide years of reliable service. Should you ever need to ship or transport your C-1, the original packing materials will provide the safest means.

INSTALLATION

The C-1 is an all solid state design. It may be operated at any angle, and has no special ventilation requirements.

The C-1's location relative to other stereo components is not critical, except that it should not be stacked with or placed adjacent to some power amplifiers which generate external hum fields.

Typically it is best to locate the C-1 close to the turntable so that connection cables may be kept short. The turntable should be on a solid, vibration-free surface to avoid problems of acoustic feedback and instability. In many systems it may be convenient to locate the preamp within arm's length of the prime listening position, for which extra-long cables will be necessary. See the appropriate appendix to this manual for further details on lead capacitance.

Although all C-1 instruments are EIA standard 19 inch wide and have handles, only charcoal units are intended for rack mounting. Conversion of champaign-gold to charcoal is possible (inquire to Carver Corp. Technical Services) at a modest cost. The decorative handles should not be removed, as the screws holding them in place also fasten the front panel to the chassis.

PREAMPLIFIER / CONTROL FUNCTIONS

The C-1 is a preamplifier control center. Familiarity with its preamplifier functions should be gained before proceeding to a setup of the Sonic Hologram. To expedite this process, we have outlined this section to provide the owner with quick information on all preamplifier functions, followed up with more detailed descriptions of specific controls and uses.

<u>Front - Panel Controls:</u>

POWER - Switches on C-1 and Switched AC Outlets.

VOLUME - Adjusts listening level at Main Output.

BALANCE - Adjusts relative levels of stereo channels.

SELECTOR - Selects Phono, Tuner, or Auxiliary inputs.

STEREO/MONO - Parallels (L+R) two channels when engaged.

SPEAKERS OFF - Mutes Main Outputs when depressed.

L BASS

L TREBLE - Tone contour controls.

R BASS

R TREBLE

TONE/PUSH ON - Engages tone contour controls.

40 Hz/LOUDNESS - Bass turnover, increases effect and range of Bass controls when engaged.

8K Hz/2K Hz - Treble turnover, increases effect and range of Treble controls when engaged.

EXTERNAL PROCESSOR - Switches an outboard signal processor or equalizer into circuit when engaged (See Rear Panel Connections).

DUB 2 1 - Allows tape copying from Tape 2 to Tape 1 when engaged.

DUB 1 2 - Allows tape copying from Tape 1 to Tape 2 when engaged.

TAPE 2 MONITOR - Allows playback or record-monitoring of a tape deck connected to the Tape 2 inputs and outputs.

TAPE 1 MONITOR - Allows playback or record-monitoring of a tape deck connected to the Tape 1 inputs and outputs.

Rear Panel Connections: Inputs

PHONO 1 - For connection of the output from a moving-magnet phono cartridge.

PHONO 2 - For connection of the output from a moving-coil phono cartridge.

TUNER - For connection of a FM or AM/FM tuner.

AUX 1 - For connection of an auxiliary line-level input.

AUX 2 - Same as Aux 1.

TAPE 1 - For connection of the Line Out or Playback Out from a tape deck.

TAPE 2 - Same as Tape 1.

EXTERNAL PROCESSOR - For connection of the output from a signal processor or equalizer.

Rear-Panel Connections: Outputs

TAPE 1 - For connection to the Line In or Record In of the same tape deck connected to Tape 1 Inputs on the C-1.

TAPE 2 - For connection to the Line In or Record In of the same tape deck connected to Tape 2 Inputs on the C-1.

EXTERNAL PROCESSOR - For connection to the inputs of the same signal processor or equalizer connected to Ext. Proc. inputs on the C-1.

MAIN 1 - Main Outputs for connection to the inputs of a stereo power amplifier used to drive a pair of loudspeakers.

MAIN 2 - Same as Main 1. May also be used to drive a second power amplifier, or may be modified for low-gain applications (see Problem-Solving Appendix).

GROUND - For termination of Phono ground lead(s).

Rear - Panel Controls:

INFRASONIC FILTER - Eliminates ultra-low-frequency signals associated with phono disc playback, when engaged.

PHONO 1 LOADING: Adjusts capacitive loading of Phono 1 input. See Appendix.

Other Rear-panel Features:

ACCESSORY OUTLETS:

SWITCHED - Active only when C-1 Power is ON. Maximum total power rating: 500 W.

UNSWITCHED - Active whenever C-1 line cord is plugged into an active AC receptacle. Maximum total power rating: 1000 W.

LINE CORD: Should only be plugged in after completing all input and output connections to other components. US models: 110V/50-60 Hz. Export/foreign models: 220V/50-60 Hz.

DETAILED FUNCTIONAL DESCRIPTION

Making Rear-Panel Connections:

Recheck the front panel to make sure the Power switch is OUT (off) and that the Volume Control is set for no output (full-counterclockwise). The line cord should be unplugged.

We will describe a typical installation with one turntable, two tape machines, an equalizer, a power amplifier and two loudspeakers.

Phono Connections

The C-1 contains two completely independent stereo pairs of phono preamplifer circuits. These two sets of phono circuits are optimized for specific uses. Although you may connect the signal cables from a turntable to either pair of phono input jacks, the type of phono cartridge you are using in the turntable will determine which Phono input set is best.

In terms of frequency response, Phono 1 and Phono 2 are identical. In terms of input impedence, gain and sensitivity, however, they differ substantially.

Phono 1 - Phono circuit has a total gain of 35 dB, appropriate for moving-magnet and moving-iron cartridges.

Its input impedence consists of 47K ohms resistance in parallel with a known capacitance (as determined by the rear-panel Phono 1 Loading switch). This prevents the complex input interactions which can cause unpredictable response aberrations in some systems. For further details on how to select the best Phono 1 loading capacitance, see the Appendix on that subject. Refer to the instructions supplied with your Turntable to determine which signal lead carries the Right channel signal and which the Left. When plugging the cables into the C-1's input sockets, be sure each plug is inserted fully into the socket, making a tight fit.

If your turntable is equipped with a separate ground lead (usually a single wire terminated with a spade lug), connect it to the ground post, identified by the electronic ground symbol. In most cases, this will minimize audible hum and buzz in the phono signal. If hum is a problem with your turntable, refer to the appendix on Problem Solving. If you have two turntables, you can connect both ground leads to the Ground post.

Normally, these are the only connections made to the Ground post, while all other components are grounded through their own signal cables.

Phono 2 - The two inputs marked Phono 2 are intended for the use of moving - coil cartridges. Phono 2 employs an additional 25 dB of linear gain (for a total of 60 dB to Tape Outputs), designed especially for the relatively low voltage output of moving - coil cartridges. Input impedence is 39 ohms. The C-1 employs a built-in "pre-amplifier" but this design is free of the compromises usually associated with active gain at this stage.

Historically, the problem of obtaining the extra gain required for the moving coil cartridge has been an expensive proposition. The problem has been as follows. To obtain low noise, a step-up transformer has been required, transforming the low cartridge impedance up to the 47K ohm input impedance of the phono stage. In this process, the noise floor of the system is generally established by the 3 to 10 ohm source resistance associated with the moving coil cartridge, and is therefore very low. However, in order to obtain signal performance comparable to even simple active electronic circuits, heroic efforts on the part of the transformer designer are required, and the resulting cost will almost always be many hundred of dollars. Less expensive transformers will be quiet, but will usually exhibit ringing, phase shift, and some low frequency distortion. The problems with most active step-up devices are simply reversed. Moderate cost, smooth high-frequency response, essentially zero phase shift and low distortion may almost be taken for granted, while heroic and expensive design techniques, including cryogenic cooling, have been required to even approach the low noise of a transformer.

So it would seem the best step-up device is no step-up device at all, which is precisely the design approach used in the C-1. A superbly accurate discrete-differential input, combined with the use of new, high current transistors which exhibit an almost unheard-of noise figure of 1.0 dB at room temperatures, made a new "super-gain" phono stage possible. It yields all the performance advantages of active circuits, with noise levels within 3 dB of the best transformers, all at moderate cost and at room temperature.

Connect the ground lead to the C-1 Ground post.

Plug the line cord(s) from turntable(s) into the Unswitched Outlets on the rear panel of the C-1.

