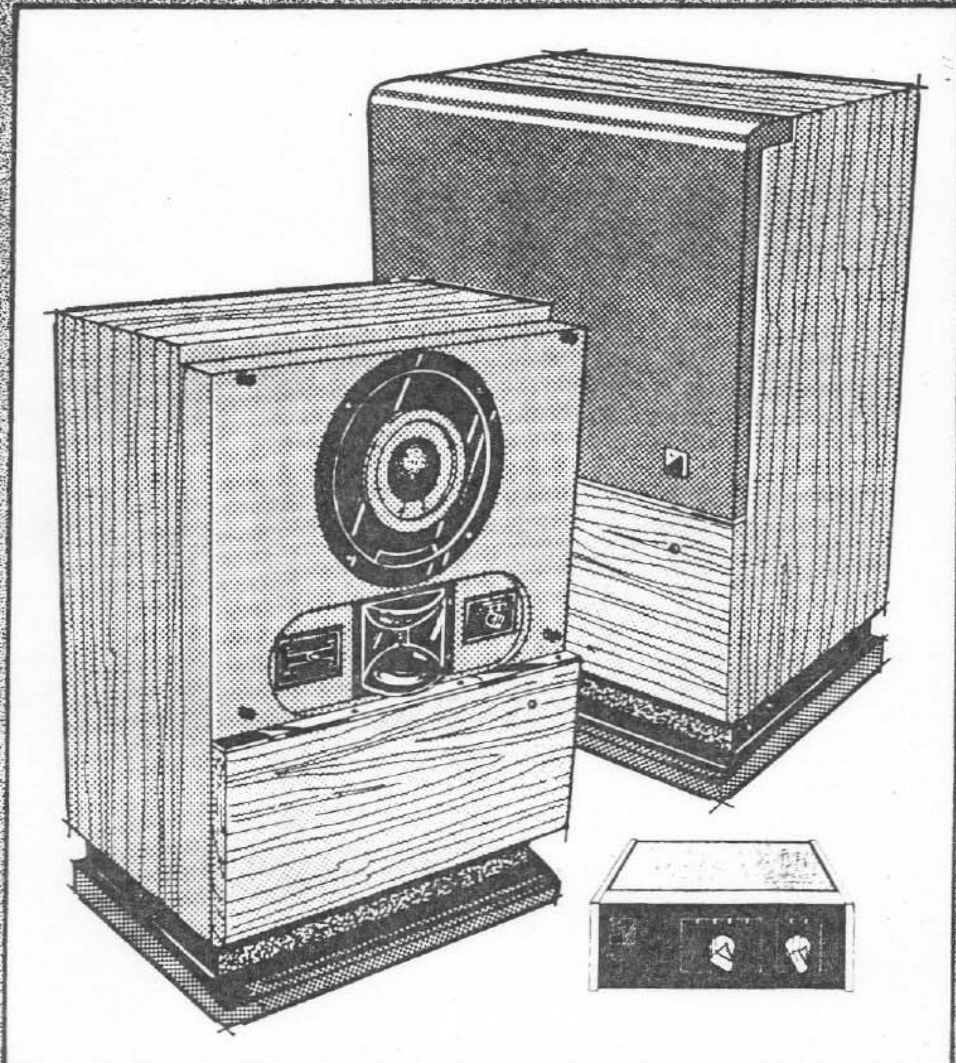
Interface™:D



Owner's Manual

INTERFACE:D OWNER'S MANUAL

The Interface:D system offers significant performance advantages over most conventional high-fidelity speaker systems. Its vented, equalized woofer design provides the basis for a truly unique combination of high accuracy, wide bandwidth, high efficiency and modest size. Because of its somewhat unusual nature, the Interface:D also has a few special connecting and operating requirements. We have tried to provide clear and detailed unpacking, connecting and operating instructions. Please follow them closely to assure correct and trouble-free operation.

The most basic and widely applicable information has been printed on a grey background. This information should be sufficient for most installations.

The other information covers more advanced, detailed system and application considerations. Nevertheless, in a manual of this scope, it is difficult to cover every possible concern in great depth. We suggest consultation with your dealer and, additionally, welcome your inquiries at Electro-Voice. Address and telephone information is part of the warranty statement at the end of this manual.

Good listening!

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UNPACKING

Unpacking the Interface:D speaker system and equalizer is straightforward, with no special precautions necessary. However, if at all practical, *retain all packing materials for possible future use* (see "Customer Service" section).

ALWAYS USE THE EQUALIZER!

The Interface:C/D equalizer should always be used with the speaker systems. A sharp cutoff below 28 Hz protects the woofer from large subsonic signals which the speaker system is not designed to reproduce.

These subsonic signals are usually non-musical in nature, most often produced by irregularities in record surfaces. The signals can produce large, damaging woofer excursions at moderately high listening levels. At

best, such excursions represent wasted amplifier power and produce increased distortion in the frequency range covered by normal program material.

Note: the octave-band and similar equalizers used by some audio enthusiasts to correct room/speaker deficiencies or custom tailor program material are typically not capable of the relatively sharp boost-and-cut contouring of the Interface:C/D equalizer below 100 Hz. Such equalizers must be used in conjunction with the Interface:C/D equalizer. For hookup instructions, see "Using the Interface:C/D Equalizer with Other Accessories."

CONNECTING THE EQUALIZER

The Interface:C/D equalizer may be easily connected to your receiver, integrated control amplifier, or separate preamplifier and power amplifier. Use the two stereo connecting cables supplied and follow the accompanying instructions and diagram.

Most Universal Connection: in the Tape Monitor Path

The tape monitor connection is most universal because virtually all consumer electronics include tape monitor facilities.

In such equipment, a Tape Monitor (also called Tape-Source) switch interrupts the normal signal path through the amplifier. The Interface:C/D equalizer, like tape recorders and other electronic accessories, should be connected to the rear-panel tape jacks, and activated by the Tape Monitor switch. One pair of Tape jacks is usually labeled "Tape Out" and the second pair, "Tape In." Other common designations and the proper interconnections are shown in the illustration on page 5.

Note that the Tape Monitor switch must be engaged at all times for the equalizer to be in the circuit. This is usually accomplished by depressing a Tape Monitor button or lever, or by turning a rotary switch to the Tape Monitor position. Common designations for the proper switch position include "Tape," "Tape In," "Tape Play," and "Playback." See the section entitled "Testing For Proper Equalizer Hookup" if, after speaker hookup, there is any doubt that the equalizer is in the circuit.

Another set of Tape jacks and Tape Monitor switching is furnished on the equalizer, so that other equipment may still be connected into the system, even though the equalizer has "used up" the Tape jacks on the main electronics. For convenience, two pairs of Tape Out jacks are provided; either pair may be used. Both may be used if the input impedance of the auxiliary equipment is above 100K ohms.

A typical system hookup is illustrated on page 5.

Alternate Connection: between Preamp and Power Amp

The Interface:C/D equalizer may also be connected between preamplifer output and power amplifier input if the system includes such separate components. This alternative is attractive if it is desired to keep existing Tape Monitor facilities free or where the preamplifier has no Tape Monitor jacks.

Even some receivers and integrated control amplifiers offer the possibility of the same connection if equipped with Pre-Out and Main-In jacks which interrupt the signal path just before the power amplifier stages. First, the "U-nails" that normally connect the Pre-Out and Main-In jacks must be removed. Then the Pre-Out jacks should be connected to the equalizer input and the equalizer output connected to the Main-In jacks.

