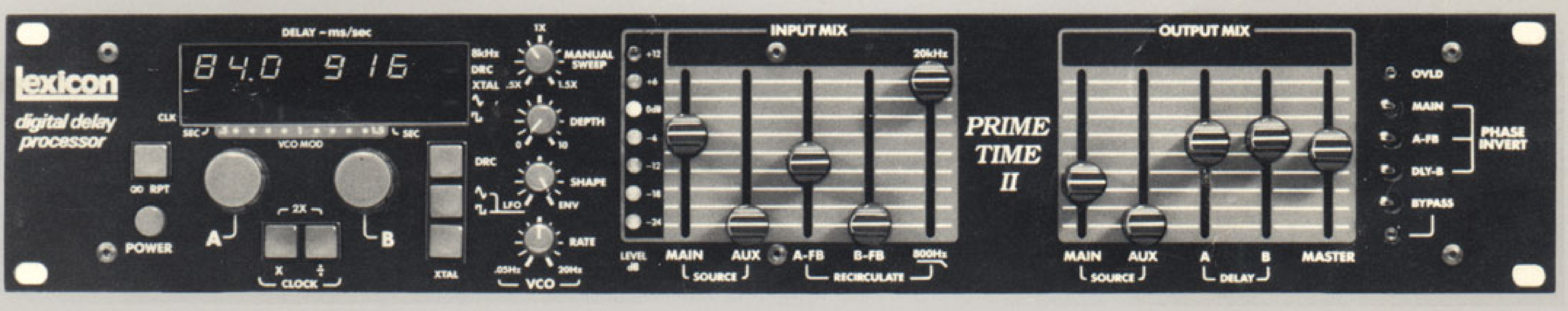
Owners Manual

PRIME TIME II

digital delay processor

Model 95



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NOTICE

Lexicon, Inc., reserves the right to make improvements at any time and without notice in the product described in this manual.

1 Introduction

1.0 Description

The Model 95 Prime Time II is a PCM (pulse-code modulation) digital delay processor with a single delay line incorporating two independently adjustable delay outputs, each with its own output jack, output and recirculation level control, and digital display. Each delay output has a range of 128 selectable delay taps. With the use of the 95's on-board mixing facilities, the signals from the two delay outputs can be independently routed or combined in any number of ways, providing enormous flexibility in creating different sonic perspectives, spatial enhancements, or special effects.

A VCO (voltage-controlled oscillator) section provides both manual or automatic time-sweep modulation of the delay settings. Even though the same sweep modulation function affects both delays, the result can be quite different for the two delay times because the characteristics of time-sweep modulation are determined by the length of the delay period being modulated.

An auxiliary input provides additional patching possibilities for interaction with signal processors to create a variety of submixes consisting of delayed and processed signal combinations, as well as many other mixing possibilities. The main and auxiliary inputs form a composite signal when fed into the 95's delay line.

The Model 95 can be equipped with various memory options for delay times up to 7.68 seconds. Other features that are standard on all units include DRC (dynamic recirculation control), infinite repeat, a metronome clock function, selectable phase inversion of various signals, and special jacks to allow output control signals to other devices and remote-control operation of various front-panel controls.

The Model 95 has been expressly designed to provide a cost-effective package incorporating the finest performance, most optimum human-engineered operation, and highest reliability currently available with present-day technology.

1.1 Precautions

The Model 95 has been carefully engineered to make it a very "well-behaved" audio device, but certain precautions should be observed as a matter of course, consistent with good practice for any piece of audio equipment.

Before turning your Model 95 on or off, lower the volume on your power amplifier or monitoring system to avoid undesirable transients.

Never connect power sources or audio power amplifiers directly into any of the Model 95's XLR connectors or phone jacks. The Model 95's inputs are designed for line or microphone level signals. If a guitar amplifier is used as a signal input, a suitable attenuator pad must be used to lower the level ahead of the inputs to the Model 95.

To prevent fire or shock, never operate the Model 95 in the rain or exposed wet locations.

Please read Section 2 "Installation and Interfacing" before operating the Model 95.

WARNING

All servicing of the Model 95 Prime Time II should be performed by qualified service personnel. Hazardous high-voltage sources are located under the top and bottom covers of the unit. To avoid electrical shock, remove the power cord before removing covers. Service procedures consistent with good safety practice should be used at all times.