Infrasonic Filter - The rear-panel push switch for the Infrasonic Filter, located near the Phono inputs, corrects for difficulties often encountered with disc playback and should be left in the Engage position.

The Infrasonic Filter is an 18 dB per octave filter with an f_3 (-3 dB point) of 15 Hz. Its response is flat within 0.5 dB down to 20 Hz, then attenuates the preamp's frequency response rapidly at lower frequencies. Disc playback is inevitably contaminated to some extent by sub-sonic energy, due to normal amounts of record wrap, tonearm/ cartridge resonance and turntable motor rumble. If not filtered out of the audio signal, this sub-audible energy can overload tape recorders, waste amplifier power, and drive woofers into excessive cone excursions, causing intermodulation distortion audible as muddy bass.

The need for infrasonic filtering increases with disc-playback level. You may not find it necessary to engage the filter at background levels. When recording discs, or playing back at high levels, it is recommended that the filter be engaged. Problems of acoustic feedback are also significantly diminished by using the filter.

"Group delay", an unavoidable consequence of the rapid attenuation of sub-sonic response, can, under certain musical circumstances, have a just perceptible consequence in the audio passband. Infrasonic filtering is thus somewhat of a compromise, but the advantages of its use commonly far outweigh the disadvantages.

The filter may be defeated for testing purposes. The switch is located on the rear panel on the assumption that it will be left engaged for normal use.

Since the filter is located between the selector switch and the Tape Outputs, it will provide its beneficial effects for all input sources except Tape, and will remove sub-sonic content from the signal fed to tape recorders, as well as to the main power amplifier.

Tuner Inputs - Connect signal output cables from an FM or AM/FM tuner to these jacks. If your tuner has both "fixed" and "variable" outputs, you may use the variable outputs and adjust the output level of the tuner so that, when you switch from Phono to Tuner, the volume level will remain approximately the same.

Aux 1 and Aux 2 Inputs - These "line level" inputs are electrically identical to the Tuner inputs, and may be used for other such signal sources, such as a second tuner, a tape-playback machine, TV audio tuner, TV audio from an AC-line isolated TV set (consult with a TV service technician before connecting the sound signal directly from an operating television set, to be sure the set's circuitry is isolated from the AC power-line), or the output of a microphone preamplifier. The Auxiliary inputs may also be used for special applications in which, for instance, the C-1 is used with an integrated amplifier or receiver with power - amp inputs and preamp - outputs.

Tape 1 and 2 Inputs and Outputs - Connect cables from the line-level outputs of two tape decks to the Tape 1 and Tape 2 Input jacks, while connecting the Tape 1 and Two 2 Outputs to the recorder's line inputs. For a single recorder, use Tape 1 Inputs and Outputs. For a second recorder, use Tape 2 Inputs and Outputs.

A tape deck's output jacks may be identified as "Line Out," "Play," or "Monitor." Tape machine input jacks may be labelled "Line In," "Aux," or "Radio," but not "Microphone."

If you are using two tape recorders, it makes no difference which is connected to Tape 1 and which to Tape 2.

If you are using a separate tape noise-reduction unit (Dolby, DBX, etc.) with a recorder, the connections should be made so that the noise-reducer is placed in the recorder's input and output signal paths.

It is recommended that DIN-type record/play sockets not be used if conventional phono jacks are available at the recorder.

For a description of the C-1 Monitor and Dub functions, see the appropriate section on Front-Panel Controls.

External Processor Inputs and Outputs - These jacks enable the connection of any of a large range of signal processing equipment and accessory devices, such as

- Equalizers, graphic or parametric.
- Dynamic-range expanders.
- Noise-reduction filters (single-pass or "non-complementary").
- Special equalizer supplied with some loudspeakers, such as KLH, Infinity, Electrovoice, or Bose.
- Other special-purpose processors such as the Cotter Audio-Bandpass Filter, the Allison Electronic Subwoofer, or the DBX Subharmonic Synthesizer.
- The "front" outputs of an external matrix decoder, ambience synthesizer or time-delay unit.

The External Processor "loop" is identical to the Tape loops in function, except that no Dub function is provided. It may be used for a third tape deck. It is located in the signal path after the tape loops, so it only affects the Main Outputs.

In general, devices which operate by means of a level-sensitive "threshold" or "transition level" control (such as a dynamic-range expander or noise suppressor) should be used in the External Processor loop, since the levels here will not be affected by the C-1's Volume, Balance, or Tone controls.

The External Processor output has the signal selected by the Selector and Tape Monitor system, and this signal is not interrupted by releasing the External Processor switch. The Main Outputs will not have signal, however, if External Processor is depressed and no input is present at the External Processor inputs.

Devices which are not level-sensitive (such as equalizers) may be connected either at External Processor or between the Main Outputs and the power amplifier. Devices such as time-delay units are usually best placed after the Main Outputs, so that the Volume Control will act as a "master" level control, and front/rear balance may be set by the time delay unit's input and output level controls.

The tape signal cannot be "pre-equalized" at the External Processor loop. In order to accomplish this, you may:

- Connect the tape deck's inputs to either set of Main Outputs, so that the input signal to the tape recorder will be affected by all the C-1's controls, by any processor connected to External Processor, and by the C-1's Sonic Hologram circuit. In this case, it will be necessary to separately adjust speaker-monitor level, if possible, with input-level controls on the power amplifier. If your power amp is not so equipped, it may be necessary to disconnect its inputs and monitor the recording signal with headphones either at the C-1's Headphone jack or via that of the recorder in use.
- Connect the tape recorder to the equalizer/processor's own Tape Monitor jacks.
- Connect the equalizer to Tape 2 Inputs and Outputs, depress Dub 2 to 1, and the equalized signal will be fed to a recorder connected to Tape 1 Inputs and Outputs.

Main Outputs - Connect either set of Main Outputs to your main/front stereo power amplifier. Two pairs of Main Outputs are provided for your convenience. Use either or both pairs, depending on your installation.

The second set of Main Outputs may be used to drive a second power amplifier driving additional speakers, to feed processed signals to a tape recorder, to drive a time-delay accessory, etc.

A signal-processor, such as an equalizer, may be connected between either Main Output set of and the inputs of the respective power amplifier which drives speakers you wish to equalize.

If you are using an electronic crossover to bi- or tri-amplify your speakers, connect a cable from either Main Output set to the crossover's inputs, then connect the crossover's High, Mid, and Low outputs, as appropriate, to the separate power amplifiers.

The output impedence of the C-1 is only a few hundred ohms, so it can drive power amplifiers having virtually any input impedence. Options include connecting several power amplifiers to the C-1's Main Outputs, to separately drive different speaker sets, or the use of extra-long connection cables to drive power amplifier(s) located close to the speakers, or self-powered loudspeakers. Conventional cables as long as 30 feet, or special low-capacitance cables as long as 60 feet, may be used to drive power amplifier(s) without difficulty.

The gain of the Carver C-1 is set to provide normal loudness levels with the Volume Control set a mid-rotation. In some installations with a high output source, a highly sensitive power amplifier, and very efficient loudspeakers, a Volume setting of "10:00" might produce the highest Volume level you ever use. This condition has several disadvantages:

- Signal-to-noise ratio (effective) of the system as a whole will not be its best, since the full headroom capability of the preamplifier is not being exploited, while it is operated close to its "noise floor."
- The Volume control taper rate will result in a greater level-change per degree of rotation than at higher settings.
- The residual mistracking of channel levels in the Volume control will more closely approach its 1 dB maximum specification at settings below "8:00."

The output level at the Main 1 Outputs is programmable in gain capability. A simple modification can be performed to reduce output level in 3, 6, and 8 dB steps, as required, to extend the range of the Volume control and correct for problems described above. Contact Carver Corporation Technical Services for further information.

Accessory Outlets - A total of six AC outlets are provided on the rear-panel of the C-1, so that the entire system need only use one wall outlet for most installations.

Recheck to verify that all component power-switches are OFF before plugging any component line cords into these or any AC outlets. If your power amp is not equipped with a power switch, recheck the C-1 power switch to make sure it is OFF and the power-indicator LED is not lit, before connecting the power amp line cord to the C-1 outlet. Make certain the C-1's AC line-cord is unplugged.