Electronics with Two Tape Monitor Circuits
Some electronics have a second Tape Monitor circuit so that two tape recorders can be conveniently accommodated. Usually, either one of the Tape Monitor circuits may be used for the Interface:C/D equalizer, as previously described and illustrated.

However, any signal source connected to the other set of Tape-In jacks on the main electronics will probably not be equalized. The actual situation depends on the internal wiring and switching of your particular electronics. A careful reading of appropriate instructions may be necessary.

Using the Interface:C/D with Other Accessories
Other accessories, such as octave-band equalizers, noise-reduction units, or matrix four-channel decoders, may be used in conjunction with the Interface:C/D equalizer. The most widely applicable hookup is probably a variation of the most universal Tape Monitor connection for the Interface:C/D equalizer, where the accessory unit is inserted either before or after the Interface:C/D equalizer as specifically outlined below.

Noise-Reduction Units, Dynamic Range Expanders, and Matrix Four-Channel Decoders. In general, these units should be connected *before* the Interface:C/D equalizer, between the Tape-Out jacks of the main electronics and the Input jacks on the equalizer. In this way, the audio signal — which may be encoded or specially processed — is treated directly by the intended unit, without the possibly detrimental intervention of the Interface:C/D equalization.

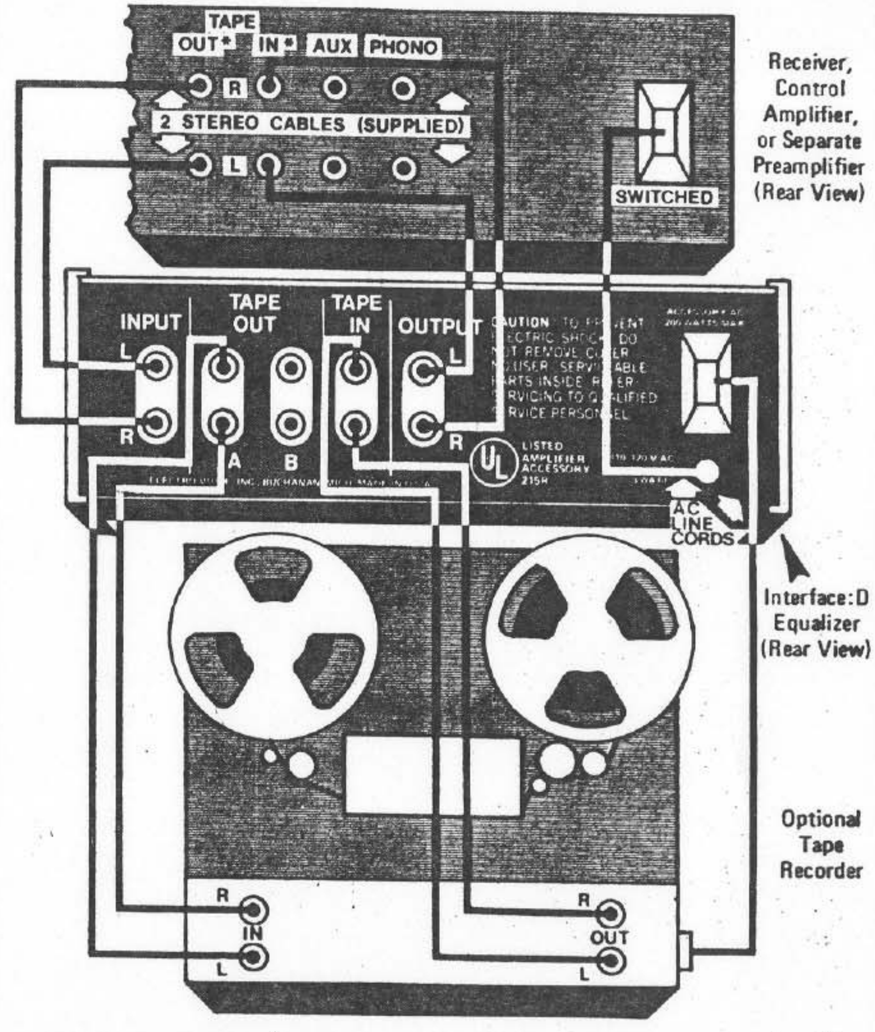
Note that if two pairs of Interface:D speaker systems are used in a four-channel system of any type, two separate equalizers are required.

Accessory Equalizers. Octave-band equalizers and other signal-shaping devices may be connected on either side of the Interface:C/D equalizer: between the Tape-Out jacks of the main electronics and the Input jacks on the Interface:C/D equalizer or between the Output jacks of the Interface:C/D equalizer and the Tape-In jacks on the main electronics.

AC Power
Connect the AC line cord of the Interface:C/D equalizer to a convenient outlet. If a switched AC outlet is available on the main electronics, power will be applied to the equalizer when the main system is turned on. An unswitched AC outlet is provided on the back of the equalizer. Maximum power capacity of this outlet is 200 watts.

A typical AC line cord hookup is part of the illustration on page 5.

INTERFACE:D EQUALIZER HOOKUP

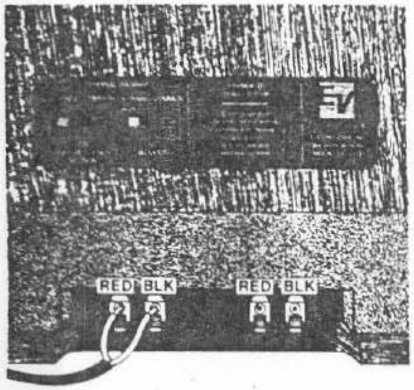


*ALTERNATE DESIGNATIONS
Tape Out: Record, Record Out
Tape In: Monitor, Play, Playback, Tape Play

RECOMMENDED TAPE MONITOR CONNECTION

CONNECTING THE SPEAKER SYSTEMS

Input Connections



There are four input terminals on the rear edge of the speaker enclosure's base. For normal operation, using the Interface:D's internal crossover, *either* terminal pair may be used. The red terminal (8 ohms, +) should be connected to the 8-ohm output terminal of the amplifier; the black terminal (common, -) should be connected to the amplifier's common output terminal. Insert each bare wire end into the small hole in the input terminal, after depressing the spring-loaded wire-release actuator. When the actuator is released, a positive connection will be made.

All four input terminals are used when the Interface:D is bi-amplified, bypassing the internal 350 Hz crossover. See the "Bi-Amplification" section.

In-phase Speaker Operation

Connecting the speakers as described above produces in-phase operation, an important condition for best stereo performance. This ensures that the speaker cones are moving in unison when the same signal is present at each set of amplifier output terminals. Such a signal condition occurs with monaural program material and, in stereo, with soloists or groups located midway between the two speakers.

In-phase operation results in a satisfyingly "solid" center image. Out-of-phase operation produces a spread, indefinite center image that changes location and character as the listener moves a foot or two back and forth between the speakers. Also, out-of-phase operation may reduce bass response, depending on room dimensions and speaker/listener locations.

An experienced listener can successfully test for in-phase operation by noting the quality of the center image on monaural program material. However, the least ambiguous check is to set the two speakers facing each other, an inch or two apart. Use program material with fairly prominent bass content and switch the amplifier to the monaural mode. This is usually accomplished by pressing a Mono or A+B button or lever, or moving a rotary switch to a similarly labeled position. Reverse the wires to *one* of the speakers (either at the speaker end or the amplifier end, *but not at both).* This will either increase or decrease the bass output. The correct connection is the one that produces the most bass.

