the direct and delayed sound, the performer can play or sing a new note as the previous note or notes are just emerging from the delay. It may be impossible to tell which attack is the original.

Delay times should be chosen with care for any given piece of music because the delay must correspond to some subdivision or multiple of the basic tempo. By the same token, rhythmic effects tend to define their own tempos.

Sensitive experimentation can produce startling, intricate rhythmic patterns from very simple inputs.

Vibrato

Vibrato is the effect produced by small, regular variations in a sound's pitch; guitar vibrato, for example, can be created by alternately stretching and relaxing the strings with a special tailpiece.

The PCM 41 can create automatic, regular vibrato for any single instrument, or mix of sounds, by means of sine wave modulation in the VCO section. Manual inflections can be created using the DELAY MULTIPLY control, or an external foot controller connected to the VCO jack.

Short delays should be used with no feedback and 100% delayed sound fed to the output. Moderate modulation DEPTH is more natural sounding, especially with realistic RATE settings (both depend on the instrument and the music).

Pitch Twisting Effects

Pulling a string toward the edge of the fretboard will create pitch twisting (pitch bending or shifting) on a guitar or bass. The PCM 41 can twist the pitch of any instrument or vocal.

Slow sine wave modulation of the VCO produces upward and downward sweeps in pitch (similar to vibrato, but the rate is much slower). Square wave modulation produces a sequence of (1) the original pitch, (2) a raised pitch, and (3) a lowered pitch; this is often referred to as an arpeggio effect.

The PCM 41's envelope modulation adds another dimension to the pitch twisting effects normally available with a delay line. Setting the WAVEFORM control to ENV position (knob centered) causes the PCM 41's delay time to increase and decrease in proportion to the envelope of the audio input. ("Envelope" refers to the moment to moment changes in the overall signal level of a program.) The result is an articulated pitch sweep, one that opens up a whole new range of musical and special effects.

If the pitch shifted output is fed back to the input (using FEEDBACK), it will again be shifted. Thus a single note at the input may result in many different output pitches as that original note is delayed, altered in pitch, and recirculated.

Since it is impossible to explore the full range of pitch twisting effects, liberal experimentation is encouraged. The main parameters to be explored are VCO DEPTH, SHAPE, DELAY TIME, and FEEDBACK. (Manual pitch shifting is also possible by means of the DELAY MULTIPLY control or a foot controller connected to the VCO jack.)

Flanging

Originally, flanging effects were created by recording the same program on two tape recorders, and playing both back in synchronization while mixing the identical programs together. By slowing down one machine slightly, then the other, different phase cancellations would occur (this was done by using hand pressure against the flange of the tape reels, hence the origin of the term "reel flanging", or "flanging" for short). The result was a series of changing phase cancellations and reinforcements, producing a "comb filter" and the characteristic swishing, tunneling and fading sound.

Flanging with a delay line is much simpler; two tape machines need not be kept in relative sync. In fact, no tape machine is necessary – flanging can be done as a live effect. A short delay is mixed with the original signal, causing cancellation (nulling) at a frequency whose period (the time for one cycle) corresponds to twice the delay time. Cancellation also occurs on odd harmonics of that frequency. The depth of cancellation depends on the level balance of direct and delayed sounds; the "=" center detent on the OUTPUT MIX control sets the unit for a 50-50 mix, the setting at which maximum cancellation occurs.

NOTE: Since flanging relies on precise phase cancellations, it must be done by electrically mixing the input and delayed signal. It cannot be achieved in stereo. There is a special flanging effect which uses two delay lines (see Section 4.4).

The classic sweeping flange occurs when the delay time is continuously varied, causing the null frequencies to sweep. As these nulls (notches in response) sweep across the various components of the program input, different harmonics and fundamentals are boosted or cut in relation to each other; the tone of the input is thus caused to constantly change.

A further variation occurs if the polarity of the delayed signal is inverted with respect to the input signal. The DELAY INV button's effect is similar to a 180° phase shift, and hence the comb filter pattern is shifted. The effect is known as "negative flange."

The flanging effect can be further altered by recirculating the delay with the FEEDBACK control. Larger amounts of FEEDBACK cause exaggerated "deep" flanges (do not confuse this with use of the DEPTH control, which only causes a wider sweep of the nulls, but no greater depth). Inverting the polarity of the feedback with the FEEDBACK INV button gives a "hollow" characteristic to the sound.

Use of the VCO's envelope (ENV position on the WAVEFORM control) produces an interesting "talking flange"; sweep is produced for each attack of the input. ENV and sine wave can be mixed for a complex sweep which is really quite pleasing. Square wave modulation, while unnatural, may be useful to create an unusual effect.

Resonant Effects

Singing in the shower is one way to obtain a natural, though moderate resonant effect. Startling resonances are possible by using the PCM 41 at short delay times with a lot of FEEDBACK. This causes a build-up of fundamental notes and harmonics whose period (the time for once cycle) is equal to the set delay time. These emphasized pitches are said to "resonate." The effect can be characterized as adding a ringing, metallic quality to the sound. Extreme resonant effects create the "Cylon" voice of TV fame.

The pitch and tone of the resonance is affected by DELAY TIME, phase (INV), amount of FEEDBACK, and OUTPUT MIX. (Like the flanging effects, resonance is necessarily a mono effect.) Care must be exercised to keep feedback below the point where the unit will spontaneously oscillate (run away or howl). Use of the HI cut and LO cut filters allows greater amounts of FEEDBACK, in many cases, before howling occurs.

Long Delay Effects

When the delay time is equal to or greater than the time for a single beat of the music, it becomes possible to play counter points and harmonies against previously performed phrases. Careful synchronization of the performance and the delay time allows "polyphonic" sound from a single melodic line.

When FEEDBACK is used at long delay times, each sound entered at the input will repeat more than once before it fades out. Very full chords and textures can thus be built up; the chords will fade at a rate which depends on the DELAY TIME and the amount of FEEDBACK; 30 seconds or longer is possible – plenty of time for the performer to change instruments or leave the stage entirely!

Infinite Repeat (Repeat/Hold) Effects

The ultimate in sound-on-sound capability can be obtained by plugging a momentary footswitch into the PCM 41's rear panel REPEAT jack. Up to 800 milliseconds of sound may be "captured" in the delay line, then repeatedly fed to the output with no fading or degradation in quality.