The three AC outlets marked "Switched" are only live when the C-1 is connected to a live AC socket and its power switch is depressed. These may be used to power all-electronic system components such as power amplifiers, tuners, and signal processors. The total power drain from these outlets should not exceed 500 watts. In order to protect the C-1 power switch from harmful arcing, do not switch power on or off when high-level music is playing and your power amplifier is drawing significant current from the AC line.

In addition, three "Unswitched" AC outlets are provided, which are always live whenever the C-1 is plugged into a live wall socket. Mechanical and electro-mechanical devices such as turntables and tape recorders should be plugged in here. A device plugged in here may be left permanently on, or may be switched off with its own switch. In order to avoid turn-on transient thumps, devices plugged in here should be powered on with the C-1 front panel controls set to bypass the equipment, or should be powered on before the C-1 power switch is depressed. Power drawn at the Unswitched outlets should total no more than 1000 watts.

<u>AC Line Cord</u> - After completing all audio and AC power connections properly to the C-1, check to see the C-1 Power switch is out, then plug the C-1 line cord into a live wall outlet with voltage appropriate to the unit (110V - 120V for units labelled as such, 220V-240V for export models).

The C-1 line cord is heavy-duty wire capable of carrying large currents required by high-power amplifiers. The C-1 itself requires only modest power (equivalent to most light bulbs) and may thus be powered by a conventional extension cord if required. But if you have a substantial power amplifer connected to one of the convenience outlets, a heavy duty (no higher than 16 gauge) must be used.

DETAILED FUNCTIONAL DESCRIPTION

Front-Panel Controls:

This section will describe the operation of the preamplifier / control functions on the C-1 front-panel, excluding the three switches in the central panel which relate to the Sonic Hologram. Leave these switches in their Out positions for now, their function will be discussed in Section Five.

The C-1 front-panel is usually divided into three sections or control groupings:

(1) The group of six switches beneath the CARVER logo.

(2) The Input Selector switch, Volume and Balance controls, and Headphone Jack.

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

In the first group, the upper three switches relate to the Sonic Hologram, and will be discussed in Section Five.

The Power Switch turns on the C-1, the three switched AC outlets, and the power LED indicator on.
To avoid possibility of any unpleasant surprises, it is good practice to turn down the Volume Control before pressing the Power switch.
If you prefer, you may leave the C-1's Power switch permanently engaged, while using an external remote or timed switch rated to handle the full power draw of your system.

The Speakers Off switch will completely mute the Main Outputs, without affecting any other Outputs, including the Headphone jack.

The Stereo/Mono switch parallels the two independent stereo channels of the C-1 at the Main Outputs, when engaged in the Mono position. It effectively combines or "sums" left and right inputs so that both channels of the Main Outputs will have identical signals.
The most common use for this switch is in checking loudspeaker phasing. Play any stereo or mono recording, press the Mono button and determine whether a solid "center" image develops between your speakers. Reverse the polarity (+, -) of one speaker connection and recheck. The polarity position which develops a clear "center" image between the speakers with fullest bass content, is the "in-phase" position for the speaker wiring.
You should also press the Mono button for playback of old monophonic discs, since this will cancel any "vertical" (L-R) rumble and surface noise for quieter playback.
When listening to a single-channel signal (such as TV audio), you may either use a "Y" cord at the selected Auxiliary or other high-level inputs, or press the Mono switch, to yield output to both channels of the C-1's Main Outputs. Stereo/Mono does not affect Tape or External Processor Outputs.

Input Program Selector - This rotary switch selects the program source which will be heard: Phono, Tuner or Auxiliary. If neither Dub switch is engaged, the same program source is also presented to the Tape and External Processor Outputs for recording.
To minimize bleedthrough, or audible signal when you switch to a Selector position which has no termination of its respective input, it is recommended that "shorting plugs" be inserted in all unused INPUTS. Do Not install shorting plugs in any OUTPUT jacks, as this would short-circuit the preamplifier output. Use of shorting plugs in unused inputs will eliminate any audible "ticks" when the selector is rotated through an unused input.

Volume Control - This is the Master level control for the stereo system. The control is a continuous - taper potentiometer to allow smooth transition from one level to another.
The taper rate of the potentiometer was chosen to provide maximum flexibility and compatibility with other components. It will mute the C-1's Main Outputs completely, for typical input signal levels, at full-counterclockwise setting. Input signal levels will vary from one source to the next, as well as from one disc, tape, or FM station to the next, so it is normal to observe preferred Volume level differing from time to time.

Balance Control - This control adjusts the relative levels of the two stereo channels. In its center-detent position, levels are exactly equal. Clockwise rotation reduces Left channel level, while counterclockwise rotation reduces Right channel level.

Small movements off center produce smaller shifts in the stereo image per degree of rotation than near the extreme left and right position, which makes slight trimming of levels more convenient.

Since this control is located in the signal path before ("upstream of") the Sonic Hologram, extreme rotation will not fully mute either channel if the Sonic Hologram push switch is engaged. Refer to Section Five for further details.

Headphone Jack - All conventional dynamic headphones (except electrostatic models) may be plugged in here. Headphone impedence may be from a few ohms to several thousand ohms, although output level may vary depending on impedence. The headphone jack is driven by a separate internal headphone amplifier, designed to provide the extra voltage and current gain needed. The signals present at the headphone jack are identical to those at the Main Outputs. They are equally affected by Volume, Balance, Tone, and the Sonic Hologram. See Section Five for a discussion of use of headphones with the Sonic Hologram.

The Speakers Off switch does not affect the Headphone jack. Normally, the Speakers Off switch is depressed when listening to headphones, so that level may be adjusted by Volume without overdriving power amplifer or speakers. The Main Outputs, and hence the speakers, are NOT muted by inserting a plug in the Headphone jack. It is recommended that headphones be disconnected from the C-1 when not in use, to avoid risk of damage to them at high Volume settings.

You may use headphone extension cables and "Y" connectors to drive two identical sets of headphones.

The Headphone socket can also be used as a front-panel convenience Main Output for a power amplifier or tape recorder. You will need an adaptor cable to convert the standard three-contact phone plug to a pair of standard phono jacks.

The controls in the recessed left half of the front-panel will be discussed in the order in which an input signal encounters them.

After the Input Selector:

Tape Monitor (1,2) - These push switches, located in the lower-right corner of the front-panel cutout, will connect the output of the corresponding recorder to the subsequent preamplifier circuitry. They may be used for tape playback, or record-monitoring with three-head recorders. Two- or three-head recorders with Dolby FM decoding capability may be set to process an input signal (e.g., FM tuner) when the appropriate Monitor switch is engaged. Similarly, a signal processor connected to Tape inputs and outputs will be in circuit when the appropriate Tape Monitor switch is pushed.

The two Tape Monitor "loops" are in series, Tape 1 is first, Tape 2 is second. If Tape 2 Monitor is engaged, Tape 1 Monitor switch will have no effect.

Tape Dub (1 2, 2 1) - The Input Selector and Tape Monitor switches choose what signal you will listen to, via the Main Outputs and Headphone Jack - The Dub switches, in conjunction with the Input Selector, determine what signal will be presented to the C-1's Tape Output jacks for recording.

With both Dub switches out, the selected input (Phono, Tuner, Aux) will be fed to both Tape 1 and Tape 2 outputs. You will hear and record the same signal.

To copy a tape recording from recorder 1 to recorder 2, engage Dub 1 2. Set recorder 1 in Play mode, recorder 2 in Record mode. You may, simultaneously, listen to a Phono, Tuner, or Aux source.

To copy from recorder 2 to recorder 1, engage Dub 2 1. The same rules apply.

When you finish copying, release the Dub switch to avoid possible confusion, and permit normal recording from the Selector.

DO NOT DEPRESS BOTH DUB SWITCHES at the same time. If you do so when both recorders in in their Record modes, or while both have their own monitor switches set to Source, you may set up a feedback loop, generating a howl or squeal. There is no condition under which there is any reason to depress both Dub switches.

When copying tapes with either Dub switch engaged, you can use the Tape Monitor switches to listen to either the source recorder or the copying recorder, interrupting any other source which might be playing.