Wire Selection

To avoid any significant amplifier power loss in the speaker lines, 18-gauge stranded wire (commonly called lamp cord or zip cord) is

satisfactory for lengths up to 30 feet. If longer speaker lines are required, use progressively larger wire sizes: 16-gauge to 50 feet, 14-gauge to 75 feet and 12-gauge to 125 feet. Always use a separate pair of wires for each speaker, even if your amplifier permits using a common ground. Resistance in a common ground wire can degrade stereo separation.

TESTING FOR PROPER EQUALIZER HOOKUP

After the speakers have been connected, there may be some doubt that the equalizer is properly connected. Simply turn the High Frequency Slope control to the Off position while a program source is playing. After a short time, the sound will become very distorted and drop in level, indicating that the equalizer is properly connected. If the sound does not change, the equalizer is not in the circuit.

Alternatively (without turning the equalizer off), the same test may be performed by switching the Input switch from Tape to Source. If the equalizer is properly connected (and nothing is plugged into its Tape In jacks), the program will be completely interrupted.

SYSTEM PLACEMENT

Placement of stereo speakers is more often determined by floor plan and furniture arrangement than by acoustic considerations. There are some general guidelines which may be helpful, however.

Preferred Locations

Usually it is possible to select a normal listening area (a sofa, chair grouping, or whatever) in the listening room. The speakers should face the listener with no obstructions between the speakers and the listener. With the Environment control in the 1/4 space (wall/floor) position, the Interface:D is designed for most efficient, balanced performance when placed within an inch or two of one wall, several feet away from room corners. (The Interface:D will also provide balanced performance when placed on the floor, away from any walls, with a 3 dB reduction in overall sensitivity; see the "Using the Environment Control" section).

Spacing between the speakers is dependent upon the listening room. If the speakers are too widely spaced, the stereo image may be disjointed, with a gap in the middle. In most rooms, a speaker separation of 6 to 12 feet will provide a good stereo image; a good guideline is to have an angle of 30° to 50° between speakers, when viewed from the listening position. Feel free to experiment.

Sound Quality and Speaker Location

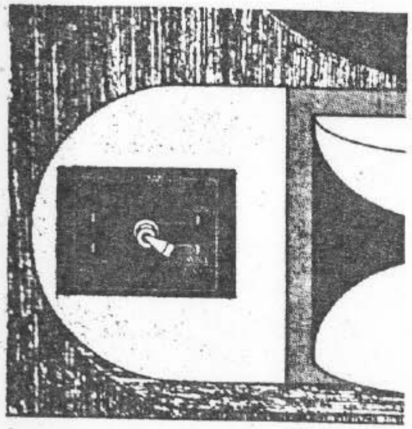
There is no doubt that different listening rooms and changes of speaker location within a given room can affect sound quality to an extent important to many enthusiasts. Some changes are subtle, while others are quite noticeable.

It is possible to theoretically predict and categorize many effects if room characteristics can be simply defined. However, real listening rooms are usually sufficiently irregular and complex to dilute and alter the clear-cut effects that might be predicted theoretically. The following broad statements about the major effects of speaker location change should help the experimentation.

Moving speakers into the corners of the room will increase the amount of bass and mid-bass heard. However, the unusually wide, uniform high-frequency dispersion of the Interface:D makes it less sensitive than many other designs to less-than-ideal corner placement. Also, placing speakers in corners tends to move the tweeter away from your ears. This sometimes creates a "heavy" sound. Again, the Interface:D's excellent high-frequency dispersion tends to alleviate the effect.

USING THE ENVIRONMENT CONTROL

The Environment control adapts the Interface:D for balanced frequency response in different room locations. The Environment control is located behind the speaker grille. The grille may be pulled off its four dowell retaining pins for easy access to the control.



1/4 Space, Floor/Wall Position

The Environment control should be set in the 1/4 space position when the enclosure is placed within 12 inches of a wall. This location provides a true acoustic "quarter space" environment for the woofer, because the woofer's downward-firing orientation places it in close proximity to the adjacent surfaces. The environment is formed by the intersection of the floor and wall which restricts woofer output to one-quarter of an entire sphere, hence the term "quarter space." The restriction increases woofer efficiency uniformly throughout its operating range (28 to 350 Hz). Also, the true quarter-space environment eliminates the response aberrations which can occur with front-mounted woofers, due to interference between the woofer's forward output and that which travels along the sides of the enclosure and is reflected by the wall behind the speaker. (Above the 350 Hz crossover, the front surface of the enclosure provides its own stable acoustic environment for the midrange and tweeter, unaffected by speaker position).

1/2 Space, Floor Position

The Environment control should be set in the 1/2 space position when the enclosure is 3 or more feet from a wall. This location provides an acoustic half-space environment because the woofer is in intimate acoustic contact with only one surface, the floor. When a conventional speaker is moved from a quarter-space into a half-space environment, a general drop in low-frequency efficiency occurs, the exact amount being different at different frequencies. A drop also occurs with the Interface:D, but the woofer's downward-firing orientation provides a reduction that is uniform with frequency at 3 dB. When the Environment control is placed in the 1/2 Space position, the midrange and tweeter are attenuated by 3 dB, to restore balanced reproduction.

In-Between Wall-to-Speaker Distances

For distances between 12 inches and 3 feet, proper control placement will depend on secondary room and speaker location effects. The control should be set by ear, while listening from the normal position with a second person changing the control position. A natural recording of the male voice, with minimal instrumental support, is the best program material. If the sound seems too thin, use the 1/2 Space position; if the sound is heavy, try the 1/4 Space position. Careful evaluation of several familiar recordings should readily reveal the preferred switch position.

USING THE EQUALIZER CONTROLS

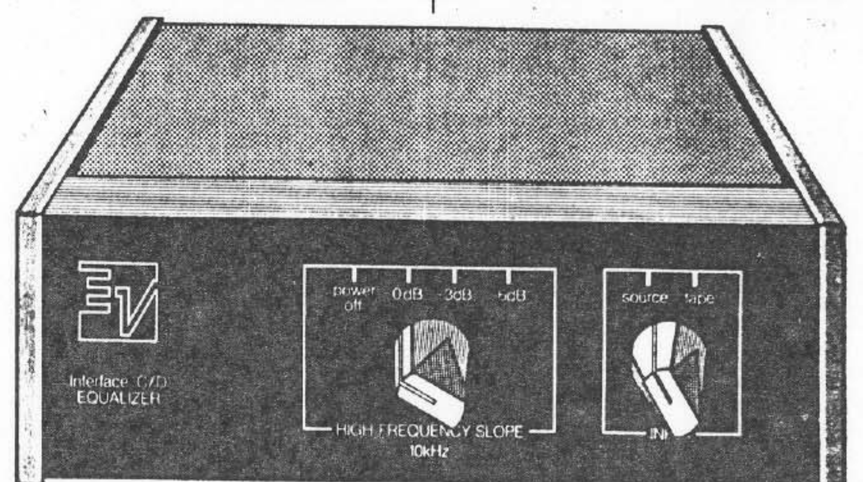
High Frequency Slope Control

The High Frequency Slope control combines the power on-off function with selection of three different high-frequency response contours. When the switch is to the extreme left, the equalizer is turned off. However, if the equalizer is connected to a switched AC outlet on the main electronics, the equalizer will be turned on with the rest of the system and the high-frequency selection can be left set at all times.

Note that the following comments apply only when the speaker's High Frequency Slope control has been set in the Flat position (see the "Using the Speaker High Frequency Slope Control" section).

The 0 dB Position. This setting produces the flattest total acoustic power output at high frequencies, approximately 5 dB down in the 10- to 15-kHz range. The electrical response varies only a dB or so from flat at high frequencies.