1.2 Location of Controls, Indicators, and Connectors

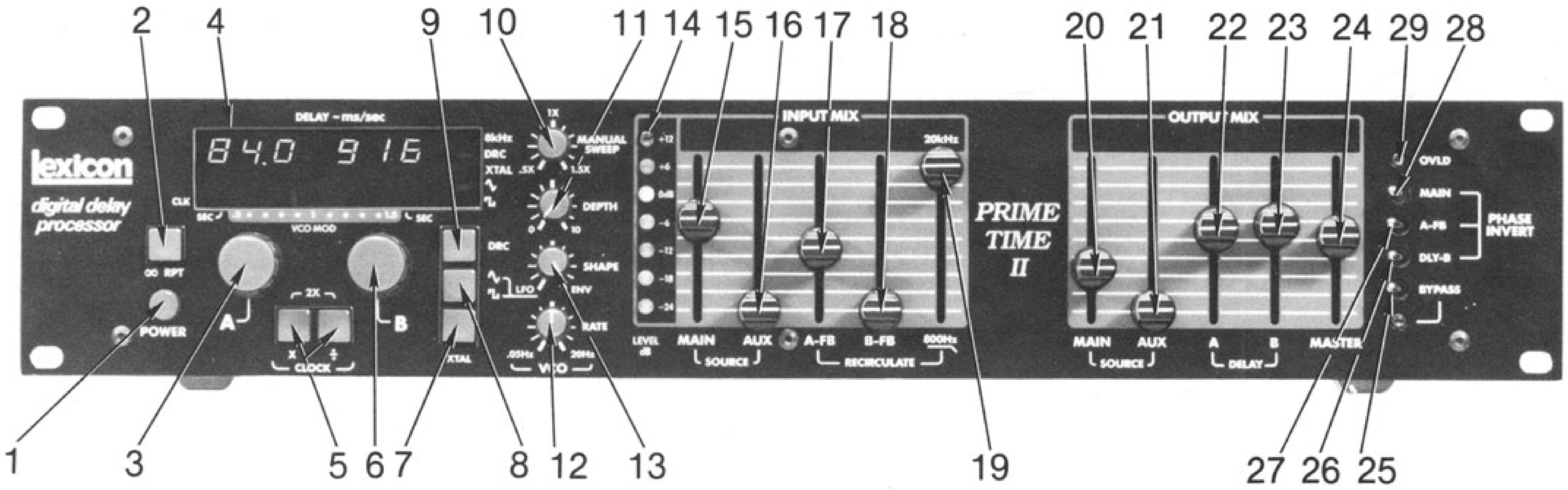


FIG. 1.1. Front panel.

1. POWER Switch

Push-on, push-off mains switch.

2. ∞ (Infinite) RPT

This momentary pushbutton engages or disengages the Infinite Repeat function. When engaged, the memory contents are trapped and repeated indefinitely. The word "Hold" and the selected delay times are alternately flashed by the display to indicate that the Infinite Repeat function has been engaged. **Note:** The rear-panel ∞ REPEAT jack and the front-panel switch can be used independently.

3. [A]

This rotary control selects one of 128 delay settings for tap "A".

4. Display Window

DELAY–ms/sec: Two 3-digit, 7-segment LED readouts indicate delay time in milliseconds or seconds for delay taps "A" and "B". The delay settings and "Hold" are alternately flashed in Infinite Repeat Mode.

CLK: This LED indicator flashes at a rate equal to the period of the interval set by the [÷] pushbutton.

SEC: These two LEDs light to indicate that the delay time of the respective tap is being displayed in seconds.

VCO MOD: This 20-segment bar-graph LED display provides a "flying spot" portrayal of modulation characteristics and rate when the VCO is engaged.

8 kHz: This LED lights when the Model 95 is in 2X mode and operating at reduced bandwidth.

DRC: This LED lights when the Dynamic Recirculation Control function (DRC) is engaged.

XTAL: This LED lights when XTAL mode is engaged.

∿ : This LED lights when the selected LFO waveform is a sine wave.

⊓⊔ : This LED lights when the selected LFO waveform is a square wave.

5. 2X/CLOCK

These two pushbuttons, when pressed simultaneously, place the unit in 2X mode. The delay times are automatically doubled and the bandwidth is reduced to 8 kHz. The Clock mode is activated when either button is pressed individually.

[÷]: This momentary pushbutton sets the number of intervals by which the total delay period will be divided.

[X]: This momentary pushbutton sets the number of pulses per interval which will be available at the CLOCK OUTPUT jack.

Note: Pressing both [X] and [÷] pushbuttons simultaneously while in Clock Display mode activates the Clock Output mode, which is disengaged when both pushbuttons are pressed simultaneously again.

6. [B]

This rotary control selects one of 128 delay settings for tap "B".

7. XTAL

This momentary pushbutton alternately enables the VCO (voltage-controlled oscillator) or the crystal oscillator as the sampling clock source.

8. LFO (Low-Frequency Oscillator)

This momentary pushbutton alternately enables sine-wave or square-wave automatic VCO time-sweep modulation.

9. DRC (Dynamic Recirculation Control)

This momentary pushbutton alternately enables and disables the Dynamic Recirculation Control function. This control automatically attenuates or increases the level of the recirculated signal as a function of the input signal level.

10. MANUAL SWEEP

This rotary control varies the delay time over a range of 3:1 (from 0.5 to 1.5 times the delay setting) by changing the clock sampling rate (both delay settings are affected).

11. DEPTH

This rotary control enables the automatic time-sweep function and sets the amount of internal modulation fed to the VCO. Maximum depth results in a 3:1 sweep.

12. RATE

This rotary control selects the frequency of the LFO, from 0.05 Hz to 20 Hz, to control the rate of the automatic time-sweep function.

13. SHAPE

This rotary control selects a continuously variable mix of low-frequency oscillator (LFO) waveform output and envelope-follower output as the time-sweep automatic modulation function.

14. LEVEL Indicator

This 7-position LED display indicates the level of signal fed to the A/D converter in 6-dB steps: "+12 dB" corresponds to the signal level at which the A/D converter will begin to hard limit (clip). **Note:** The LEVEL display is preceded by the compression/limiter circuits and thus becomes nonlinear at high input levels when the LIMITER switch is engaged.