If you use one tape "loop" for a tape recorder and the other for a signal processor, such as a noise-filter or equalizer, you can use the Dub and Monitor switches in combination to "pre-equalize" or process signals fed to the tape recorder. Use the appropriate Dub switch to route selected signal first to the Monitor loop with the processor, then from that loop to the other Monitor loop with the tape recorder (e.g., Equalizer in Tape 2 loop, recorder in Tape 1 loop: depress Dub 2 1). Then use the Monitor switch for the tape recorder (as per e.g. above, Tape 1 Monitor) to listen to either the selected source or the pre-equalized tape signal. If the processor has its own "bypass" switch, it must, of course, be disengaged. The Tape Monitor switch for the loop with the signal-processor can then be used to simply switch the processor into the signal path for listening. Engage the other Dub switch (as per e.g., above Dub 1 2) to play tapes via the processor, or "post-equalize."

External Processor - This circuit "loop" is in the preamp's signal path immediately after the Tape 1 and Tape 2 "loops." It is intended for use with any single-pass, play-back-only signal processors, especially those which involve dynamic, threshold-controlled processing.

Engaging the External Processor push switch will route signal present at the External Processor inputs to the subsequent circuitry of the preamp. The signal chosen by the Selector, Tape Monitor and Dub switches is always present at the External Processor outputs.

The External Processor switch may, alternatively, be used as a selector for another line-level input source, connected to the External Processor Inputs. Pushing External Processor will then override any other Selector or Tape Monitor signal choice for listening.

The External Processor loop may also be used as a third tape loop, however there is, of course, no Dub function.

Tone Controls - The C-1 is equipped with a tone-control stage in the signal path after all input selection, Tape and External Processor switching. The Tone controls may be switched in or out of circuit with the Tone/Push On switch. The tone-contoured signal may thus be quickly compared to the laboratory-flat frequency response of the C-1 high level line amp.

The C-1 tone controls provide separate left and right controls for adjusting the Bass and Treble tonal balance of each channel. Normally, Bass and Treble are adjusted equally in both channels, but there are circumstances in which different settings for the channels may be useful. These include:

- Source imbalances of a phono cartridge or tape recorder, or
- An asymmetry in listening room acoustics.

The best procedure is to begin with identical signals in the two channels. A pink-noise generator is ideal, although FM interstation hiss is adequate. Listen alternatively to each channel, using Balance, and adjust the relative settings of the tone controls until the two channels sound as nearly alike as possible. A Real-Time Analyzer is a useful reference if available.

In addition, turnover controls are provided to alter the effect and range of both Bass and Treble controls. The frequencies listed on these switches indicate the approximate point in the audible spectrum where the tone controls begin to have an effect.

The Tone Controls are asymmetrical in boost and cut. They have been designed to be very appropriate to real music listening situations. Boost exhibits minimal "shelving", and reaches a maximum of +12 dB at full clockwise rotation, with the turnover switches depressed. With turnover switches out, boost is reduced to a maximum of +10 dB, but shelving, or flattening of boosted frequency response, still occurs only at the very extremes of the audible spectrum. Bass may thus be boosted without "boom" associated with shelving, in order to increase deep "punch" without mid-bass "fatness." Treble boost will field greater apparent definition, without the "shrillness" which results from treble-boost shelving.

In the cut (counterclockwise) position, treble and bass are not "rolled-off", as with most tone controls, rather energy output in the upper and lower ranges is reduced while maintaining flat frequency response in those ranges. Cut does not exceed -5 dB, since frequency response attenuation beyond this yields a "dead" sound character which is not musically advantageous. Cut is not substantially affected by the tone-turnover frequency switches. This shelving characteristic for the cut mode is very useful, for example, if a recording sounds too "bright." The relative harmonic energy can be reduced without affecting the balance of the harmonic content. Thus, a violin will still sound like a violin, retaining its "air" and "sheen," while reducing excessive "brightness."

We encourage the routine use of the tone controls as a significant adjunct to the enhancement of your musical experience.

THE SONIC HOLOGRAM GENERATOR

We shall begin with a brief summary of the initial setup procedure to make it possible to quickly achieve the improved imaging which the Hologram offers. We shall follow this with a more complete description of how the Hologram works to aid in a fuller understanding of its theory and in the fine-tuning of the Hologram.

<u>THE PROPERLY FUNCTIONING HOLOGRAPHIC IMAGE</u> - At this point, we are going to describe, in words, the sound field of a properly operating Sonic Hologram. Consider the diagram (fig. A).

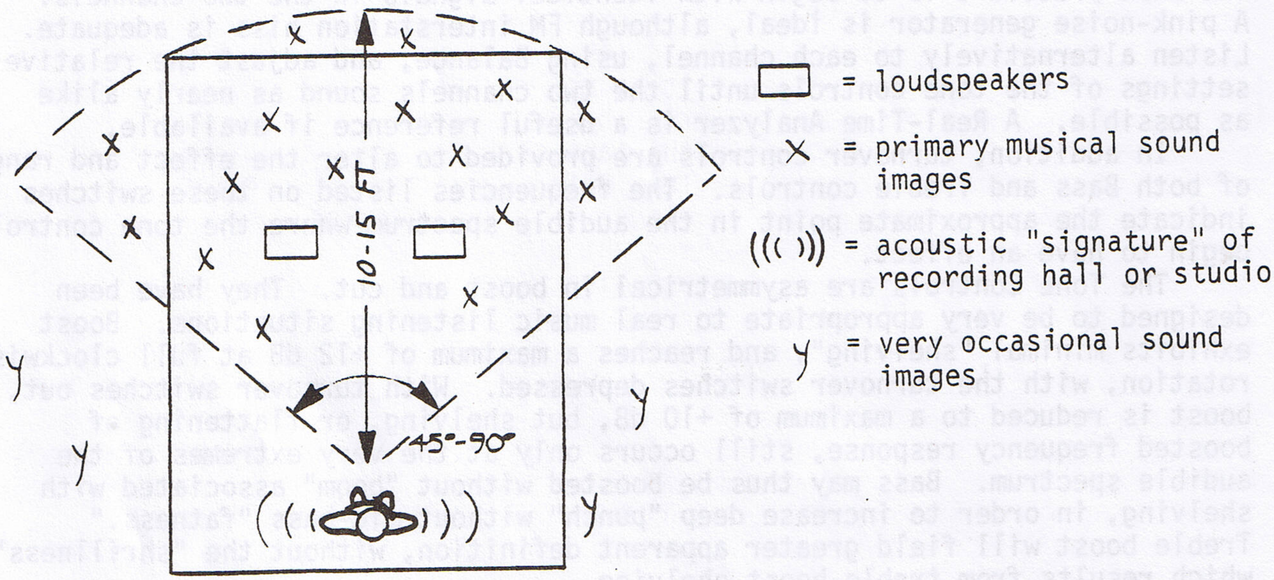


FIG. A

The musical instruments will be spread in a large proscenium arc in front of you. The angle will range from 45 degrees to 90 degrees.

Sound images will exist to the left and to the right, well beyond the limits of the loudspeakers. A very occasional sound image may exist all the way to your right or to your left. The "front-to-back" or "stage" depth will be from 10 to 20 feet in front of you, with sound images clearly floating in space behind, and from time to time, in front of the loudspeakers. You can turn your head to "look" at the sound images and they will stay "put" in space. Sound images will clearly emerge from beyond the limits of the room boundaries. At your listening position, and from all around you, including over your shoulders, you will sense, (and this is very hard to describe) an almost palpable "feel" of the space or "sonic signature" that belongs to the concert hall or recording studio. This sense of the hall is related to very low frequency standing-wave energy, and the complete illusion will be convincing, believable and well defined.

Initial Setup - By following these steps, you can begin to acquaint yourself with what the Sonic Hologram can achieve.

1. Temporarily place your main pair of speakers into the listening room so that they are at least two feet from the wall behind them. Four to six feet would be even better. Temporarily obtain a chair that may be easily moved about.

2. Place the speakers at seated-ear level, on stands is appropriate. Move them so they are clearly closer together than you will be away from them at your listening position.

3. Refer to Fig. C. Obtain a chair that can easily be moved about, and place it in a position as shown at the listener. As carefully as you can, have the chair on the perpendicular bisector of the line running between the two loudspeakers. This imaginary line is called the "preferred axis" Ideally, the listener should have a wall behind him approximately 1 to 4 feet.

Put a chair in the preferred listening position. Take a steel tape rule of length L and have a helper hold it to the center of the chair. (The top or the seat or the back. Just so it's the center). Position the two speakers on the radius of the arc formed by the tape rule. Choose an angle somewhat narrower than the normal stereo angle. Refer to fig (1).