We prefer the 0 dB position for the finest low-noise program material. With such material, the aural effect of a relatively flat acoustic power output



is quite amazing, and is one of the **most** outstanding characteristics of **the** Interface:D.

The -3 dB Position. The second **position** provides a total acoustic **power** output which is down an **additional** 3 dB at 10,000 Hz. Many **listeners** would consider this the "normal" position.

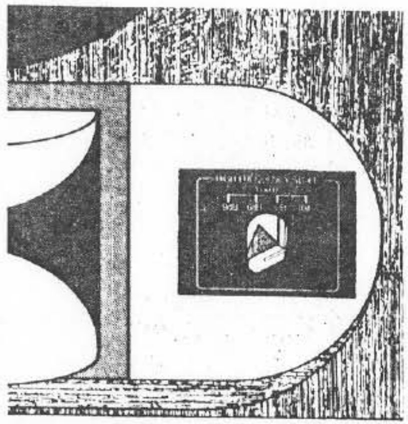
The -3 dB position is useful in treating recordings of high quality except for excessive "brightness" and/or audible tape hiss. Let your ears be the judge!

The -6 dB Position. The last setting attenuates the high frequencies by 6 dB at 10,000 Hz, resulting in a sound character similar to the "duller" high-fidelity systems available. This setting produces the most listenable results from program sources having high levels of distortion and/or noise, perhaps in addition to restricted high-frequency content.

Input Switch
The Input switch chooses the desired input to the equalizer. In the Source position, whatever program material is coming from the Tape-Out jacks of the amplifier will continue through to the balance of the system. When Tape is selected, a tape machine connected to the Tape-In jacks will play.

USING THE SPEAKER HIGH FREQUENCY SLOPE CONTROL

In normal operation, the High Frequency Slope control should be set on the flat position, with high-frequency response controlled at the equalizer (see "Using the Equalizer Controls" section). However, the High Frequency Slope control may be used to replace or augment the roll-off provided by the equalizer. The High Frequency Slope control is located behind the speaker grille. The grille may be pulled off its four dowell retaining pins for easy access to the control.



The response characteristics of the High Frequency Slope control closely parallel those of the equalizer, with attenuation increments of 3 dB at 10,000 Hz. Just keep in mind that if both controls are set to attenuate, their combined effect is additive. For example, if both the equalizer and the speaker controls are set at -3 dB, the combined effect is -6 dB, just as if either one of the controls had been set at -6 dB.

USING SYSTEM TONE CONTROLS

Generally speaking, the more sophisticated the listener, the less he uses tone controls. The equalizer contouring is an integral part of the total system design of the Interface:D, providing the proper compensation to achieve the performance goals. While the same degree of control flexibility used with conventional speaker systems may be employed, we suggest that minimal use of Loudness, Bass, Treble, and the like, will result in better overall performance.

EXTENSION SPEAKERS

Other Interface:D Systems
Once the equalizer is connected into a stereo system, additional Interface:D systems can be connected to the same amplifier and will receive the proper frequency contouring. Interface:D speaker systems are available without equalizer on special order from your dealer.

Conventional Speakers
If conventional speakers are connected to the same amplifier, they will also receive the contoured signal. However, the degree of contouring is minimal compared to some other available equalizers, so that other speaker systems will not be unduly endangered, especially if high listening levels are avoided. (The equalizer sends about four times normal power to the extension speakers in the region of peak boost, around 32 Hz). The low-frequency cutoff below 28 Hz should benefit most systems.

Interface: 1, 2, and 3 (Unequalized)
The unequalized Interface:1, 2, and 3 may be appropriately used as extension speakers. Follow the recommendations in the "Conventional Speakers" section, above.

Other Interface Models (Equalized)
Interface:C Interface:D and Interface:C use the same equalizer. Therefore, in a music system set up for Interface:D's, Inteface:C's may be added to the same amplifier and will receive the proper frequency contouring. Interface:C speaker systems are available without equalizer on special order from your dealer.

Interface:A Series II and Interface:B Series II. Not unexpectedly, the capabilities of these other equalized Interface models are best realized by using the equalizer specifically designed for each system. The deviation from ideal depends on the particular model.

The Series II Interface:A and B are not endangered by the Interface:C/D equalizer as long as very low bass (around 30 Hz) at the highest listening levels is avoided. For the Interface:A Series II, frequency response in the 40-80 Hz range will be depressed about 1.5 dB and approximately 2.5 times normal power will be fed to the woofer between 20 and 30 Hz. For the Interface:B Series II, the 40-80 Hz range will be depressed less than 1 dB and approximately 1.5 times normal power will be fed to the woofer between 20 and 30 Hz. For both systems, the Interface:C/D equalizer will provide a 3 dB under-emphasis at 10,000 Hz when the High Frequency Slope control is in the 0 dB position, equivalent to the response obtained when the correct equalizer is used in the -3 dB position.

If you have a question about a specific combination, do not hesitate to contact Electro-Voice.

AMPLIFIER POWER AND SOUND PRESSURE LEVEL RECOMMENDATIONS

Casual discussions of amplifier power requirements usually result in a wide range of "answers." This is so because power levels vary *immensely* with speaker efficiency, room acoustics, and desired listening levels. Nevertheless, the following commentary should help produce an answer that is right for you.

Fortunately, the bulk of the discussion can be simplified by fixing some of the variables. First, the efficiency of the Interface:D has been assumed. Second, the recommendations are based on a listening room of average acoustics (a precise description of these "average acoustics" follows in the section entitled "Room Acoustics and Amplifier Power"). With these variables fixed, we must deal only with the question of appropriate listening levels and the amplifier power required for these levels.

Understanding The Recommendations
The "Detailed Specifications Summary" section of this manual specifies the rated amplifier power requirements for a broad range of listening levels. The text of recommendations which follows illustrates these specifications and is designed to relate them to the real-world listening experience of the audio enthusiast.

Notes on Amplifier Power Ratings. All amplifier power recommendations are given in average sine wave watts (sometimes called "RMS" or "continuous" watts) per channel, all channels operating, over a minimum frequency range of 28 to 15,000 Hz. Common deviations from this rating method will not change attainable listening levels significantly. Also, to simplify the discussion, it is assumed that the amplifier is well behaved when operating at or slightly beyond its power output capability.

Notes on Listening Levels. All listening levels are expressed as sound pressure levels in decibels (dB). The dB is a term frequently used in audio but often misunderstood. For example, very few people have a real conception of what a "90 dB" sound pressure level sounds like. We hope to clarify this situation. Furthermore, the audible effect of specific increases and decreases in sound pressure level are not commonly known. A 1 dB change in overall program level is just audible to the critical ear. A 3 dB change is noticeable, but would be interpreted as only a modest change in level. Yet a 3 dB level increase requires a *doubling* of amplifier power output. A 6 dB change in level would seem fairly substantial; such an increase requires four times amplifier power.

Notes on Listening Position. The sound pressure levels noted below are those observed when the listener is in the "reverberant field" of the listening room. Sound pressure levels are highest very near the speakers. As the listener moves away from the speaker, the sound pressure level drops, as would be expected. However, in a room with average acoustics and with the Interface:D, this drop in level *stops* at about 6 feet from the speaker. Beyond this distance, the listener is in the reverberant field and the sound pressure level of wide-range program material remains virtually constant because nearly all of the audible sound energy is reflected energy.

Minimum Recommendations

Sound Pressure Level. It has been said that a sound pressure level of 85 dB is the maximum average intensity people want to experience in their homes.[1] However, it is our considered opinion that a quality music system should be able to provide a long-term average

program level of approximately 90 dB in order not to be found inadequate by nearly every serious music listener at some time. It is this so-called long-term average level that the ear interprets as a given loudness over any several-second musical crescendo. Also, it is this average level that is expressed by the relatively slow-moving indicator of a sound pressure level meter.