To experiment with the effect, the delay is set for the longest possible time of 800 ms (DLY switch engaged to "X2" mode, DELAY MULTIPLY control set at full CW "X2" position, and 200/400 ms LED illuminated). FEEDBACK and VCO DEPTH should both be at "0".

The performer should play rhythmic or melodic phrases, setting OUTPUT MIX as desired, until he "gets the feel" of the delay time. At this point the unit is still not in REPEAT mode (REPEAT LED should not be illuminated). When ready, the performer continues playing and hits the foot switch once firmly, which should activate the REPEAT mode. A short phrase is now repeating. If OUTPUT MIX is adjusted to pick up the input as well, the performer can solo against the repeating phrase. To release the phrase and return to "normal," the foot switch is again actuated. Enter another repeat by hitting the foot switch once more. A little experience will lead to good control of the repeat process.

"Multi-tracking" is possible by bringing up the FEEDBACK control; this has no effect so long as REPEAT is on, but the moment REPEAT is released, the entered phrase will fade out rather than stop abruptly. With a slow enough fade (i.e., enough FEEDBACK), a new phrase can be entered in a subsequent REPEAT before the previous one has faded much. The two phrases are thus "layered" on top of each other, a pretty neat trick. The "layering" can be repeated indefinitely!

REPEAT mode offers still more possibilities. The pitch and duration of a "captured" segment can be altered. This is done by changing the delay time once the unit is in REPEAT mode (see note below). The pitch/duration change can be done manually using the DELAY MULTIPLY control, a foot controller connected to the VCO jack, and/or the DLY X1/X2 switch. Automatic pitch/duration changes can be achieved by using VCO modulation.

If the DELAY MULTIPLY control was centered (X1) when the REPEAT was initiated, then the pitch can be doubled and duration halved (X.5) or pitch can be halved and duration doubled (X2); if the control was at X2 to begin with, then pitch can be raised two octaves and duration cut to a fourth the original. The DLY X1/X2 switch can further shift the pitch up an octave and halve the duration (going from X2 to X1), or cut pitch in half and double the duration (going from X1 to X2). Thus a captured phrase of 800 ms can be shortened to 100 ms and raised 3 octaves; conversely, shorter segments can be lengthened and shifted down in pitch.

NOTE: The length of the repeated segment is dependent only upon the setting of the DELAY MULTIPLY control and the DLY X1/X2. The DELAY SELECT buttons and delay time LEDs are irrelevant to the length of the repeated segment, although when a "0" delay time is selected the repeat will be muted (it is still retained in memory, however, and will reappear when any other delay time is selected). The duration of the memory is always equal to the maximum DELAY SELECT setting (200 ms or 400 ms) times the DELAY MULTIPLY function. Thus, if DELAY SELECT is set at any time other than 200/400 ms, the system will "jump" to a longer time when REPEAT mode is initiated. For this reason, it is suggested that the DELAY SELECT control be set at maximum when playing with REPEAT effects, giving the player an accurate preview of the duration of the repeated segment.

4.2 FRONT PANEL SETTINGS FOR BASIC DELAY EFFECTS (PATCH DIAGRAMS)

The effects possible with the PCM 41 are literally unlimited. It is not simply a matter of the many combinations and permutations of control and switch settings; consider the fact that for any given setup, the audible effect may be completely different when the program source changes. Bear this fact in mind because a given "flange" or other setup may sound great with one instrument and awful with another. For example, some tunneling effects may be dramatic with vocals, yet scarcely noticeable with a guitar. For this reason, the patch diagrams that follow are intended only as suggested starting points. By all means experiment and derive your own favorite effects.

NOTE: In the following front panel drawings, darkened buttons are engaged (pressed in). Control settings are marked on the knobs except where the setting does not alter the basic effect (e.g., wherever VCO DEPTH is at "0", the WAVEFORM and RATE settings are irrelevant). Input Level is nominally shown in center position, although this must be set in accordance with the program level.

It is easy to see that, except for the longest delays, the same delay time can be achieved two ways: with DLY X1/X2 in X2 mode, or in X1 mode if DELAY SELECT button is incremented upward. Wider bandwidth and consequently better audio quality are obtained in X1 mode.

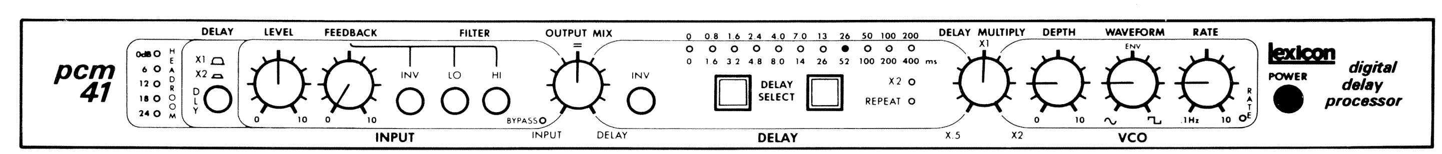


Figure 4-5 – Double Tracking
For stereo doubling, set OUTPUT MIX fully CW to "DELAY;" use the MAIN OUTPUT for one channel and the DIRECT OUTPUT for the other. For more or less pitch deviation, vary the VCO's DEPTH, WAVEFORM and RATE.

4.0 WHERE TO USE THE PCM 41

Within a given sound system, there are a number of different locations where the PCM 41 can be inserted. The examples shown below are representative of typical live performance and studio setups. Front panel settings to achieve specific effects are given in subsequent pages of this section.

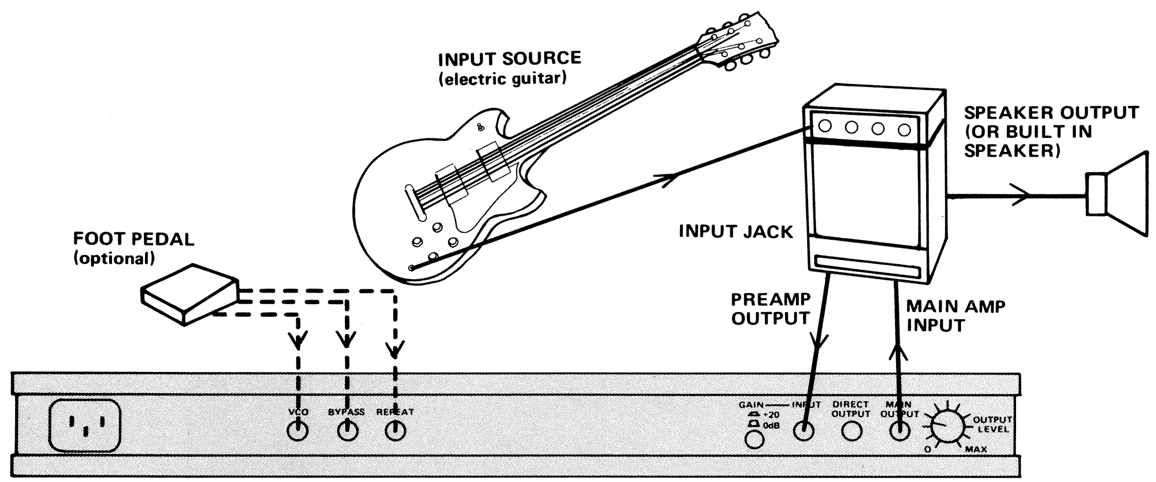


Figure 4-1 — Basic Mono Setup (with Guitar Amp)