Be very careful not to use a string. This could result in it not being the same length when positioning the right and left speaker, because it will stretch. Locate the speakers on the radius of the arc within ¼ inch. Yes ¼ inch!

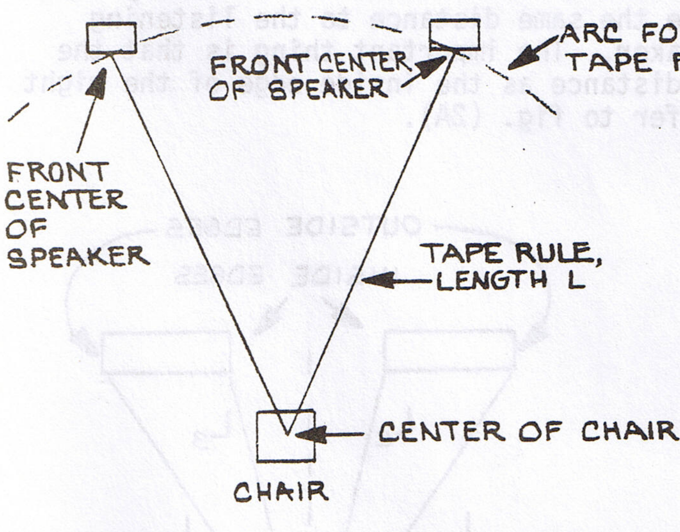
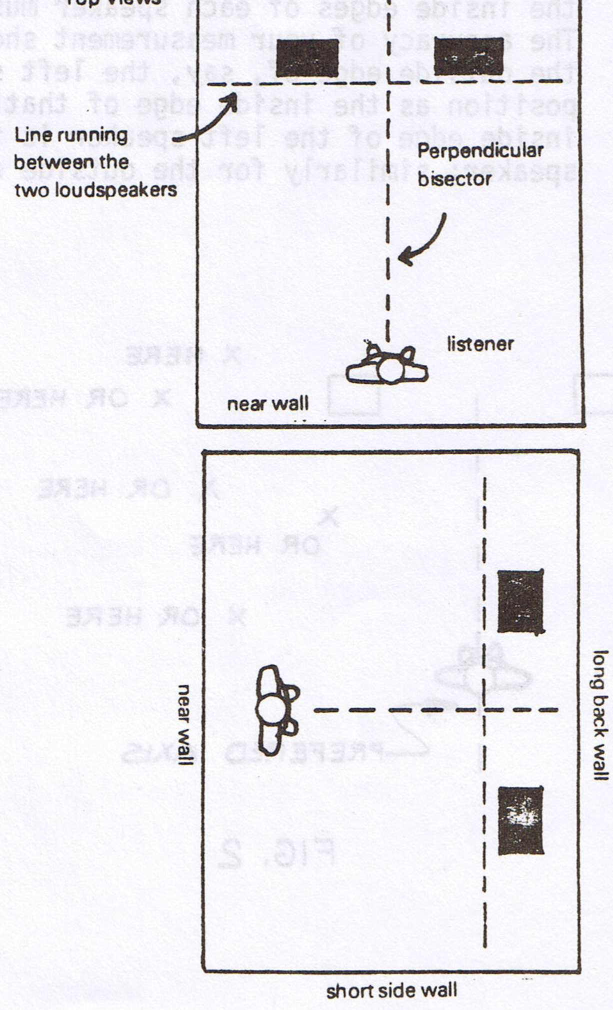


FIG. I

FIGURE C
Top Views



4. Play a music program from a stereo source. Set the Balance control fully to clockwise. Sound should come from one speaker only.

5. Depress Sonic Hologram button. Depress also Injection Ratio (Theoretical). Leave Separation switch out ("Normal").

6. Walk up to each speaker in turn, verify music from both speakers, perhaps at different levels.

7. Sit in the chair, face the speakers.

8. The sound image should seem to originate over to the right side, clearly beyond the right loudspeaker. It may appear in front of, or behind the plane of the loudspeaker. Figure (2) shows where the sound should come from.

9. Repeat, with the Balance control set fully to the counter clockwise. Locate the preferred axis, where the sound will shift from the left speaker to the left side of the room, and from the right speaker to the right side of the room. Note your best position.

10. Return the Balance control to its center-detent position.

11. The sound-stage should now have a significantly broader and deeper perspective than before. A great deal of added "depth" should present itself. Sound images may appear to originate from beyond room-boundaries, or even directly to the left or right. Experiment with a variety of recordings, always returning to the center-line. Enjoy Holography!

12. If your loudspeakers are unusually wide, for example Dahlquist Model DQ-10's you must make certain that the outside edge of the right speaker and the outside edge of the left speaker are both the same distance to the listening position. Similarly the inside edges of each speaker must be the same distance to the listening position. The accuracy of your measurement should be within ¼ inch. It is not necessary that the outside edge of, say, the left speaker be the same distance to the listening position as the inside edge of that same speaker. The important thing is that the inside edge of the left speaker is the same distance as the inside edge of the right speaker; similarly for the outside edge. Refer to fig. (2A).

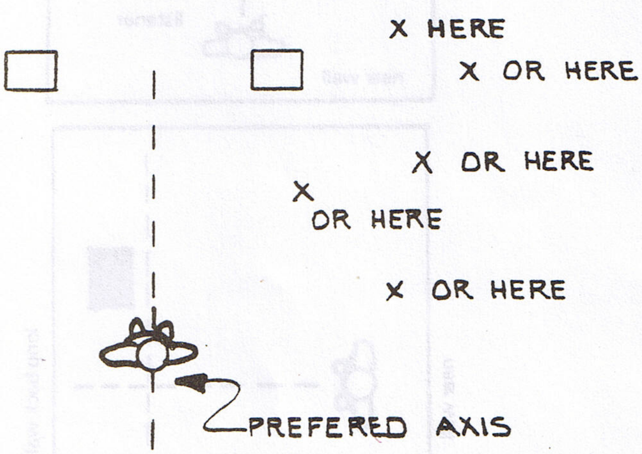


FIG. 2

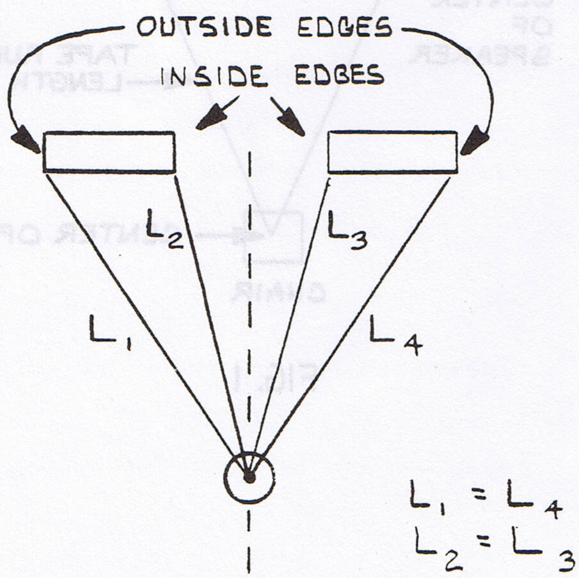


FIG. 2A

UNUSUALLY WIDE SPEAKERS

THEORY OF OPERATION

This section describes the concept of sonic holography and outlines how the Carver Sonic Hologram Generator works to produce its dramatic sonic benefits. You don't need to study the theory in order to make it work for you. We suggest that you read this section at your convenience in order to understand the logical underpinnings of the Sonic Holography controls and set-up instructions.

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 voice 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 Egyptain 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 pleasant vehicle for listening to recorded music, and -- like a Renassance 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 audiphiles 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 distrotions, 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 <u>not</u> 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 while 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 adaption.

Now, the problem with stereo is simple: each ear hears 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.

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 produced 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 simulataneously 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 panaramic 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, generally tending to give the sensation of "sound inside ones head". The reason, again, is too many sonic arrivals; in this case caused by multiple interactive reflections associated with the ear canal and the headphone diaphram/earcup. However, its most important limitation is economic: most listeners don't like to be confined to headphone listening, so binaural recordings have very little sales potential. As a practical matter most recordings must be engineered for loudspeaker playback.