Additionally, we feel that a music system must be able to reproduce short-duration peaks (on the order of 10 milliseconds) 10 dB higher than the average, or 100 dB. Musical signals are full of such peaks. While they contribute very little to perceived loudness, they are essential to accurate reproduction.

A 90 dB average level will seem quite loud to many people, certainly far above a background music level (60 dB) or the level of ordinary conversation (65 dB). A 90 dB average level is very likely to represent a practical upper limit-of-pleasure for many commercial recordings where compression and background noise have compromised the integrity of the original signal.

Amplifier Power. With the Environment control in the 1/4 Space position, one Interface:D speaker system will produce the sound pressure levels described above with a 1.5 watt amplifier. The 90 dB average level is reached with an input of only .15 watts, with the full 1.5 watts producing the 100 dB instantaneous peaks. (With the Environment control in the 1/2 Space position, power requirements are doubled).

This amazingly modest requirement is due to the Interface:D's extremely high efficiency, about 10 dB higher than typical acoustic suspension speaker systems. This means that the Interface:D's power requirement is *one-tenth* that often recommended for these acoustic suspension systems.

Typical Requirements

Sound Pressure Levels. Although the 90 dB minimum recommended average level capability will satisfy a broad range of listeners, this level falls far short of the levels associated with most live music. Many enthusiasts will find live music levels enjoyable for the highest quality commercial program sources and well-executed live recordings. For example, while "loud" classical music reaches only the relatively modest level of 80 dB, "very loud" classical music goes well beyond the 90 dB "minimum" – ranging from 90 to 100 dB. The short-duration peaks required for realistic reproduction can be another 20 dB higher. Loud rock music is on the order of 115 dB average level.[1] It is therefore our opinion that many enthusiasts will find average level capabilities in the 95-100 dB range most appropriate. This means that the sound pressure levels of most live classical music can be attained. For contemporary rock and electronic music, the 95-100 dB capability represents a reasonable compromise among several variables: the actual levels of live rock, typical program sources, and neighbors.

Amplifier Power. With the Environment control in the 1/4 Space position, one Interface:D speaker system will produce the sound pressure levels noted with amplifiers ranging in capability from 4.7 (95 dB average level) to 15 watts (100 dB average level). Such levels would require amplifiers from 43 to 140 watts for typical acoustic suspension designs. The long-term average levels are produced by .47 to 1.5 watts with only the instantaneous peaks utilizing the full capacity of 4.7 to 15 watts. (With the Environment control in the 1/2 Space position, power requirements are doubled).

Maximum Recommended Power
Amplifiers much larger than the minimum recommended may be used: up to around 500 watts per channel. However, care and intelligence are required to see that the high power is used *only* to reproduce the harmless, short-duration program peaks that are 10 to 20 dB above the average levels. When this condition is fulfilled, the long-term average power delivered by the amplifier will be within the Interface:D's ratings: 50 watts minimum from 28 to 6,000 Hz, dropping gradually above 6,000 Hz to 10 watts at 10,000 Hz, and to 5 watts at 20,000 Hz. (Note that the Interface:D's power capacity is fully twice that of the usual speaker system; at mid and high frequencies, the advantage is *four* to *eight* times). The required condition is virtually assured if the signal from the amplifier is distortion-free and accidental inputs are avoided. Damaging accidental inputs include insertion or unplugging of the power cord or audio connectors while the amplifier is operating and dropping the phonograph pickup arm on the record surface under similar volume conditions. Note that the Interface:D's integral Tweeter Protector virtually eliminates damage from high-frequency amplifier distortion components and accidental inputs such as rewinding a tape recorder without tape lifters with the volume at normal playback levels (see the "Speaker Protection at High Listening Levels" section).

Sound Pressure Level. With a 500-watt amplifier and the Environment control in the 1/4 Space position, a single Interface:D will produce an average mid-band level of 115 dB. (In this context, "mid-band" refers to the frequency range from about 100 Hz to 1000 Hz, where most program material is concentrated). This maximum level capability is 10 to 15 dB higher than the usual high-fidelity speaker, so that even rock music can be reproduced at live-performance levels. Also, the Interface:D's maximum output ability above 1000 Hz is far greater than most speakers (110 dB up to 6,000 Hz; 103 dB at 10,000 Hz), so that the demanding high-frequency energy content of many contemporary studio recordings poses no problem.[2]

Room Acoustics and Amplifier Power

Description of "Average Listening Room." The professional acoustician would describe the average listening room used in the preceding discussion as a reverberant space having a room constant (R) of 200 ft.^{2} This specification is a direct function of the room's surface area in square feet and the average percentage of sound energy absorbed by the room's surfaces and furnishings. For illustrative purposes, a room constant of 200 ft^2 would result from the following specifics:

1. "Average" sound absorption (average absorption coefficient = .15). This would be provided by plaster ceiling and walls, carpeted floor, some draped surfaces, and typical soft furniture.
2. "Average" surface area (about 1100 ft^2) such as would result from a 19 ft x 15 ft room with an 8 ft ceiling.

How Amplifier Requirements Vary with Room Acoustics. Room acoustics affect the amplifier power required to achieve a given sound pressure level. Rooms larger than the average room have a larger surface area and thus require more amplifier power. Smaller rooms require less power. Rooms with more sound absorption than our average room (with "dead" acoustics) require more amplifier power. Rooms with less absorption ("live" acoustics) require less power.

A really complete treatment of the effects noted above cannot be given here. However, some examples will be useful in providing general guidelines:

1. A 10 ft x 20 ft x 30 ft "large" room with average absorption will require approximately twice the amplifier power as the average room.
2. A "medium-live" (average absorption coefficient = .1) room with the same dimensions as the average room will require approximately 40% less amplifier power.
3. A "medium-dead" room (average absorption coefficient = .25) with the same dimensions as the average room will require approximately twice the amplifier power.

SPEAKER PROTECTION AT HIGH LISTENING LEVELS

The Interface:D provides a unique combination of high conversion efficiency and high input power capacity that extends well into the high-frequency range. This combination permits operation at high levels in large listening rooms with less-than-usual concern for the safety of individual speaker components. However, the following information will be useful for the most demanding playback situations.

Integral Protection
Each Interface:D speaker system incorporates a TS1 Tweeter Protector. The tweeter is continuously monitored by a solid state full wave bridge rectifier. The rectifier drives a relay switch in the tweeter circuit which restricts power to the tweeter when the input exceeds a safe level. Operation is restored when levels are again safe.

When the Tweeter Protector is activated, a light bulb is switched into the tweeter signal path, and it's glow is visible on the front of the enclosure. The resistance of the bulb cuts tweeter power to a safe level and the glowing of the bulb gives positive indication of the TS1's operation. Because the tweeter is never completely off, the audible effect of the Tweeter Protector is small.

Tweeter power handling characteristics are such that their short-term power capacity (10 milliseconds or so) is about 10 times the long-term average. This compliments the nature of program material, since recordings usually have short-term peaks about 10 dB above the average levels. Although these peaks contribute little to perceived loudness (this is more directly related to the long-term average levels), the peaks are necessary for high-accuracy reproduction. Most speaker protective devices do not deal effectively with the peak-to-average characteristics of music, but the TS1, tailored exactly for the nature of program material, is a significant exception. The time constant of its rectifier circuit has been chosen to pass large signals of short duration (less than 10 milliseconds) while interrupting longer signals at much lower input levels.