15. MAIN SOURCE

This slide control sets the level of signal from the MAIN INPUT that is fed into the A/D converter and the INPUT MIX OUTPUT.

16. AUX SOURCE

This slide control sets the level of signal from the AUX. INPUT that is fed into the A/D converter and the INPUT MIX OUTPUT.

17. A-FB

This slide control sets the level of delayed signal recirculated from delay tap "A" and fed into the Input Mix section.

18. B-FB

This slide control sets the level of delayed signal recirculated from delay tap "B" and fed into the Input Mix section.

19. "800 Hz–20 kHz" Filter

This slide control adjusts the cut-off frequency of a single-pole low-pass filter in the recirculation path. This filter acts upon both the A-FB and B-FB signals equally.

20. MAIN SOURCE

This slide control sets the level of signal from the MAIN INPUT that is fed to the MASTER control in the Output Mix section.

21. AUX SOURCE

This slide control sets the level of signal from the AUX. INPUT that is fed into the MASTER control in the Output Mix section.

22. DELAY A

This slide control sets the level of signal from delay tap "A" that is fed into the MASTER control in the Output Mix section.

23. DELAY B

This slide control sets the level of signal from delay tap "B" that is fed into the MASTER control in the Output Mix section.

24. MASTER

This slide control sets the level of signal from the Output Mix section that is fed to the MASTER OUTPUT.

25. BYPASS

This toggle switch engages or disengages Bypass mode; the LED lights when Bypass mode is engaged.

26. DLY-B

This switch inverts the phase of the Delay-B signal that is fed into the Output Mix section.

27. A-FB

This switch inverts the phase of the Delay-A signal that is fed back to the Input Mix section.

28. MAIN

This switch inverts the phase of the MAIN INPUT signal that is fed directly to the Output Mix section.

29. OVLD (Overload) Indicator

This LED lights to indicate when the level of signal fed to the MASTER OUTPUT is overloading that circuit.

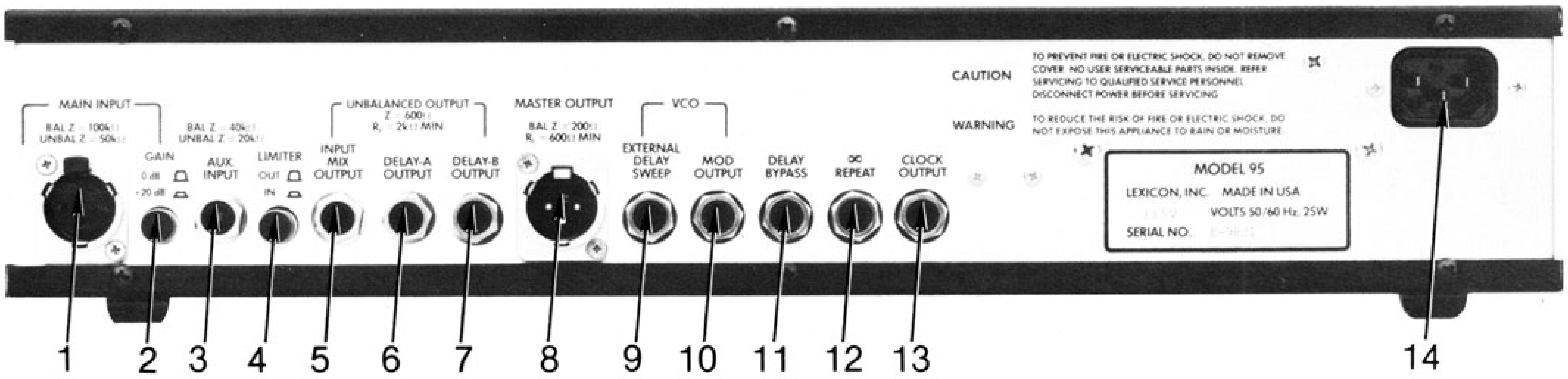


FIG. 1.2. Rear panel.

1. MAIN INPUT

This XLR-3 female connector provides the main audio input. It is an electronically balanced differential input that also accepts unbalanced signal connection.

2. GAIN switch

This switch can provide 20 dB of gain in the MAIN INPUT to match low-level input sources.

3. AUX. INPUT

This 1/4″ tip-ring-sleeve phone jack provides a secondary audio input. It is an electronically balanced differential input that also accepts unbalanced signal connection.

4. LIMITER switch

This switch can engage a two-step limiter circuit in the input stage of the delay line to protect against overloading of the A/D converter.

5. INPUT MIX OUTPUT

This 1/4″ phone jack provides output of the signals from the Input Mix section on the front panel for independent submixes or stereo or quad effects.

6. DELAY-A OUTPUT

This 1/4″ phone jack provides output of the signals from the Delay-A tap.

7. DELAY-B OUTPUT

This 1/4″ phone jack provides output of the signals from the Delay-B tap.

8. MASTER OUTPUT

This XLR-3 male connector is the main audio output of the Model 95. It is an electronically balanced output that also accepts unbalanced signal connection.

9. EXTERNAL DELAY SWEEP

This stereo phone jack allows connection to an external voltage source for delay sweep control.