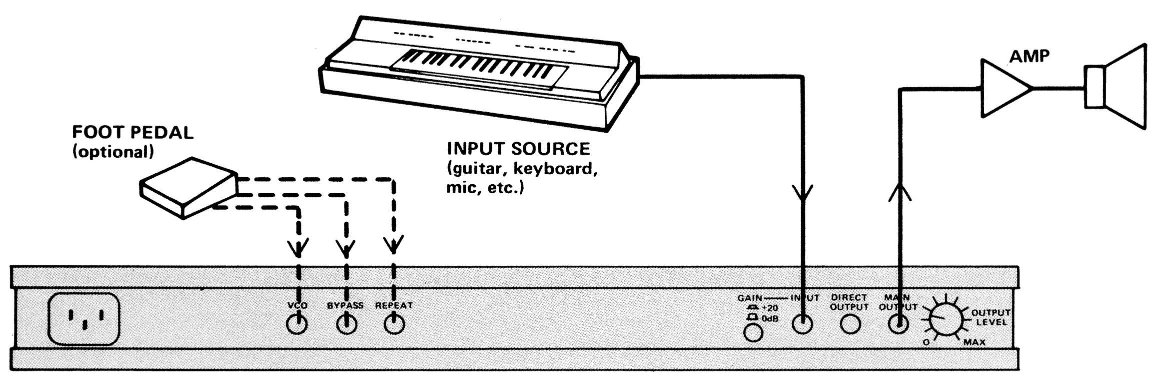


Figure 4-2 — Basic Mono Setup (with Low Level Input)

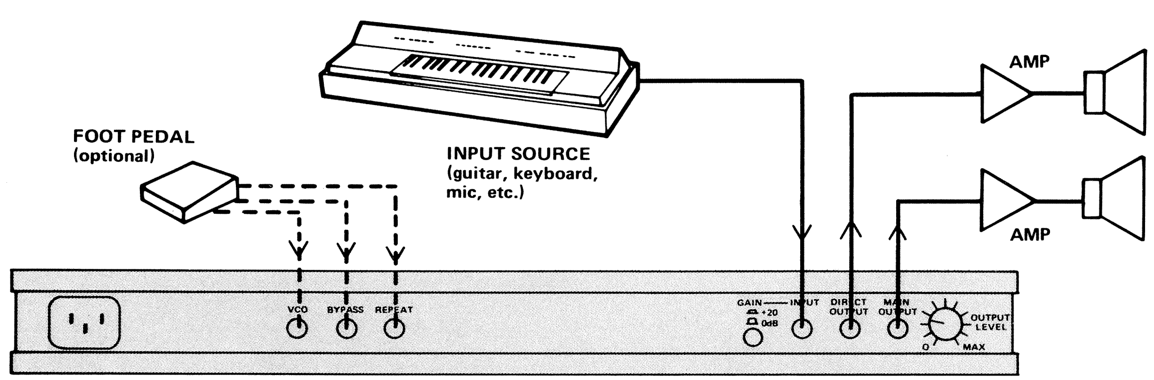


Figure 4-3 — Basic Stereo Setup

4.3 USING TWO PCM 41's FOR MORE COMPLEX SPECIAL EFFECTS

Figures 4-12 through 4-14 illustrate three different ways to use two PCM 41 delay lines in conjunction with one another, each setup having its own advantages. The actual front panel settings for any given effect can be derived from the basic information in Section 4.2; specific settings are shown for tape phasing effects in Figure 4-15.

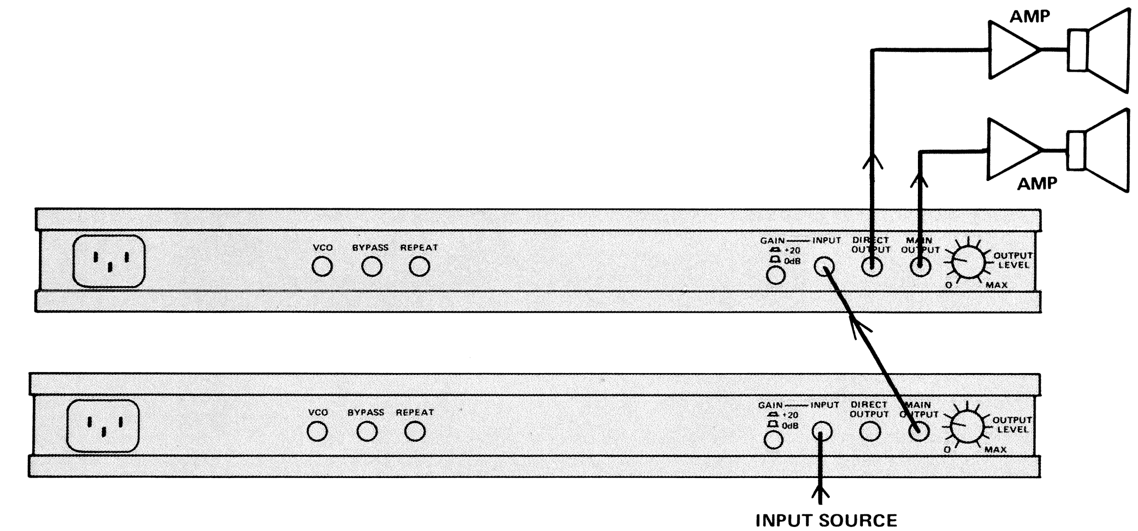


Figure 4-12 — Series Connection of Two PCM 41's

In this setup, processed sound from one unit feeds the second. Typical uses would be to chorus a sound and then add echo, or to pitch twist it and then flange it as well.