When to Expect Tweeter Protector Operation
The Interface:D is highly unusual in that it's high-frequency power capacity is for the most part equal to or greater than that of a typical woofer (minimum 50 watts long-term average to 6,000 Hz, dropping gradually to 10 watts at 10,000 Hz). This high output ability at high frequencies means that the Tweeter Protector will be rarely activated, even with accidental inputs. A listing of three situations which may operate the Tweeter Protector follows.

Accidental High-Level High-Frequency Inputs. Very large high-frequency inputs are generated by rewinding a tape recorder without tape lifters with the volume at normal playback levels or by oscillations from a defective electronic component or system hookup.

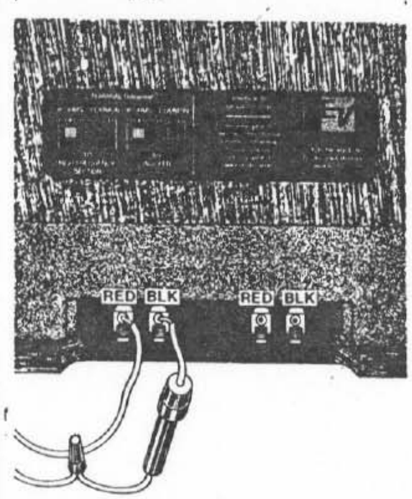
High-Level Reproduction of Program Material with Higher-Than-Usual High-Frequency Energy. The symphony orchestra, voice, and even most rock music have most of their energy concentrated below 1,000 Hz, and 5,000 Hz levels are at least 10 dB down. Thus, if 20 watts are delivered to the woofer, only 2 watts reaches the tweeter. However, many recordings of contemporary music (electronic synthesizers, close miked brass and percussion) have energy levels that are nearly uniform to beyond 5,000 Hz.[2] Such recordings will activate the Tweeter Protector at somewhat lower overall sound levels than will more traditional music.

High-Level Reproduction with Under-Powered Amplifiers. Your ears interpret the long-term average sound level as "loudness." In a typical living room, a single Interface:D can produce a 115 dB average level with a 50-watts-per-channel amplifier. However, the instantaneous program peaks which are typically 10 dB above the average will be "clipped" off as the amplifier runs out of power. Such clipped program material is rich in high-level high-frequency distortion products that are fed directly to the system tweeter, perhaps activating the Tweeter Protector.[2] A 500-watts-per-channel amplifier would actually be safer to use because peak clipping would be eliminated. (Note that even when peak clipping does not actuate the Tweeter Protector – the most usual case with the Interface:D, it produces a rough, raucous sound that is often mistaken for "speaker distortion" actually, the speaker is only faithfully reproducing the high-frequency distortion components in the "clipped off" input signal).[3]

Fusing for Full-Range Protection
Fusing should rarely be required. However, when the possibility of carelessness or inexperience is combined with high listening levels and large power amplifiers (say, in excess of 150 watts per channel) it is wise to fuse each speaker system. This

will prevent inadvertent damage to system low-frequency components. A Littelfuse brand 3AG "Slo-Blo" fuse of 1¼-ampere rating is recommended for each Interface:D speaker system. This particular fuse has been found to have a good current-versus-time characteristic, allowing higher (yet satisfactorily limited) current for relatively short periods of time and increased protection for more extended periods.

A fuse should be inserted in one of the speaker leads feeding each system. Inline fuse sockets may be used, or a fuse block may be glued to the rear of each enclosure. Both types of holders are readily available. A typical fuse installation is shown below:

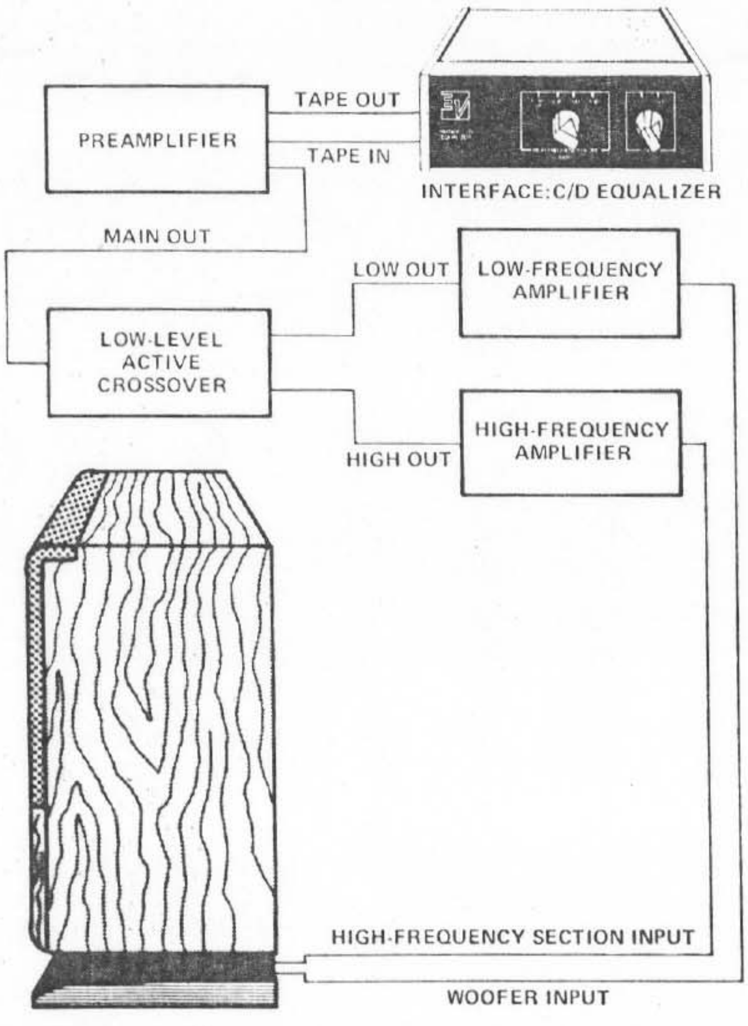


Protection Limitation
Keep in mind that any speaker protection system is a tradeoff between two extremes: guaranteed protection and high listening levels. We feel that our recommendations do not excessively limit listening levels, yet provide a very reasonable assurance of protection. As a result, however, there are conceivably some program materials that fail to actuate the protection circuitry yet result in speaker damage. Our experience indicates that such damage should be rare.

BI-AMPLIFICATION
The Interface:D may be bi-amplified at the 350 Hz woofer-to-midrange crossover by using an external low-level crossover and an additional channel of amplification. A simplified block diagram is shown on Page 18, one channel only. The equalizer is shown in the usual tape monitor connection, but it may be inserted elsewhere in the system as long as it is *before* the input to the low-level active crossover. See the "Connecting the Equalizer" section for variations.

Changes for Bi-Amplification
The Interface:D's internal crossover is modified for bi-amplification by substituting a special 9-pin "bi-amp" plug for the "normal" plug usually installed in the crossover. The bi-amp plug is contained in a clear plastic bag fastened to the rear of the enclosure. The bi-amp plug may be positively identified by its two wire jumpers; the normal plug contains four jumpers. The plugs are changed as follows:

1. Place the speaker enclosure on its back (a carpeted or other soft surface will protect the painted finish).
2. Remove the normal plug from the crossover. It is clearly visible through a hole in the black-painted enclosure base.
3. Install the bi-amp plug and right the enclosure.