10. MOD (Modulation) OUTPUT

This stereo phone jack provides output of the modulation control signals of the Model 95 to enable control of external devices such as synthesizers or other delay units.

11. DELAY BYPASS

This 1/4″ phone jack allows connection to an optional footswitch for remote control of the Bypass function.

12. ∞ (Infinite) REPEAT

This 1/4″ phone jack allows a momentary switch closure to toggle the Model 95 into or out of the Infinite Repeat mode. Either an optional footswitch or standard 0- to +5-V logic signal can be connected to this jack.

13. CLOCK OUTPUT

This 1/4″ phone jack provides output of the clock pulses per beat programmed from the front panel. These pulses can be used to drive synthesizer sequencers or automatic drum units, or they can be amplified and used as an audible metronome.

14. POWER CONNECTOR

The power connector accepts a standard IEC (NEMA) power cord (included). Before plugging in the unit, make sure the AC mains match the operating voltage specified on rear panel. Also, before connecting the power cord, verify that the front-panel POWER switch is off (button out).

1.3 How To Use This Manual

This manual has been organized with the goal of providing appropriate instruction and application information for persons at all levels of experience and with many different backgrounds. Briefly, the remaining sections of this manual are arranged as follows:

Section 2: Installation and Interfacing

This section contains the information and technical data needed to set up and interface the Model 95 with studio or sound reinforcement equipment, musical instruments, etc.

Section 3: Basic Operation—How To Get Started

This section is a "guided tour" of the basic operation of the Model 95, presuming *no* technical knowledge or previous experience at all. Beginning users may wish to work through the entire section to form a solid base of understanding, while the more experienced person may wish to use only some parts to clarify the function of the Model 95's more exotic features, or may elect to skip this section entirely.

Section 4: Applications

This section is a very cursory survey of the possible applications of the Model 95, with emphasis on understanding the "why" as well as the "how" of using this device in actual performance or recording sessions. Examples of patches, effects, etc., are presented, but the intent throughout is to equip the user with the background knowledge and basic application concepts to be able to work competently and creatively to achieve the desired results in any context. The experienced audio engineer may elect to briefly scan this section.

Section 5: Detailed Description of Controls and Indicators

This is a reference section, with thorough functional descriptions of each feature on the front and rear panels of the Model 95.

Section 6: Specifications

This section is a capsulized listing of the performance specifications of the Model 95.

Section 7: Service and Warranty

This section contains service and warranty information.

2
Installation and Interfacing

2.0 Introduction

This section of the manual contains important information for interconnecting the Model 95 with other audio and musical equipment. Wiring information for common types of audio and control connection is presented, with discussion of the signal characteristics and their implication in normal application. Several interconnect diagrams for common situations are presented.

2.1 Mounting

The Model 95 is designed to be mounted in a standard 19″ rack, occupying 3 3/4″ of vertical space and extending 13 1/2″ behind the front panel. If units directly above or below the Model 95 have very deep frames, or generate large amounts of heat (e.g., power amplifiers), space should be left between such units and the Model 95 to allow for adequate ventilation.

If the Model 95 is to be shipped while rack mounted, the rear panel should be supported to protect from shock and vibration.

2.2 Power Requirements

The Model 95 is factory set to operate at either 100, 115, or 230 VAC (50-60 Hz). The factory-set voltage is indicated on the rear panel next to the power receptacle. The maximum power required is 40 W. Units set for 100 or 115 VAC are fused at 3/4 ampere, and those set for 230 VAC are fused at 0.4 ampere.

WARNING

The operating voltage should not be changed except by a qualified service technician.

The Model 95's power cord has a standard 3-pin IEC connector to simplify adaptation to power sources in various countries and to provide chassis-to-mains grounding according to accepted safety standards.

2.3 Audio Signal Interfaces

2.3.1 MAIN INPUT

The MAIN INPUT is a 3-pin XLR female connector, with pin 2 = HIGH. The input impedance is 50 kΩ or greater for balanced signals, and 20 kΩ or greater for unbalanced signals. Signal levels from −20 to +19 dBV can be accommodated. (The GAIN pushbutton switch should be engaged for signal levels lower than 0 dBV. This activates a gain circuit in the MAIN INPUT to increase input sensitivity by 20 dB to match low-level inputs.)

This input is typically used with low-to-medium impedance signals. High-impedance musical instruments may be input directly, but some high-end loss from loading might be experienced. When using an electric guitar as the signal source, if possible, use a preamplifier output as the input to the Model 95.

If the signal source has other than XLR connectors (pin 2 = HIGH), it will be necessary to obtain a suitable adapter or cable. Refer to Fig. 2.1 for wiring diagram.

2.3.2 AUX. INPUT

The auxiliary input has characteristics similar to the MAIN INPUT, except that it does not have a gain circuit to match low-level signals, and the connector is a 1/4″ phone jack (balanced: tip = HIGH, ring = LOW, sleeve = ground). Refer to Fig. 2.1 for wiring diagram.