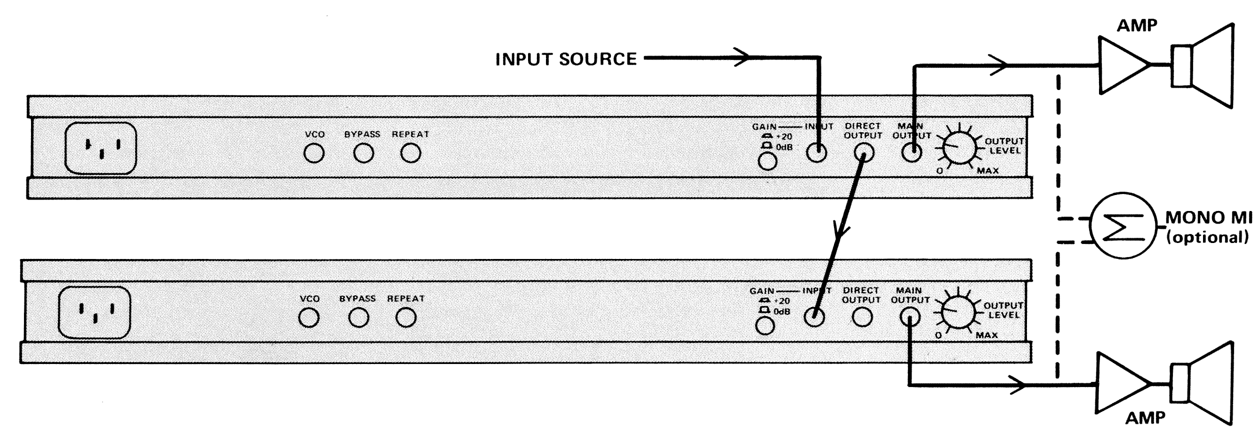


Figure 4-13 — Parallel Connection of Two PCM 41's

In this setup, the same unprocessed input source feeds both units' INPUTS. The MAIN OUTPUTs from both units are sent to a stereo recording or sound system, a good setup to use for stereo chorusing.

When the MAIN OUTPUTs are combined externally to mono, the setup becomes perfect for tape phasing effects that are deeper than regular flanging. (See Figure 4-15 and Section 4.4 for details of tape phasing.)

3.0 MOUNTING

The PCM 41 is designed to be mounted in a standard 19 inch relay rack, where it occupies one rack space (1-3/4″) and extends 11″ behind the front panel. If the unit is to be shipped in a rack, we suggest supporting the rear of the chassis to protect the delay line from vibration and mechanical shock.

The PCM 41 is fitted with rubber feet, and may therefore be rested on any flat surface. Whether in a rack or not, be sure to provide adequate ventilation around the top and bottom covers of the unit. In general, electronic equipment operates with greater stability and reliability when kept cool.

Since the PCM 41 processes low level audio signals, it is advisable to locate it away from strong electro-magnetic fields such as those generated by power transformers, motors, fluorescent ballasts, etc.

3.1 POWER REQUIREMENTS

The PCM 41 is set at the factory to operate from either 100, 115, or 230 volts (50 Hz to 60 Hz); the factory set voltage is indicated on the rear panel. The PCM 41 draws a maximum of 20 watts. Units set for 100 or 115 volts are fused at 1/4 ampere, while 230 volt units are fused at 1/8 ampere.

Be aware that voltages are normally set at the factory: they should not be changed except by a qualified service technician.

The power cord supplied with the PCM 41 utilizes a standard IEC connector, simplifying adaptation to power sources in various countries. The AC cord provides chassis grounding to the mains ground in order to comply with accepted safety standards (i.e., it is a 3-wire cord).

3.2 AUDIO SIGNAL CONNECTIONS

All jacks on the PCM 41 accept 1/4″ phone plugs. Some jacks accept only mono plugs (Tip-Sleeve), some stereo plugs (Tip-Ring-Sleeve), and one accepts both. Refer to this subsection to ensure you are using the proper plug and cable for a given connection.

Input

The audio INPUT jack is a balanced differential input, when used with a T-R-S phone plug (Tip = signal high, Ring = signal low, Sleeve = shield ground).

The audio INPUT jack can be used with unbalanced input sources when fed by a T-S phone plug (Tip = signal high, Sleeve = signal low). This creates a traditional single-ended input.

NOTE: If a T-R-S plug is used with a single-ended source, be sure to short the plug's ring and sleeve terminals together.

The PCM 41 has a high input impedance (40,000 ohms). As such, it can interface with sources having low or high impedance outputs.

Output

Both the MAIN and DIRECT audio outputs, are unbalanced (single-ended), and are intended for use with a T-S phone plug. Load impedances of 5,000 ohms or higher can be driven at full level; loads as low as 600 ohms are permissible, although the output level will be reduced slightly. If the PCM 41 is used to drive microphone-level inputs, it will be necessary to insert a 20 dB or 30 dB attenuation pad somewhere between the PCM 41 output and the low level input (preferably at the opposite end of the cable from the PCM 41).

NOTE: See Section Four for overall setup block diagrams. These illustrate where the PCM 41 is connected relative to the other components of the sound system.

3.3 OPERATING LEVELS

Input Levels

The PCM 41 is equipped to interface with a wide variety of input sources including microphones, electric instruments, mixer outputs, and so forth. Since the delay line input is high impedance, it will operate equally well with low or high impedance sources. It is important, however, to set the PCM 41 properly so that high level sources will not overload it, and low level sources will be processed with adequate signal-to-noise ratios.

For precise matching of the PCM 41 input sensitivity to the signal source, first set the rear panel 20 dB GAIN switch appropriately, then adjust the front panel INPUT LEVEL control for peak HEADROOM displays of 0 dB, with average levels around –12 dB.

Engage the GAIN switch (20 dB extra sensitivity) when using sources with peak levels ranging from –23 dBV to 0 dBV (0.07 V to 1.0 V rms). This includes electric pickups, high output microphones, low level mixer outputs, and some electronic instruments or instrument preamplifier outputs.

The GAIN switch should be out (unity gain) when using sources with peak levels ranging from –3 dBV to +19 dBV (0.70 V to 9.0 V rms). This includes high level mixer outputs, some electronic instrument/preamplifier outputs, and some direct box outputs.