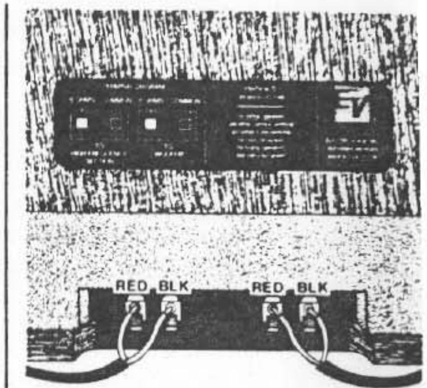
Speaker Input Connections and Crossover Characteristics
With the bi-amp plug installed, one pair of speaker input terminals is for the woofer and the other pair is for the high-frequency section. The terminals are positively identified by a label on the rear of the enclosure.

The crossover frequency should be 350 Hz with either a second-order Butterworth (12-dB-per-octave slopes)

or third-order Butterworth (18-dB-per-octave slopes) response characteristic. For flattest response in the crossover region when a second-order crossover is used, the polarity between the woofer and high-frequency sections should be the same, i. e., the positive output terminal of the low-frequency amplifier should go to the positive woofer input on the Interface:D and the positive output terminal of the high-frequency amplifier should go to the positive terminal of the high-frequency input on the Interface:D. This instruction assumes, of course, that there is no polarity shift between the high- and low-frequency amplifier channels. This condition is fulfilled if the two channels of a stereo amplifier are employed. *When a third-order crossover is used, the polarity between the woofer and high-frequency sections should be reversed.*

Level Sets
When a bi-amplified Interface:D is placed in a quarter-space environment (see "Using the Environment Control" section), the Environment control should be set in the 1/4 Space position and power amplifier and low-level crossover controls (if any) should be set for equal gain.

If a bi-amplified Interface:D is placed in a half-space environment, the required high-frequency section attenuation (3 dB) may be effected by switching the Environment control to the 1/2 Space position. However, amplifier power will be used more efficiently if the attenuation is obtained by lowering the gain of the high-frequency amplifier (or high-frequency output of the crossover) by 3 dB and leaving the Environment control in the 1/4 Space position. The 3 dB attenuation may be evaluated by ear, although the most accurate setting will be obtained if the system is driven with a sine wave whose frequency is above the 350 Hz crossover (say, 1000 Hz) and the gain reduction is measured with an appropriate AC voltmeter placed on the output of the high-frequency power amplifier.



Amplifier Power Division
Because the Interface:D's woofer-to-midrange crossover is relatively low (350 Hz) an even split in amplifier power is appropriate for most program material. For example, if a 100-watt amplifier is used for the woofer section, a 100-watt unit would be good for the high-frequency section. The two channels of a stereo power amplifier are convenient to use.

Advantages of Bi-Amplification
Although it is possible to discuss the advantages of bi-amplification in a general way, specific performance improvements are very much related

to the particular speaker components involved. The following comments will help you make a decision on bi-amplification of the Interface:D.

Elimination of Crossover Component Losses. Nearly all speaker systems utilize capacitors and inductors in an internal crossover network to limit the range of frequencies delivered to each transducer. In the Interface:D, such components are also used to equalize the frequency response of each individual speaker component, providing the uniform overall system response desired. Some crossover components are typically in series with the speaker components, i. e., between the amplifier output and the component input terminals. The resistive losses associated with real-world crossover components serve to separate, or de-couple, the speaker from the amplifier. This separation reduces the amplifier's inherent control of the speaker's motion by reducing the high damping factor associated with good amplifier design. (The small speaker connecting wire sometimes used by audio enthusiasts has a similar effect; see the "Connecting the Speaker Systems" section of this manual). Crossover component losses are inherently higher for lower crossover frequencies, because of the larger component values required. The losses are neglible for higher frequencies (such as the Interface:D's midrange-to-tweeter crossover at 3000 Hz).

The Interface:D's low-frequency crossover utilizes specially designed iron core inductors developed for low resistive losses. The core substantially reduces the amount of wire required for the desired inductance, keeping resistive losses low. The core material, mass, and configuration have been chosen to avoid the distortion-producing saturation associated with typical inexpensive iron-core inductors. Because of the high-frequency crossover, the usual loss-eliminating advantage of bi-amplification is significantly reduced, to the point that no audible or measurable performance changes should be expected.

Reduced Audibility of Amplifier Overload. Bi-amplification of the Interface:D will reduce the audibility of power amplifier overload, or clipping. For example, if program material in the woofer range clips a power amplifier loaded by a conventional full-range speaker system, high-frequency distortion products are sent to system midrange and tweeter components, where they are reproduced with unpleasant clarity. If the system is bi-amplified, however, the distortion products are not so clearly reproduced. Instead, they are fed only to the woofer, whose response falls rapidly at high frequencies.

Bi-amplification is also advantageous if the clipping occurs above woofer crossover. For example, in a normal full-range speaker system, midrange clipping produces intermodulation distortion products related to the low-frequency program material. In the bi-amplified case, this intermodulation is non-existant, since the low frequencies are reproduced by another amplifier.

Reduction of Intermodulation Distortion. Even when amplifiers are operated within their output capabilities, bi-amplification can reduce distortion. For instance, intermodulation products are generated when an amplifier is called upon to reproduce low and high frequencies simultaneously. To the extent that the frequency range handled by each amplifier is restricted by bi-amplification, intermodulation distortion is reduced. The intermodulation distortion of good amplifiers is quite low, however, so the effect of bi-amplification is likely to be inaudible or, at best, very subtle.

References:

1. For a lucid treatment of live-music sound pressure levels and perceived loudness: C. Stark, "The Dynamic Range of Music," **Hi Fi/Stereo Review**, June, 1968, and C. Stark, "The Sense of Hearing," **Stereo Review**, September, 1969.
2. For a good introduction to the frequency distribution of various program materials: L. Feldman, "Mystery of the Failing Tweeters," **Radio-Electronics**, October, 1976.
3. For an excellent introduction to the audible effect of peak clipping: R. Allison, "Loudspeaker Power Needs," **Stereo Review**, September, 1973.

DETAILED SPECIFICATIONS SUMMARY

The following specifications summary is extraordinarily complete by industry standards, complete enough to warrant some explanation. "Specs" are very popular with audio enthusiasts. They are often treated by manufacturers as a major ingredient in a successful advertising program. Other firms ignore specifications altogether, on the basis that no industry measurement standards exist and, if they did, could not begin to predict how a speaker will sound in your listening room.

At Electro-Voice, we subscribe to neither the "advertising" or "ignore-them" approach to specifications. Our design goal is to design an *accurate* transducer, one which changes the input as little as possible, which makes efficient use of amplifier power, which reproduces the dynamic range and sound pressure levels of live music when called upon to do so, and which sacrifices no single important performance characteristic for another.

The most valuable tool we have to assess our pursuit of accuracy is our ears. We listen to music, live and recorded, and we try to make our speakers *sound like music.* But our listening-based pursuit of accuracy is very much aided by a host of objective measurements. While even a state-of-the-art complete set of measurements cannot predict how a speaker will sound, such measurements are immensely helpful in explaining *why* a speaker sounds as it does, so we can do something about it if necessary.

With this background, we present to you the following detailed specifications summary. A truly complete set of measurements and appropriate commentary is not possible here; some measurements would have to be presented in graphical form to be most meaningful. Nonetheless, you will find these specifications an indication of our commitment to accurate reproduction as well as a useful shorthand guide to what performance you may expect in your listening room.