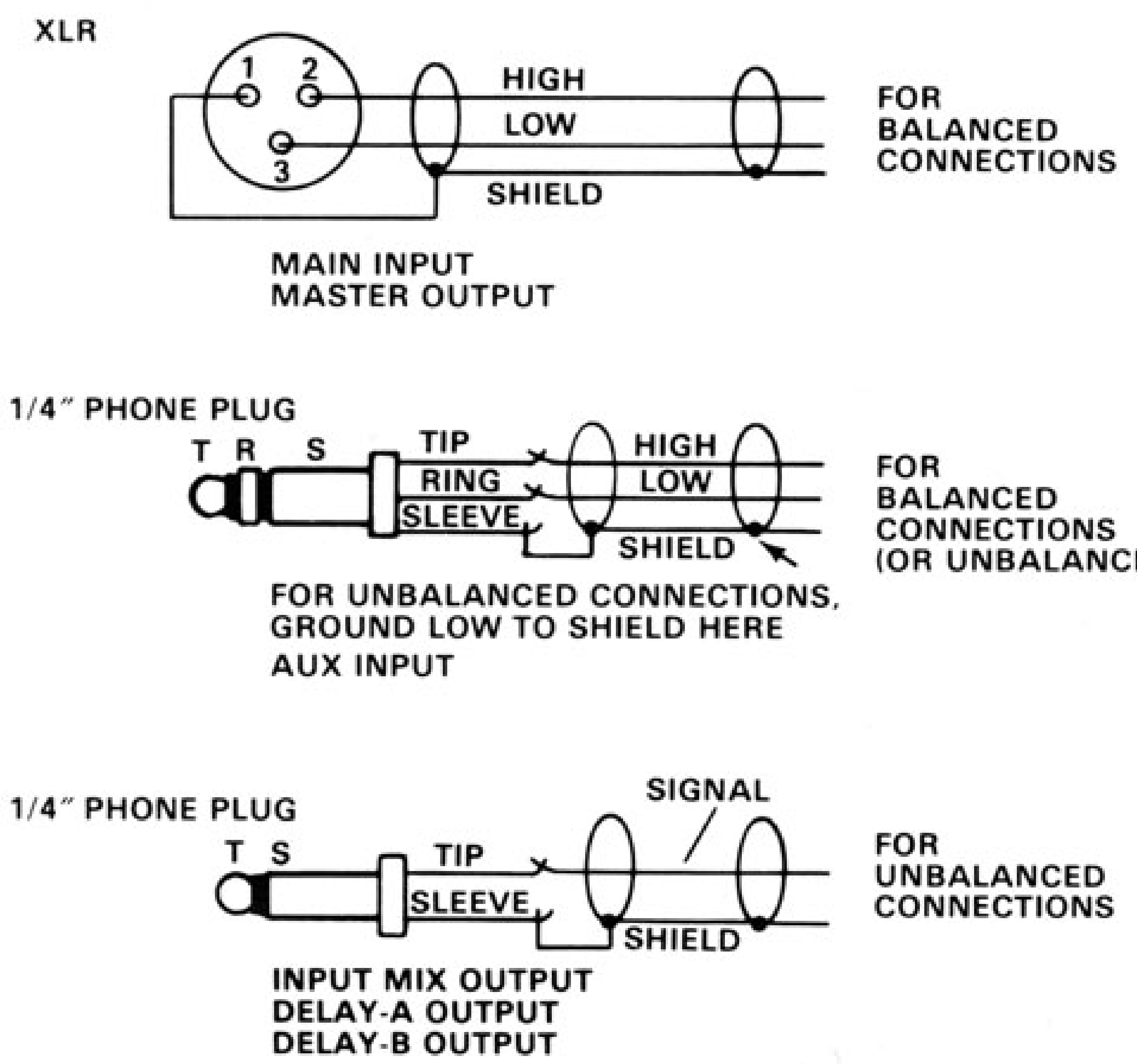


FIG. 2.1. Audio signal connections.

2.3.3 INPUT MIX OUTPUT

This standard 1/4″ phone jack outputs the combination of direct and recirculated signals set up by the Input Mix section on the front panel. It is useful for setting up stereo configurations, synthesizing stereo and quad from mono or stereo sources, respectively, or creating a variety of independent submixes. This is an unbalanced output at line level (+16 dBV, maximum, into a 2-kΩ load), with an output impedance of 200 Ω. If this output is connected to a mic or instrument level input, such as a guitar amplifier, it will be necessary to provide an external pad to prevent overloading. Refer to Fig. 2.1 for wiring diagram.

2.3.4 DELAY-A OUTPUT

This standard 1/4″ phone jack outputs the Delay-A signal, unmixed with any other signals. The levels and other characteristics are the same as that of the INPUT MIX OUTPUT above. Refer to Fig. 2.1 for wiring diagram.

2.3.5 DELAY-B OUTPUT

This standard 1/4″ phone jack outputs the unmixed signal from Delay-B, and has the same characteristics as the INPUT MIX OUTPUT and DELAY-A OUTPUT. Refer to Fig. 2.1 for wiring diagram.

2.3.6 MASTER OUTPUT

This male XLR connector is the main output, which carries the combination of signals set up at the Output Mix section on the front panel. The output level is variable over a wide range, so that this output may be interfaced directly with mic, instrument, or line-level inputs. The output is electronically balanced (pin 2 = HIGH), but can be used as an unbalanced output by grounding pin 3. The output impedance is 200 Ω for balanced or unbalanced signals. The maximum level is +22 dBV, balanced, or +16 dBV, unbalanced. Refer to Fig. 2.1 for wiring diagram.