When the signal level is not known, start at the 0 dB GAIN position (button out), and adjust the INPUT LEVEL to bring the HEADROOM display within the desired range. If there is inadequate gain to achieve peaks of –6 dB when INPUT LEVEL is at maximum, temporarily turn down INPUT LEVEL again. Then engage the GAIN switch to +20 dB position, and bring up INPUT LEVEL as needed.

If there is distortion on program peaks, then the INPUT LEVEL and GAIN combination is too sensitive (too high); reduce one or both. Conversely, if there is no distortion, but rather excess noise, sensitivity is too low. If the sensitivity is inadequate with the GAIN switch in +20 dB position and INPUT LEVEL at maximum, there may be a problem in the input cable, or the source may simply be too low in level.

WARNING

Never connect the PCM 41 input directly to power sources, improperly isolated AC/DC devices, or power amplifiers. Disregarding these precautions can lead to electrical shock, and may do permanent damage to the PCM 41.

Main Output Levels

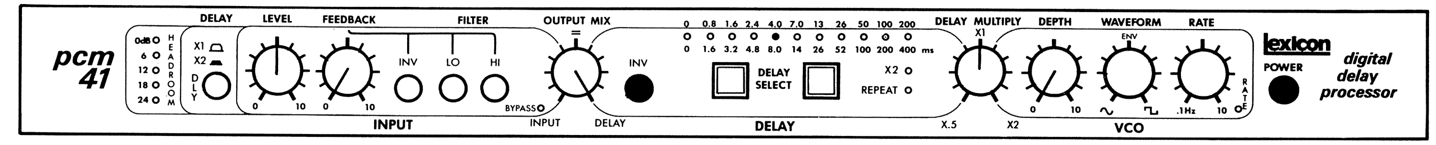
The PCM 41 MAIN OUTPUT is designed to drive 5k ohm or higher impedance loads at the full level of 9 volts rms (+19 dBV). It is possible to drive low impedance loads, down to 600 ohms, although maximum output level is reduced to some 5 volts rms (+14 dBV).

The INPUT LEVEL control should not be used to set output level; its purpose is to maintain good signal-to-noise ratio within the PCM 41, and to avoid distortion, by establishing operating ranges of –18 to –12 dB average, with peaks at 0 dB on the HEADROOM display.

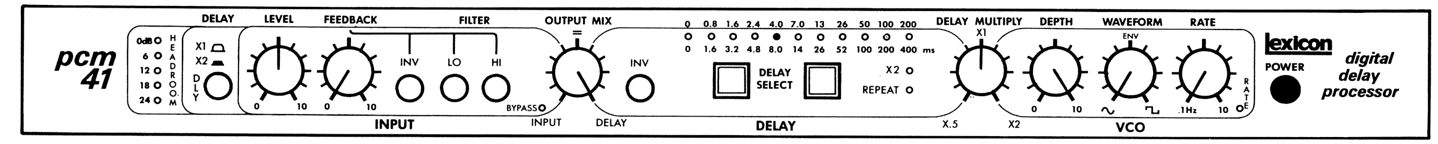
Once the proper input level is established, the rear panel OUTPUT LEVEL control can be used to adjust the MAIN OUTPUT level to the optimum value for driving a wide range of devices. A full CCW setting ("0") completely attenuates the output (kills the signal). In general, best signal-to-noise performance results at settings of mid-scale or higher.

CAUTION

The PCM 41 outputs are protected against short circuits, but should never be connected to signal sources.



Unit "A"



Unit "B"

Figure 4-15 – Tape Phasing Using a Pair of PCM 41's

In this setup, the unit "A" DIRECT OUTPUT feeds the unit "B" INPUT (i.e., the same input source feeds both units). The MAIN OUTPUTS of both units are externally mixed together; the connection is illustrated in Figure 4-13.

To create the effect, follow this procedure:

1. Set DELAY SELECT on both units to "0".
2. Apply an input signal, and adjust the balance of the OUTPUT LEVELS until the output disappears, or at least until the maximum cancellation and reduction of level occurs.
3. Set both units to the indicated delay time of 4.0 ms.
4. Adjust DELAY MULTIPLY on unit "A" until the sweep almost, but not quite, reaches absolute null.
5. Enjoy the sound. Try turning DELAY INV off on unit "A", and experiment with different delays.

4.5 USE OF THE LEXICON MODEL 93 WITH THE PCM 41

The PCM 41 makes an excellent companion for Lexicon's Model 93 PRIME TIME processor, allowing new possibilities and variations on established setups. Figures 4-16 through 4-18 illustrate but a few of the many conceivable interconnections. The creative performer or engineer is encouraged to experiment (the possible control setting combinations of the two units become far to numerous to cover here).

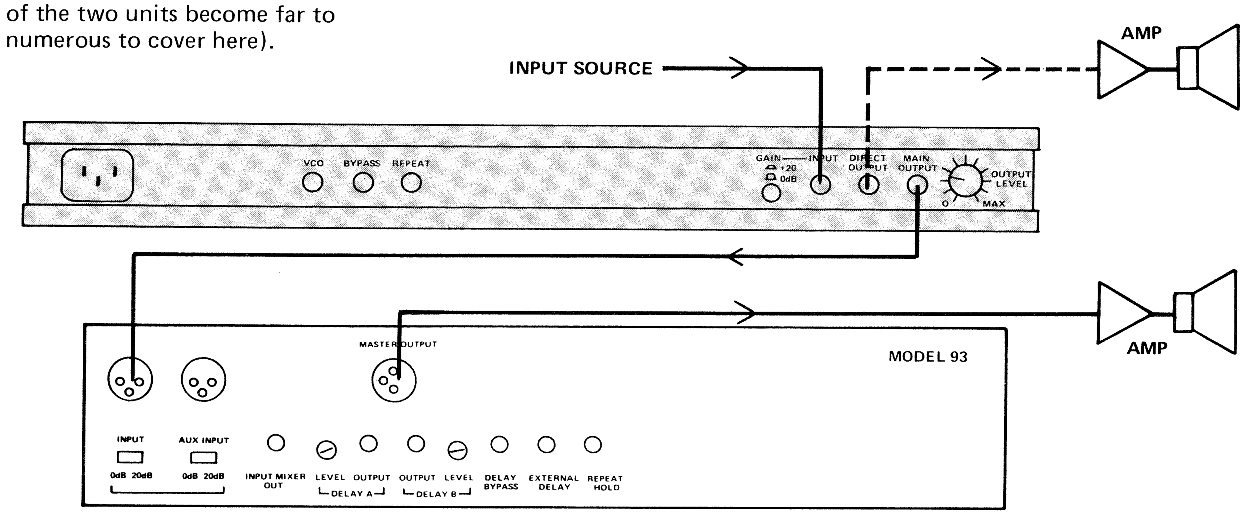


Figure 4-16 – Series Connection of PCM 41 and Model 93

Controls & Indicators

2.0 GENERAL

This section briefly describes the PCM 41's various controls, connectors and indicators. More detailed information about actual installation is contained in Section Three; applications suggestions are in Section Four. When levels are given in dBV, 0 dBV is referenced to 1.0 Vrms.