The goal of the Carver Sonic Hologram Generator is to eliminate the "extra" two sonic arrivals that occur with conventional stereo playback, but which do not occur in real life. The ear/brain system can thus receive the unambiguous timing and phase information that exists when we listen to real sonic events with only two arrivals, one per ear. A great deal of the subtlety of a real performance, including a clear sense of the size, or "sonic signature" of the performance environment can be recovered from the recording, which is all but lost in conventional stereo playback.

As noted earlier, the essence of the problem is that both ears hear the sounds from both loudspeakers. The sound from each speaker reachs 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. So, conceptually, the object of sonic holography is 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 from 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 0.2mS 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 opposite-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 in such a way that acoustic interference patterns of the sounds occur in the air near each ear, revealing the true three-dimensional sound image that was hidden in the stereo recording. Recall that "stereo," in Greek, means "solid" or three-dimensional. Ideally, stereo is intended not only to paint a sonic image onto the wall between the speakers but to yield realistic perception of depth as well.

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 ears. 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.

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 free-standing 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 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 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 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. 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 your 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 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 were 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.

MAKING HOLOGRAPHY WORK: FINDING THE PREFERRED 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 by delivering 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 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, the timing of the arrivals will not be correct and the holographic effect will be substantially diminished.

It is NOT sufficient 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. The speakers should be equidistant from your chair to within an accuracy of one fourth of an inch or so. But this does not mean that your head must also be laterally centered to the same accuracy. Carefully study Fig. (4).

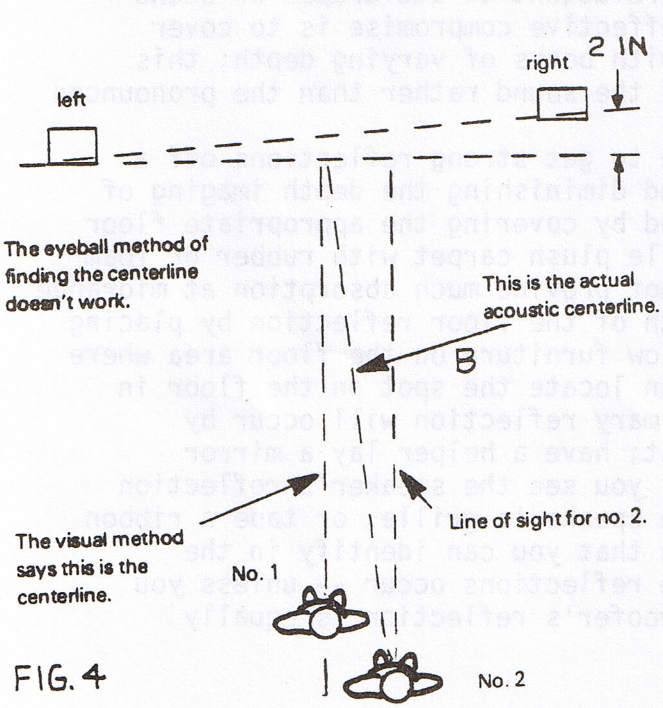


FIG.4

In the following example, the right speaker is set back two inches relative to the left one; too small to be easily visible. This causes the "real" or acoustic centerline to shift to the right, as shown. Person number two is in the correct listening position, but it will seem to him, visually, that he is too far over to the right.

Person number one will swear (visually) that he is precisely between the speakers, in fact, he is actually too far over to the left side, and the Sonic Hologram will not work properly. If number one shifts over to the right so as to intersect line B, he will be in the optimum position.

Inspecting Fig. 4 shows us that if the right speaker were moved forward a mere 2 inches, the acoustic centerline at the listener location would shift more than a foot or so towards the visual centerline. A steel tape rule is an easy way to cause the visual and acoustic centerlines to coincide; and it will give you the correct listening location. Refer to fig. (1).

Speaker alignment can easily be established with the aid of a steel tape measure or a spare length of speaker wire. Fasten one end of the tape measure 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). Mark it where it touches the center of either 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 tape - measure test, until you have the distances to both speakers equal.

If you have an odd-shaped speaker and you are not sure what point to measure to, measure to the midrange drive.

MAKING HOLOGRAPHY WORK: THE CONTROLS

All of the work in Sonic Holography is in the initial set-up -- controlling early reflections and matching the speaker-to-listener distances. The routine operating controls are extremely simple.

SONIC HOLOGRAM

This pushbutton switches the entire holographic imaging circuit into the signal path (button IN) or bypass it completely (button OUT).

HOLOGRAPHIC INJECTION RATIO

This pushbutton switch can be set at either THEORETICAL (button IN) or NORMAL (button OUT), and you should feel free to try either setting with various recordings to see which gives better results. THEORETICAL gives a slightly more pronounced holographic effect, with maximum enhanced depth and breadth in the image, and it tends to work best with recordings made using modern multiple-microphone mix down techniques. But you may find that a piano recorded with multiple microphones (to elicit "stereo" from the piano) may seem located in two places in space. So with these, the normal setting works best. Orchestral recordings made with just two or three microphones in a concert hall will also most often work best in the normal position.

SEPARATION/BLEND

This switch will slightly reduce the L-R stereo separation in the signal which the Hologram Generator treats with inverted, delayed crossfeed. It may be depressed to help compensate for recordings of a single solo instrument which may sound like two separate instruments, or for early-stereo recordings in which left-only and right-only signals may not image realistically. The separation function is distinct from Injection Ratio, in that it affects the stereo content of the signal, while Injection Ratio alters the crossfeed levels in the Hologram Generator.

Pushing Separation/Blend will also produce a smoother "holographic" effect for most headphone listening. Although the problems which the Hologram corrects for do not occur with headphones, feel free to experiment with the effect - the C-1 Headphone jack will receive a hologram signal when the Sonic Hologram is engaged.

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 farther. 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 between the wall and 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.

If these characteristics are not obvious to you when you first switch on the Sonic Holgram Gen ator, 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 your-f 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 accliminating 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:

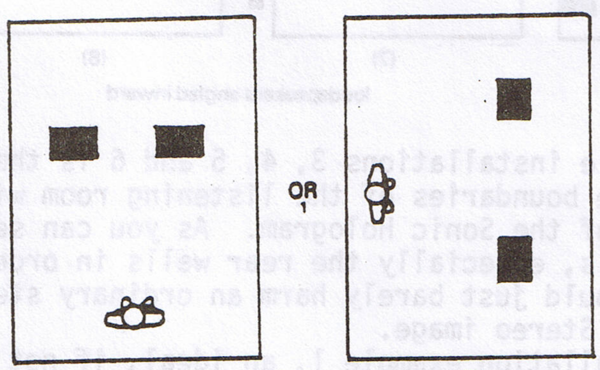
(1) Are the two speakers positioned at precisely equal distances from your chair, as measured with a tape measure?

(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 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.