2.4 Control Signal Interfaces

2.4.1 EXTERNAL DELAY SWEEP

This input jack allows the delay times of the unit to be swept by an external foot pedal or a control voltage. When a plug is inserted into the 1/4″ phone jack, the MANUAL SWEEP control on the front panel is disabled and its control function is replaced by the voltage signal at the tip of the plug. When connected to a Lexicon A-CP-41 Delay Control Pedal (optional) or an external potentiometer by a 3-conductor, tip-ring-sleeve plug, the external pot acts exactly like the front-panel MANUAL SWEEP control. When XTAL mode is disabled, the delay times can be swept (both settings are affected) for user-articulated flanging, or to adjust echo times, etc. Figure 2.2 shows the wiring diagram for the EXTERNAL DELAY SWEEP pot.

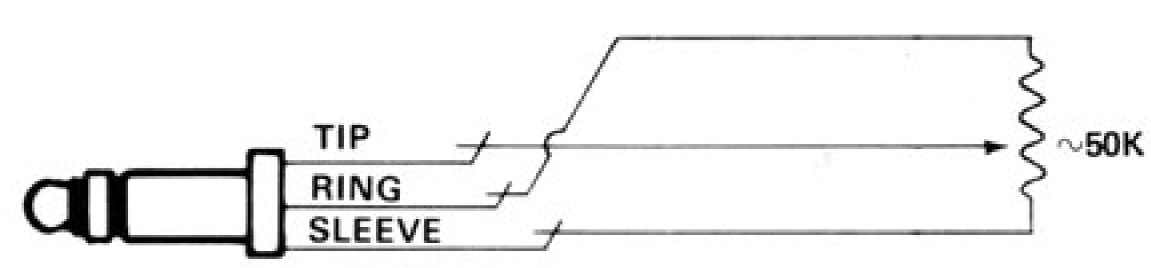


FIG. 2.2. Wiring diagram, EXTERNAL DELAY SWEEP pot.

An external voltage in the range of 0 to +10 VDC can be used in place of a potentiometer. The internal clock (the function which is actually modulated) has a linear response over a 3:1 range from 0 to +10 V (0 V corresponds to 1.5x, or the longest times, and +10 V corresponds to 0.5x, the shortest delay factor). Any external waveform can be used, as long as it lies in the proper voltage range.

2.4.2 MOD OUTPUT

The modulation output provides a delay control signal corresponding to the mix of waveform and signal-envelope characteristics developed by the SHAPE control on the front panel. This signal can be connected to other delay units as a control input, used to modulate synthesizer oscillators and other functions, or it can be input back to the EXTERNAL DELAY SWEEP through a volume pedal to provide a remote foot control of the sweep depth (see Fig. 2.3 for connection diagram).

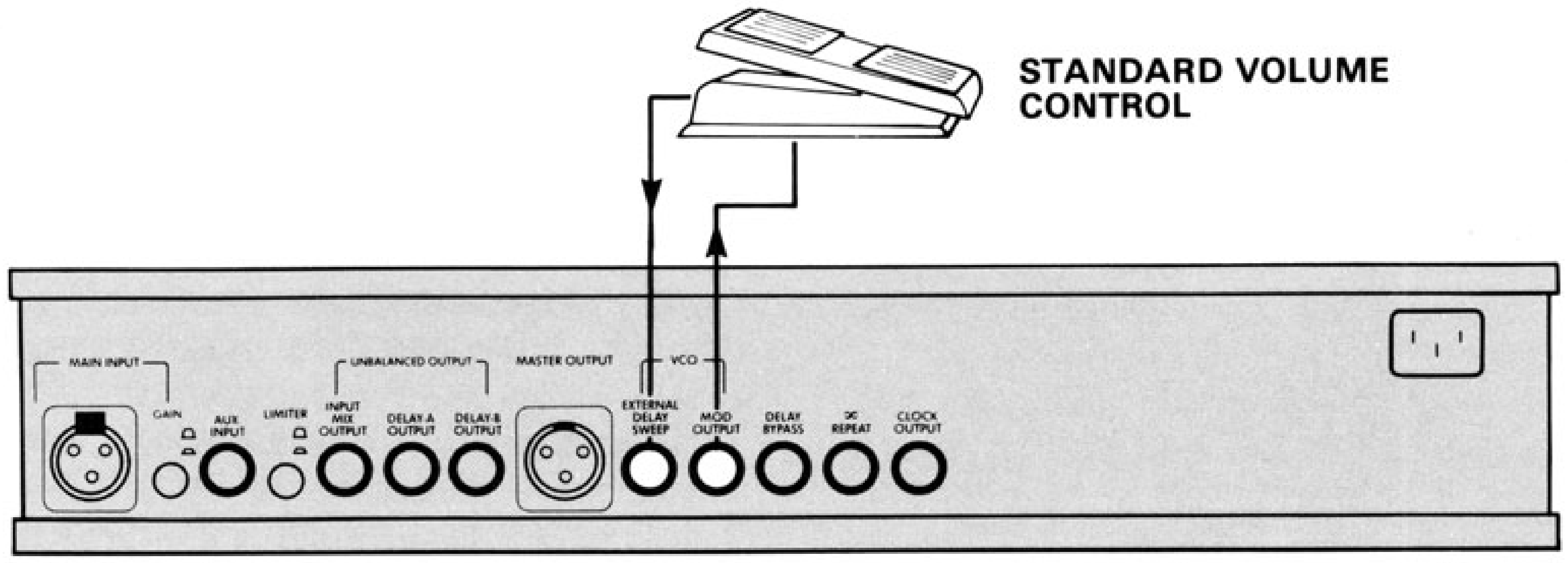


FIG. 2.3. Connecting MOD OUT and EXTERNAL DELAY SWEEP for foot control.