2.1 FRONT PANEL

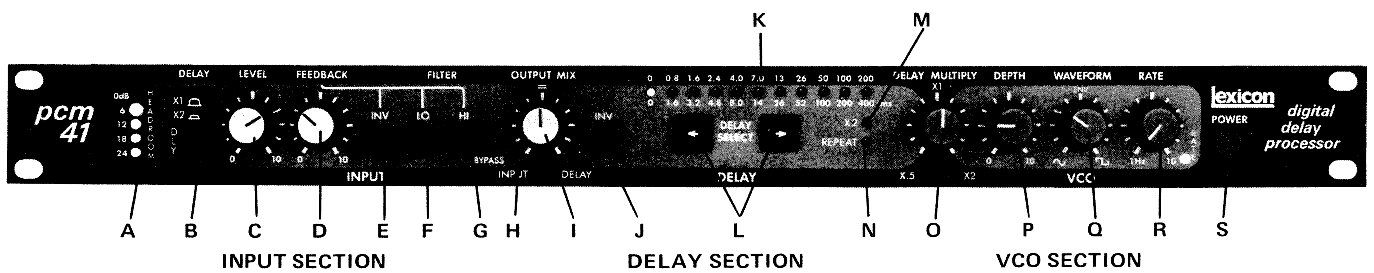


Figure 2-1 – PCM 41 Front Panel

A. Headroom Display
The HEADROOM display monitors the audio level after the INPUT LEVEL control. The display has 5 LED's which indicate peak levels, relative to system overload, from 0 dB (no headroom) to –24 dB (24 dB headroom remaining).

B. Delay Range (X1/X2) Switch
The DLY X1/X2 switch determines whether the PCM 41 is in the short time delay range (X1) or the long range (X2). When the button is out, the unit is set for the short range (0-200 ms) and 16 kHz bandwidth. Engaging the button selects the long range (0-400 ms), 6 kHz bandwidth, and illuminates the X2 LED [Fig. 2-1, "M"].

C. Input Level Control
The INPUT LEVEL control adjusts the input gain for properly matching the PCM 41's sensitivity to the input signal. Set this control so that the HEADROOM display [Fig. 2-1, "A"] occasionally indicates "0 dB" on program peaks, with average levels at about "–12 dB" (in the green display zone).

NOTE: A rear panel switch provides an additional 20 dB of gain for low level inputs, such as the output of an instrument pickup. For details, see Section 2.2, paragraph "E".

D. Feedback Control
Turning the FEEDBACK control clockwise feeds more of the delayed audio back to the delay line input, mixing it with the input signal. Increasing the FEEDBACK setting from #0 to #10 tends to sustain sounds and create resonant effects.

CAUTION: If FEEDBACK is set too high, oscillation or run-away may occur. The best remedy is to lower the FEEDBACK setting.

E. INV Switch (Feedback Invert)
Engaging the FEEDBACK INVERT button (INV) reverses the polarity of the delayed audio which is being fed back to the delay line input. Inversion of the feedback signal alters phase cancellation characteristics, thereby providing new dimensions and tonal color to various resonance, echo and flanging effects. (This button therefore has no effect if the FEEDBACK control is set at #0.)

F. LO Cut Filter (Feedback)
(See "G".)

G. HI Cut Filter (Feedback)
Engaging these filter buttons cuts out the low and/or high frequency portions of the delayed audio which is being fed back to the delay line input. In some cases, restricting the bandwidth of the feedback will allow more resonance (a more sustained effect) to be obtained before run-away occurs. Use of the HI cut and LO cut filters will change the overall sound quality, too, although their exact effect depends on the program material and the settings of other controls. (These buttons have no effect if the FEEDBACK control is set at #0.)

H. Bypass LED
In BYPASS mode, the MAIN OUTPUT audio is taken directly from the INPUT LEVEL control; the output is free of all signal processing, other than gain conditioning. This LED is illuminated when BYPASS mode is activated via a foot switch (not included) which plugs into the PCM 41 rear panel.

I. Output Mix Control
The OUTPUT MIX control establishes the ratio of direct sound to delayed sound in the PCM 41 MAIN OUTPUT. Full counterclockwise position (toward INPUT) provides 100% direct sound: no PCM 41 effects. Full clockwise position (toward DELAY) provides 100% delayed sound. The control is detented in center position, where the blend is 50%-50% direct-delayed, a good starting point for quick setup of flanging effects.

J. INV Switch (Delayed Audio Invert)
Engaging the DELAYED AUDIO INVERT button (INV) reverses the polarity of the delayed sound which is fed to the OUTPUT MIX control [Fig. 2-1, "I"].

K. Delay Time Display
One of these 11 LEDs is always illuminated to indicate the selected basic delay time. Two scales are provided, the upper for normal delay range (X1 = 0-200 milliseconds), and the lower for extended delay range (X2 = 0-400 milliseconds).

NOTE: The set delay time is only a nominal value, applicable when no VCO modulation is used, and when the DELAY MULTIPLY control [Fig. 2-1, "O"] is centered. DELAY MULTIPLY (and/or the VCO) can halve the actual delay, or double it – to a maximum of 800 ms when the unit is in DLY X2 mode and the 400 ms LED is illuminated.

L. Delay Select Buttons
The two DELAY SELECT buttons allow the basic delay time to be increased (press the button on the right) or decreased (press the button on the left). When the PCM 41 POWER switch is first turned ON, 0 ms delay is automatically selected.

M. X2 LED
This LED is illuminated whenever the X1/X2 DLY button [Fig. 2-1, "B"] is engaged (i.e., when the long delay range is selected). The LED is a visible reminder to refer to the lower scale of the DELAY TIME DISPLAY.