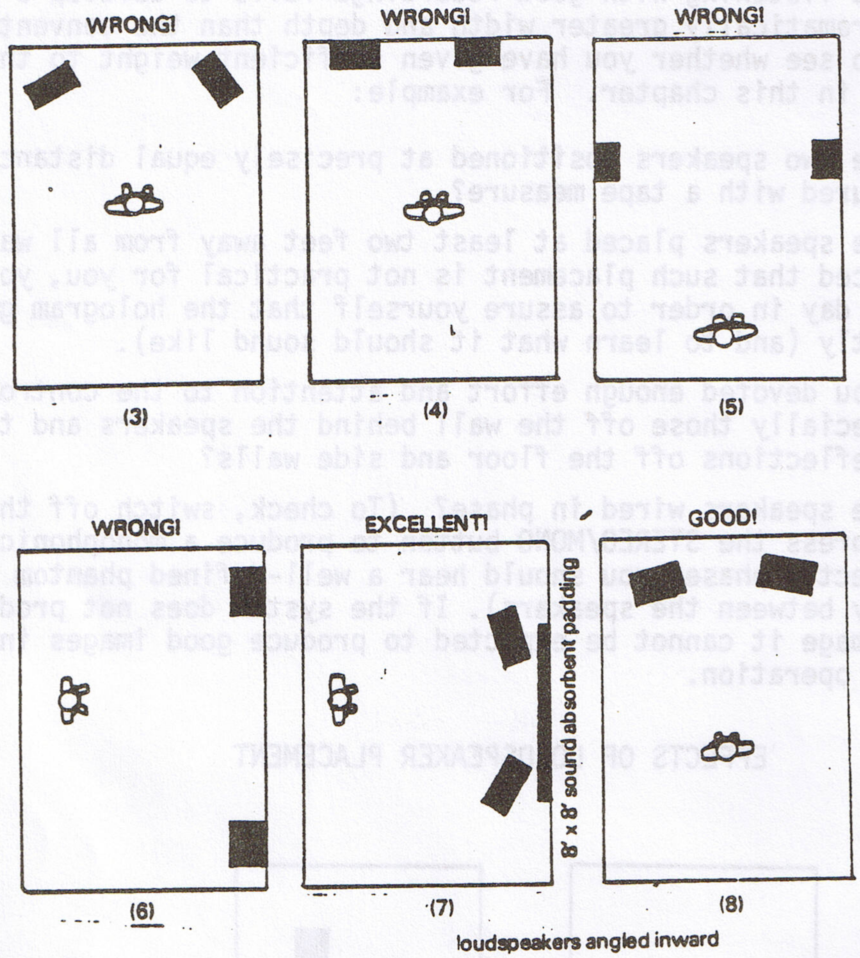
EFFECTS OF LOUDSPEAKER PLACEMENT



(1) Example (1)   (2) Example (2)

The installation shown in Example #1 will yield a large front-to-back depth and moderate stage width. The installation in Example #2 will yield a very wide stage with more moderate front-to-back depth.

Shown below in Examples 3, 4, 5 and 6 are speaker placements that are common to ordinary stereo applications but will not work properly with the Sonic Hologram. Example 7 will work well. Especially if sound absorbent padding is used.



The difficulty with example installations 3, 4, 5 and 6 is that in each case, sound wave reflections from the boundaries of the listening room will severely dilute, or even destroy, the illusion of the Sonic Hologram. As you can see, figure 1 places the speakers away from the walls, especially the rear walls in order to minimize reflections. Reflections that would just barely harm an ordinary stereo image can easily destroy the Holographic Stereo image.

Let's take a look at installation example 1, an ideal, if not fully practical installation in most listening rooms. There are several things that combine to make this setup excellent. First, the speakers are located away from the side walls, so as to minimize reflections from the side. Also, they are located away from the rear wall, also minimizing reflections. Secondly, the listener is situated so that he is close to a near wall and immersed in a sound field that is made up of direct sound from the speakers and reflected sound from the near wall behind him.

Example (7) is an excellent choice. The speakers are located away from the side walls. The listener has behind him a reflective wall. The reflective wall behind the listener will serve to increase the front-to-back image. The loudspeakers are sharply angled toward the listener. This helps to increase front-to-back depth, and also to minimize the amount of the side wall reflections.

The loudspeakers are one foot away from the wall. The sound absorbing padding causes the Sonic Hologram to "think" that there is a large space behind the speakers; this "virtual" space gives the holographic image a place in which to build and to occupy. The padding is optional; it serves to increase front-to-back depth.

EFFECTS OF SPEAKER DESIGN

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 may 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 design), 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.

(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 ½" or 3/4" layer of sound-absorbing felt with holes cut to expose the drivers.

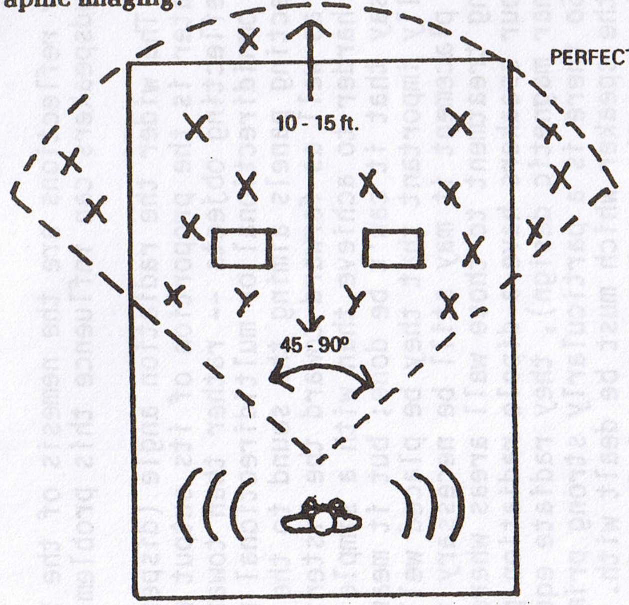
MULTILPLE DRIVER "TIME ALIGNMENT" IS NOT, REPEAT, NOT IMPORTANT TO SONIC HOLOGRAM.

As long as each driver can be approximated as a point sound source, the Sonic Hologram will work equally well, regardless if the loudspeaker is "time aligned" or not.

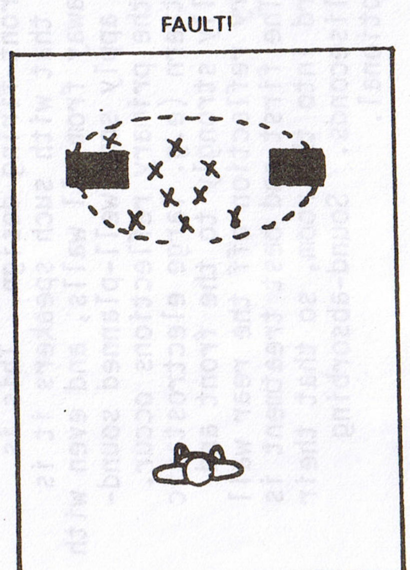
If your speaker does not have visible evidence of design effort to reduce early reflections, this by no means implies trouble. We have found that most speakers with a conventional front panel surface do not have serious reflections, and do indeed work very well with the Sonic Hologram. It is impossible to predict by simply looking at the loudspeaker.

We are proud of this achievement and hope these instructions make it possible for you to share in the experience.

Diagrammed below are sound images shown with an X for a properly set up system. Also shown, are a number of systems that are set up but not operating properly for one reason or another. Study the illustrations carefully, for an understanding of these drawings will help you arrange your system for the best possible holographic imaging.



This is the most common fault of all and occurs almost universally when the hologram is switched on for the first time. It is caused by not being on the preferred axis and/or too many reflections.



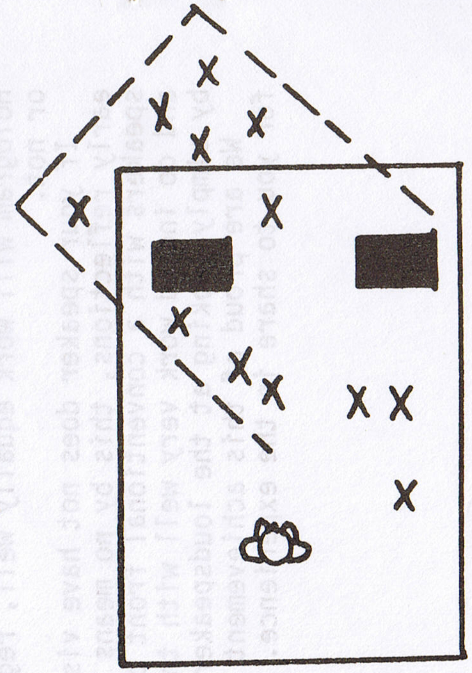
Fault!
Sound images fail to extend beyond the confines of the loudspeakers. Sound image behaves pretty much like ordinary stereo.

Very common fault. Will almost always occur until system is optimized. Do not worry.

Fault!
Sound images fail to extend beyond the right wall.

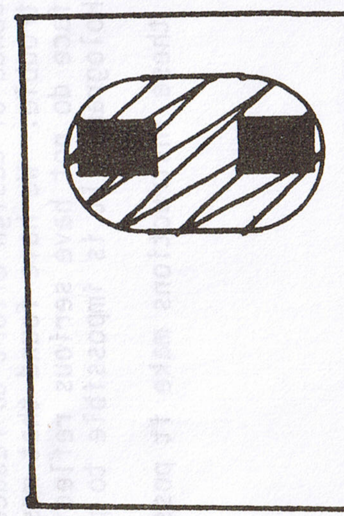
Cause: Large reflective surface on the left, absorbent surface on the right. Or, a larger area on the left side.

Cure: Tow speakers inward: place sound absorbing sheet where images bump into wall: Move speakers away from wall.