This jack provides two complementary signals. The tip carries the modulation signal exactly as it appears internally. The ring of the jack carries the same signal but in inverse polarity. The MOD OUTPUT can then be used to modulate another delay in opposite phase from the original. This inverted signal can be accessed by wiring a connector cable especially for it, or by partially retracting an ordinary plug from the jack (see Fig. 2.4 for wiring diagram).

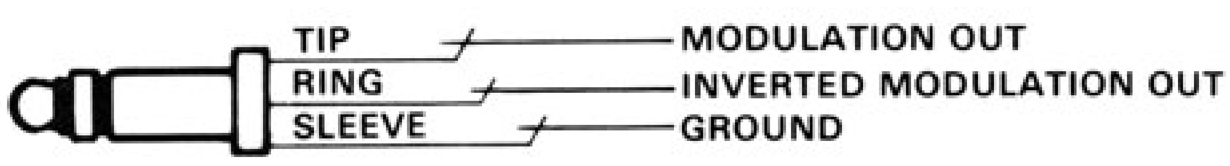


FIG. 2.4. Accessing normal and inverted MOD OUT.

2.4.3 DELAY BYPASS

The DELAY BYPASS input allows an external switch to be used to activate the Bypass function. When the tip and the sleeve of the jack are connected together, the result is exactly like operating the front-panel BYPASS switch. The delayed audio signals are shutoff from the INPUT MIX OUTPUT and MASTER OUTPUT and the signals from the Input Mix section are routed directly to the MASTER output level control. Note that the DELAY-A OUTPUT and DELAY-B OUTPUT are not affected except that recirculation is eliminated. This jack can be used with an optional Lexicon A-FS-97 single footswitch or an A-FS-41 dual footswitch assembly, or any SPST switch. See Fig. 2.5 for wiring diagram.

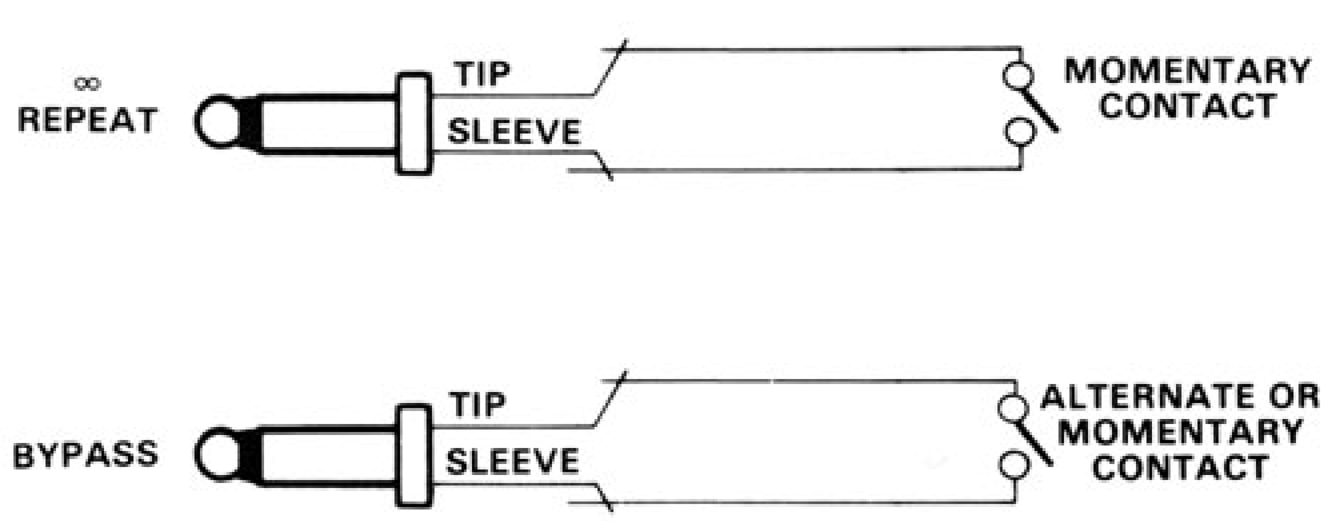


FIG. 2.5. Wiring diagrams for EXTERNAL BYPASS and Infinite Repeat.

2.4.4 ∞ REPEAT

The Infinite Repeat input jack allows external switching of the Infinite (∞) Repeat function. A momentary switch closure between the tip and the sleeve portions of the jack will cause the unit to toggle into or out of the Infinite Repeat mode. This jack can be used with optional Lexicon footswitches or any SPST switch. See Fig. 2.5 for wiring diagram.