N. Repeat LED
The REPEAT LED is illuminated when the PCM 41 is set for infinite REPEAT mode (or "hold" mode), which is done by means of a foot switch that plugs into the rear panel. In this mode, a brief segment of the audio input is "captured", and replayed within the PCM 41 indefinitely. There is no signal degradation or fade out until either the REPEAT mode is cancelled by a second actuation of the foot switch, or until power is interrupted. (Maximum repeated segment is 800 ms.)

O. Delay Multiply Control
The DELAY MULTIPLY control allows continuous 4:1 adjustment of any selected delay setting. Full counterclockwise rotation (toward "X.5") will halve the selected delay. Full clockwise rotation (toward "X2") will double the selected delay. The mid position ("X1") is detented to allow positive indexing of the calibrated display.

NOTE: The "X1" and "X2" on this control should not be confused with the "X1" and "X2" on the DLY RANGE switch [Fig. 2-1, "B"]. Switching DLY RANGE from X1 to X2 changes the overall mode of the delay line from a maximum nominal delay of 200 ms to a maximum nominal delay of 400 ms with reduced signal bandwidth. Adjusting the DELAY MULTIPLY control from center (X1) to full clockwise (X2) modifies the voltage fed to the clock, which adds to the set delay, up to a maximum of 400 ms or 800 ms. Turning DELAY MULTIPLY counterclockwise from X1 to X.5 reduces the actual delay to half the set value.

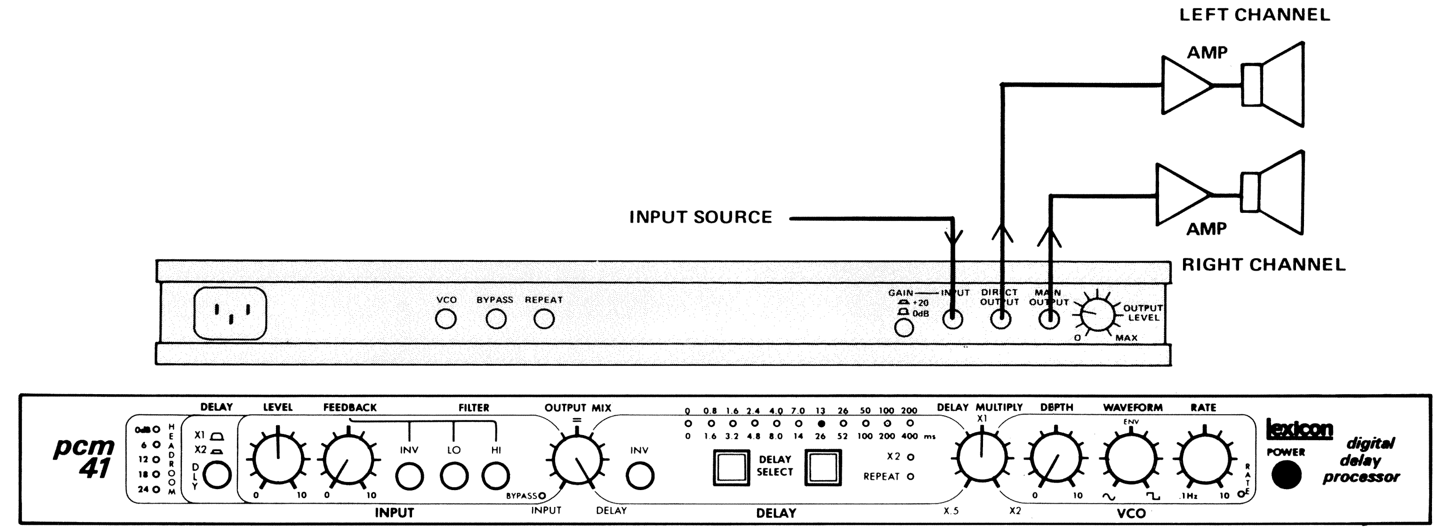


Figure 4-19 – Stereo Image Placement Using the PCM 41

4.7 AVOIDING ECHOES IN DISTRIBUTED SOUND REINFORCEMENT SYSTEMS

Most people think of a delay line as a device which can be used to create echoes. An important application, however, is the exact opposite – avoiding echoes. In large sound reinforcement systems, both indoor and outdoor, where speakers are placed in widely separated locations, the differing arrival times of sound at the listener's ear tends to destroy intelligibility (think of the last time you tried to hear a page in a train station or air terminal). Time delay cannot do much about reflections from walls and ceilings, but it can be used to avoid the confusion that results when a listener first hears the sound from a nearby speaker which is a distance from the place where the performer is located, and later hears the sound from the speaker which is nearest to the performer. Such "temporal confusion" not only destroys intelligibility, it is very fatiguing and tends to shorten attention spans.

If the sound at the remote speaker is delayed long enough so it leaves that speaker 5 ms to 30 ms after the sound from the near-performer speaker, the listener at the remote position will fuse the two sounds into a single, intelligible signal. Listeners near the performer's speaker will have no problem; they still hear the near-performer speaker first, and the remote speaker sounds like a distant echo.

Figure 4-20 illustrates a typical sound reinforcement setup. Since there may be several remote speakers at about the same distance from the main speaker, all can be fed by the same delay, and are hence referred to as a remote "zone." If another set of remote speakers is located at a greater distance, a second delay line (set at a longer delay) and another set of amplifiers will be required. Very complex systems may have many such zones. For more information on this subject, request Application Note AN-2 "Application of Digital Delay Units to Sound Reinforcement," available from Lexicon, Inc.