INTERFACE:D DETAILED SPECIFICATIONS

EACH SPEAKER SYSTEM

(Equalizer and System High Frequency Slope controls at 0 dB, Environment control set for quarter space)

Frequency Response, 1 Meter on Axis, Swept One-Third Octave Random Noise, Quarter-Space Anechoic Environment
± 3 dB 28-18,000 Hz

Total Acoustic Power Output vs. Frequency, Quarter-Space Anechoic Environment
± 3 dB 28-18,000 Hz

Low-Frequency Acoustic Power Output vs. Frequency (Below 100 Hz), Small Signal
3-dB-Down:
28 Hz
10-dB-Down:
23 Hz

Dispersion Angle Included by 6-dB-Down Points, Horizontal Plane, Indicated Bands of Random Noise, Anechoic Environment
500-1000 Hz Octave Bands:
170° ± 5°
2000-16,000 Hz Octave Bands:
115° ± 10°
16,000 Hz 1/3-Octave Band:
110°

Dispersion Angle Included by 6-dB-Down Points, Vertical Plane, Indicated Bands of Random Noise, Anechoic Environment
500-1000 Hz Octave Bands:
170° ± 15°
2000-8000 Hz Octave Bands:
90° ± 10°

Sound Pressure Level at 1 Meter, 1 Watt into Nominal Impedance, 300-10,000 Hz Average, Anechoic Environment
97 dB

Suggested Amplifier Power Ratings, Continuous Average Power per Channel at 8 Ohms (Long-Term Average Power Capacity not to be exceeded)
Minimum:
1.5 watts
Typical:
4.7 to 15 watts
Practical Upper Limit:
47 watts
Maximum:
500 watts

Long-Term Average Sound Pressure Levels, with Instantaneous Peaks 10 dB above Average, at Midband Frequencies (100-3000 Hz), in Reverberant Field of Typical Living Room (R = 200 square ft), at Indicated Watts per Channel Available
Medium/Loud (90 dB)
1.5 watts
Loud (95 dB
4.7 watts
Loud/Very Loud (100 dB)
15 watts
Very Loud (105 dB)
47 watts
Too Damn Loud, Max. (115 dB)
500 watts

Maximum Long-Term Average Sound Pressure Levels, with Instantaneous Peaks 10 dB above Average, at Indicated Frequencies, in Reverberant Field of Typical Living Room (R = 200 square ft), with Maximum Long-Term Average Power Applied
Midband (100-3000 Hz):
115 dB
10,000 Hz:
103 dB

Long-Term Average Power Capacity at 8 Ohms
28-6000 Hz:
50 watts minimum
6000-10,000 Hz:
50 watts, dropping to 10 watts at 10,000 Hz
20,000 Hz:
5 watts

Short-Term Power Capacity (10 ms) at 8 Ohms
28-6000 Hz:
500 watts minimum
6000-10,000 Hz:
500 watts, dropping to 100 watts at 10,000 Hz
20,000 Hz:
50 watts

Quarter-Space Reference Efficiency
3.0%

Maximum High-Frequency Acoustic Power Output (10,000 Hz)
Long-Term:
.12 watts
Short-Term (10 ms):
1.2 watts

Maximum Midband Acoustic Power Output (100-6000 Hz)
Long-Term:
1.5 watts
Short-Term (10 ms):
15 watts

Crossover Frequencies
Acoustic:
40 Hz
Electrical:
350 & 3000 Hz

Impedance
Nominal:
8 ohms
Minimum:
5 ohms

Transducer Compliment
12-in dynamic woofer, downward facing
6-1/2-in vented midrange
Neckless radial horn tweeter

Controls
High Frequency Slope (0 dB, -3 dB, -6 dB, and -9 dB at 10,000 Hz) & Environment (quarter space/half space)

Bi-Amplification
A low-level, active crossover (12-dB-per-octave minimum slope) may be used in place of the integral 350 Hz crossover

Tweeter Protection
Integral TS1 time-variable turn-off circuit with indicator light

Dimensions
21-3/4 in wide X 32 in high X 15-1/2 in deep

Cabinet
Walnut veneer

Net Weight
114 lb

EQUALIZER

(Each channel, High Frequency Slope control at 0 dB)

Midband Gain
Unity

Maximum Equalization
6 dB at 32 Hz, fixed

Maximum Input Signal, RMS Sine Wave
Midband (80-3000 Hz):
7 volts
32 & 20,000 Hz:
3.5 volts

Noise Output, 20-20,000 Hz Bandwidth
80 dB below 200 millivolts

Total Harmonic Distortion
1.0 V RMS Input, 20-20,000 Hz:
.01% or less
3.5 V RMS Input, 20-20,000 Hz:
.05% or less
5.0 V RMS Input, 55-8000 Hz:
.05% or less

Intermodulation Distortion, 60 Hz and 7000 Hz in a 4:1 Ratio, 1.5 V RMS Equivalent Sine Wave Input
.005%

Channel Separation, Source Impedance Less Than 1000 Ohms, 20-20,000 Hz
60 dB minimum

Output Impedance
1200 ohms

Minimum Load Impedance
8000 ohms

Input Impedance
100,000 ohms

High Frequency Slope Control
0 dB, -3 dB, -6 dB at 10,000 Hz & power off

Power Requirements
110/120 volts, 50/60 Hz, 3 watts

Accessory AC Outlet
200 watts, unswitched

Dimensions
2 in-high X 8 in wide X 7 in deep

Net Weight
2 lb, 10 oz

Specifications subject to change without notice.

CUSTOMER SERVICE

Shipping Damage

Electro-Voice products are packed to provide protection well in excess of the shipping requirements of the Interstate Commerce Commission. Responsibility for delivery in good condition was accepted by the carrier, and therefore any damage claims must be made by the receiver against the carrier. If shipping damage has occurred, contact the carrier immediately, requesting inspection and instructions; or contact the dealer from whom the unit was purchased.

Reshipment

We strongly encourage you to retain all packaging materials for possible future use. Only original packaging materials are certain to provide full protection, whether used for units requiring service or simply for normal household moving. Bear in mind that a carrier can refuse a damage claim if *they judge* substitute packaging to be inadequate.

When reshipping by commercial carrier, the vent substitute must be lightened by removing the steel tube and end cap (see "Speaker System Assembly" section). *Shipping with the vent substitute weights installed virtually assures destruction of the vent substitute cone assembly. Such damage will not be repaired under warranty.* Note that when returning the Interface:D to Electro-Voice that the vent substitute's steel tube and end cap need not accompany the speaker systems. All required testing and service will be made using parts retained for the purpose at Electro-Voice.

When necessary, Electro-Voice can supply replacement packaging for a nominal charge. Contact the Service Department.

WARRANTY (Limited)

Interface:D is guaranteed against malfunction due to defects in workmanship and materials. If such malfunction occurs, Interface:D will be repaired or replaced (at our option) as follows:

> Speaker systems will be repaired or replaced without charge for parts or labor for a period of five years from date of original purchase.
>
> Equalizer will be repaired or replaced without charge for parts for a period of three years from date of original purchase and without charge for labor for a period of one year from date of original purchase.

All units must be delivered prepaid to the proper Electro-Voice service facility and will be returned prepaid. Warranty does not cover finish or appearance items or malfunction due to abuse or operation at other than specified conditions. Repair by other than Electro-Voice or its authorized service agencies will void this guarantee.

For instructions on return of Electro-Voice products for repair to the factory or authorized service agencies, please write: Service Department, Electro-Voice, Inc., 600 Cecil Street, Buchanan, Michigan, 49107 (Phone 616/695-6831) or 7473 Avenue 304, Visalia, CA 93277 (209/625-1330,-1).

Electro-Voice also maintains complete facilities for non-warranty service.

Part Number 53317–806

Litho in U.S.A.