Fault!
Sound images blurry, fuzzy. Localization difficult. Musical instruments "smeared" over a large space. Imaging fails to take place.

Cause: Extremely early reflections that occur within the first 300 micro seconds; i.e., 300 millionths of a second. It is almost always caused by reflections associated with the front panel baffle board of the loudspeaker. It is very easily solved by placing a cutout of 3/4" felt around each of the drivers on the front panel board of your speaker.



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APPENDICES

APPENDIX A

LOUDNESS Switch - The Bass-turnover switch is labeled "Loudness" in its push position. With Tone/Push On engaged and the Bass control(s) set for boost, that boost will correspond to the proper acoustic compensation for low listening levels, if the Loudness switch is depressed. This Loudness is not related to the Volume control, and may thus be adjusted to your taste with the Bass controls, separately for each channel, at all listening levels.

Muting System - The C-1 employs an electronic "clamper" to mute the Main Outputs and the Headphone output to reduce turn-on and turn-off transients. This muting system will turn off the signal to your power amplifier and headphones:

- For about 3-5 seconds after initial Power-On, whether by the front panel power switch or by a remote or timed switch.
- Immediately at turn-off, whether by the Power switch, or by remote or timed switch.

The muting system will not interrupt the Tape or External Processor Outputs.

The Main and Headphone Outputs will mute if the line voltage drops below 80% of full voltage, to prevent thumping.

Muting is accomplished by an "electronic relay", which, when it is "closed", has no audible or measurable effect on the signal. It is controlled by a toggling device, so muting cannot be partially "on", and will never degrade the signal.

Line Voltage - Units built for 110V-120V line voltage can be converted to 220V-240V by means of a modification. This is not simply a switch, and must be performed by a Carver Authorized Service Station. Contact Carver Technical Services for more information.

APPENDIX B

Selecting Phono 1 Input Capacitance - 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. If it is less than 100 pF, set the PHONO 1 LOADING switch to 0 pF. If the computed value is greater than 100 pF, you should 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 chance of 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 100 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 extreme highs.

If you use a high-output moving-coil cartridge, you should use Phono 2. Slight linear differences in high-frequency response, due to differences in the resistive loads of the two phono stages, will occur in comparing Phono 1 to Phono 2 with a high output moving-coil. Phono 2 will be slightly quieter with the low source impedence of the moving-coil as well.

APPENDIX C

SPECIFICATIONS

<u>Phono Inputs</u>

RIAA ± 0.25 dB "extended" RIAA curve. Overload (Phono 1)= 100 mV @ 1 kHz.

<u>High-Level Inputs</u>

Freq. resp. 5 Hz to 200K Hz +1 -3 dB, Infrasonic Filter, Tone, Hologram OUT.

<u>Infrasonic Filter</u>

18 dB per octave below 20 Hz. f_3= 15 Hz

<u>Noise</u>

Phono 1: 82 dB, IHF-A, below 5mVrms at 1K Hz

Phono 2: 86 dB, IHF-A

High Level: 96 dBV, IHF-A, below 2 Vrms

Hologram: 92 dBV, IHF-A

Infrasonic Filter: 95 dBV, IHF-A

Tone: 96 dBV, IHF-A

Distortion

THD: 0.04% or less, below 3 Vrms out.

IM (CCIR or SMPTE): 0.04% or less

TIM unmeasurable.

Sonic Hologram Generator (Patented)

Image resolution: 5 degrees horizontal, 20 degrees vertical, in Theoretical mode.

Dimensions

19 x 3½ x 10"

Weight

6 lb. (2.7 kg)

APPENDIX D

PROBLEM SOLVING: HUM, NOISE, RFI

Under normal operating conditions you will not hear any hum originating in the circuitry of the C-1. 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' signal levels to normal. If your power amplifier lacks input level controls, the C-1 may be modified to reduce its fixed output level. Contact Carver Corporation Technical Services for further information.

Except for the condition described above, audible hum will nearly always be found to be due to problems external to the C-1 -- 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 house 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.

Radio Frequency Interference - RFI 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-1. If you encounter RFI the first step is to 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 simple 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 interference to tolerable levels simply by turning the tape deck 90 degrees or moving it to another location in the room.

APPENDIX E

GENERAL TROUBLESHOOTING

In view of the C-1's input/output flexibility, 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-1 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 causes. 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 Stereo/Mono button? Was the preamp's AC line cord accidentally pulled partially out of its wall socket during housecleaning earlier in the day? 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?

PROBLEM	SUGGESTED CAUSE
No sound.	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).
No sound (power on).	1. Input Selector set to inactive input 2. Either Tape Monitor button depressed with no tape machine running.

PROBLEM	SUGGESTED CAUSE
No sound (power on).	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).
No sound in one channel.	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 counterclockwise. 4. Imperfect contact in switch (especially any lever or slide switch in a program source or signal processor, as well as the various signal-routing switches in the preamp). 5. Speaker fuse blown.
Loud howl, squeal, or whistle.	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.
Solo voices or instruments sound thin, shrill, or distorted.	1. Treble controls set to maximum boost. 2. Phono cartridge wired out of phase.
No holographic projection of sound images to one side, or a better image-shift on one side than on the other.	1. Reflective surface on side of room opposite to side on which the image is not projected. Move opposite-side speaker further away from walls. 2. Setup of speakers not symmetrical in listening room resulting in the condition described above. Center the speakers.

PROBLEM	SUGGESTED CAUSE
Hologram does not produce a definite center point for listening on the acoustic preferred axis. Left-right head motion during setup results in many partial preferred axes.	1. One speaker farther away from listener than the other. Use tape-measure as outlined earlier 2. Loudspeakers A. Too close to reflective wall surfaces. B. Too far apart (listening angle too wide). C. Have too wide a dispersion pattern. D. Have front-board diffraction problems. 3. Listening room too "Live" especially in area of loudspeakers.

APPENDIX F

CHANGING THE GAIN OF THE PHONO STAGE

The following gain changes may be performed by by the dealer.

Factory set gain of Phono 1 is 35 dB. The gain may be changed as follows:

30 dB - change R 207 and R 307 to 3.9 K ohms
35 dB - nominal 2.2 K ohms
40 dB - change R 207 and R 307 to 1.2 K ohms

These resistors are located on the circuit board close to the Phono 1 stage. They are identified by white ink lettering.

Factory set excess gain of Phono 2 is 25 dB. The gain may be changed as follows:

20 dB - change R 219 and R 319 to 220 ohms
25 dB - nominal 120 ohms
30 dB - change R 219 and R 319 to 68 ohms

APPENDIX G

CHANGING THE GAIN OF THE LINE AMPLIFIER

The line amplifier gain may be reduced as follows:

	R 258	R 358	R 259	R 359
Reduced 3 dB	910 ohm	910 ohm	4.3 K	4.3 K
Reduced 6 dB	1.3 K	1.3 K	2.2 K	2.2 K
Reduced 9 dB	1.8 K	1.8 K	1.5 K	1.5 K

These resistors are located close to the main out 1 jacks. They are identified by white ink lettering.

APPENDIX H

SERVICE/WARRANTY

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 1237
Lynnwood, WA 98046

or call (206) 775-1202. 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.

APPENDIX I

PATENT NOTICE

The purchase of the Carver C-1 gives an implied license only to use the apparatus to play sound recordings, but not to make sound recordings. For those purchasers who wish to use the C-1 to make a small number of single recordings for the purchaser's private entertainment and not for a commercial purpose, present policy is to grant a limited license to permit use of this apparatus only for that purpose on a royalty free basis, with the understanding that such single recordings are not to be copied or sold.

However, the purchaser is warned that making a sound recording from an existing sound recording may be a violation of a third party's copyright. The royalty-free license granted should not be construed as permission to make a copy or a modified copy from any copyrighted work, since permission to do so should be obtained from the copyright holder.

For those who may wish to make sound recordings incorporating the technology of the SONIC HOLOGRAM GENERATOR on a commercial basis, consideration is presently being given to a licensing program. Contact Robert W. Carver, P.O. Box 1237, Lynnwood, Washington 98046.

Carver Corporation
19210 33rd Avenue West • Lynnwood, WA 98036
P.O. Box 1237 • Lynnwood, WA 98046
(206) 775-1202