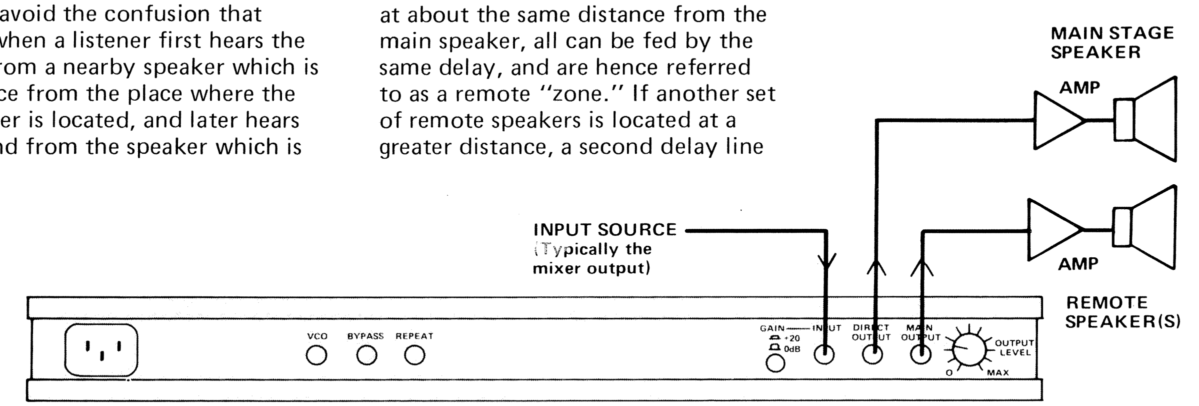


Figure 4-20 – Sound Reinforcement with One Remote Zone of Speakers
The DELAY time depends on the distance ("D") between the main and remote speakers. Delay time, in ms, should be 0.885 x D, plus an additional 5 to 30 ms. For example, if D = 100 feet, delay time should be set at 88.5 + 5 to 30 = 93.5 to 118.5 ms.

UNPACKING AND INSPECTION

Remove the PCM 41 from its packing material, and SAVE ALL PACKING MATERIALS in the event it becomes necessary to reship the unit. Thoroughly inspect both the PCM 41 and the packing material for indications of shipping damage, and report any damage found to the carrier.

TABLE OF CONTENTS

SECTION SIX
Specifications

6.1 GENERAL PERFORMANCE

Total Distortion and Noise
0.04% typical, 0.07% maximum @ 1 kHz, DELAY/DELAY MULTIPLY = X1; 0.1% typical over bandpass of 20 Hz to 15 kHz.

Frequency Response
20 Hz to 15 kHz, +0, −1 dB in X1 mode;
10 Hz to 16 kHz, +0, −3 dB in X1 mode.
20 Hz to 6.0 kHz, +0, −3 dB in X2 mode.

Dynamic Range
Better than 90 dB, 20 Hz — 20 kHz bandwidth.

Delay Capacity
@ DLY X1: 200 milliseconds at full bandwidth;
400 milliseconds at 6 kHz bandwidth.
@ DLY X2: 400 milliseconds at full bandwidth;
800 milliseconds at 6 kHz bandwidth.

Delay Selection
11 pushbutton selected taps; each tap is continuously variable over a 4:1 range via the Delay Multiply control. Taps are at: 0, 0.8, 1.6, 2.4, 4, 7.0, 12, 26, 50, 100 and 200 ms in DLY X1 mode (double these times in DLY X2 mode).

VCO Modulation
Depth is adjustable from 0 (none) to a 4:1 sweep of delay time. Rate is adjustable from 0.1 Hz to 10 Hz. An LED flashes to indicate the VCO modulation rate.

VCO Shape
Continuous adjustment is available between Sine wave and Envelope, or Square wave and Envelope functions.

6.2 INTERFACE INFORMATION

Input Type
Balanced differential input, via standard 1/4" Tip-Ring-Sleeve phone jack, offers 40 dB common mode rejection. Will also accept unbalanced input.

Input Impedance
Balanced or unbalanced input; 40K ohms impedance.

Input Level
With adjustable INPUT LEVEL control and 20 dB GAIN switch on rear panel, maximum input levels of from −23 dBV to +19 dBV (0.07 V to 9 V rms) can be accommodated.*

Output Type
Both Main and Direct outputs are unbalanced, and have standard 1/4" Tip-Sleeve phone jacks.

Output Impedance
Main Out: 100 ohms actual source impedance.
Direct Out: 600 ohms actual source impedance.
Both outputs are intended for driving high impedance loads, but are operable with loads as low as 600 ohms.

Output Level
Main Out: +19 dBV (9 V rms) maximum. Will drive loads of 2K ohms or greater at full level. An Output Level control varies the level.

Direct Out: Level depends only on Input Level and 20 dB Gain switch; when Headroom indicates 0 dB, output level is +11 dBV (3.5 V rms).

Remote Jacks
Rear panel 1/4" phone jacks accommodate external switches and controls as follows:

Bypass: When the Tip and Sleeve of the mono phone jack are shorted together, the PCM 41 is placed in bypass mode.

Repeat: When the Tip and Sleeve of the mono phone jack are momentarily shorted, the PCM 41 captures a brief segment of the input and holds it in repeat mode. A subsequent momentary contact of Tip and Sleeve releases the segment, returning the unit to normal operation.

VCO: This stereo phone jack may be connected to a potentiometer or a voltage source; whenever a plug is inserted in the jack, the Delay Multiply control is replaced by the external device. A pot should be from 10K to 500K ohms (50K ohms ideally), Tip wired to the wiper, Ring and Sleeve to the pot inputs. A positive voltage source of 0 to 10 V can feed the Tip, with Ring and Sleeve connected to the common.

**The 0 dBV reference is 1.0 volts rms.*

6.3 CONTROLS AND INDICATORS

Headroom Indicator
5 LEDs display input level in 6 dB steps. The upper limit of "0 dB" is the point at which the analog to digital converter clips.

Remote Indicators
Bypass and Repeat LED's on the front panel are illuminated when the respective functions have been activated via the remote input jacks.

Delay Range
A latching pushbutton selects X1 or X2 range (200 ms or 400 ms maximum indicated delay); an LED is illuminated in X2 mode. The Delay Select pushbuttons increment the nominal delay time up and down, as shown by the 11 Delay Time LED's.

Mixing Controls
Level controls are provided for the Input and Feedback circuits. Feedback also includes its own Invert (polarity) switch and Lo cut (80 Hz) and Hi cut (4 kHz) filters. The Main Output is provided with an Output Mix control to blend direct and delayed sound (center detent for equal blend), a delay Invert switch, and an Output Level control.

6.4 POWER AND DIMENSIONS

AC Requirements
115 or 230 volts (selectable), 50 or 60 Hz, 20 watts maximum. Standard IEC power connector on rear of unit; 3-prong cord provided.

Protection
Mains are fused (standard U.S. 3AG fuses).

Export Models
Mains and secondaries are fused (European 20 mm fuses). An RFI power line filter is also installed. NOTE: A 100 volt export model is available on special order.

Dimensions
Standard 19" relay rack mount (483 mm).
1-3/4" high (44 mm) by 11" deep (280 mm).

Weight
Net, 5.5 lbs (2.5 kg); shipping 8 lbs. (3.6 kg).

All specifications are subject to change without notice.