2.4.5 CLOCK OUTPUT

The metronome clock output is a train of narrow, positive-going pulses from 0 to +5V. If an audible metronome is desired, this output can be connected to a monitor amplifier with an ordinary phone cable. If the Clock function is to be used to drive an external clock input on a synthesizer, sequencer, or automatic rhythm unit, the type of connector used will depend on the unit to be interfaced. Many devices can be directly connected with a common cable or adaptor. Some units have different requirements as to the type of trigger pulse used, or may require an unusual type of connector. These can be easily interfaced with a relatively simple adapting connector. Figure 2.6 shows connections to two common types of devices that require special connection.

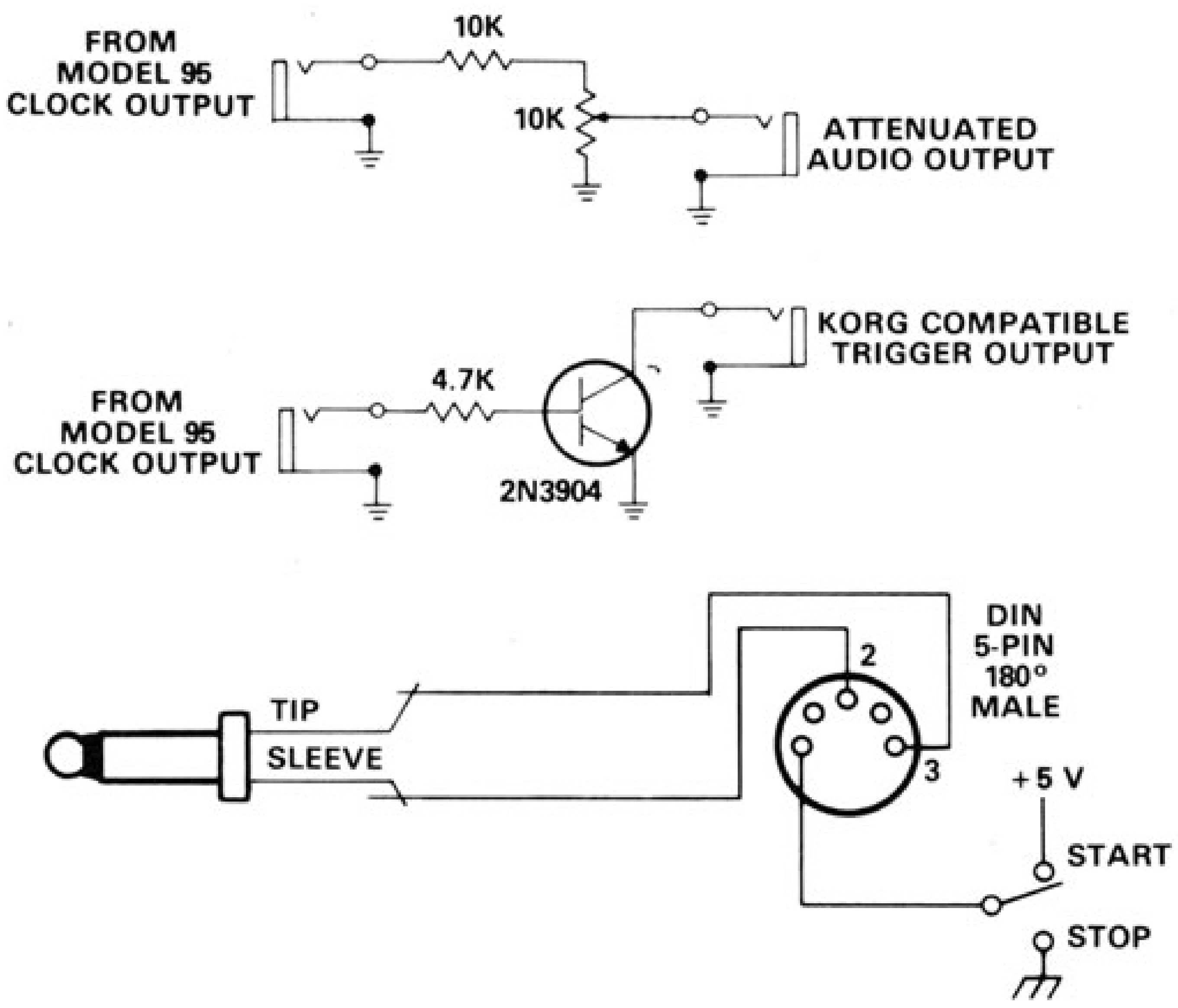


FIG. 2.6. Wiring diagrams, CLOCK OUTPUT interface.

When interfacing the Clock output to other devices, it is also important to take note of the division of time that will be required. Different types of sequencing and rhythm devices use different numbers of clock pulses to represent a unit of time. Many sequencer and keyboard arpeggiators use one pulse per event, but most drum units and some sequencers use values such as 16, 24, 32, or 48 clock pulses per quarter-note unit. When programming the clock rate from the front panel, the left-side of the display window indicates the number of pulses produced for each division of the loop. This number can be set for the value appropriate to the unit in use and the time division desired. Figure 2.7 gives the values for a few common types of units at different time divisions.

	X	÷
LINN	48	2,4,8,16
OBERHEIM	48	2,4,8,16
ROLAND	24	2,4,8,16

FIG. 2.7. Clock programming for common rhythm units.

2.5 Interconnect Diagrams

The Model 95 can be effectively connected with audio systems in a number of ways. Figures 2.8 through 2.13 show the most common of these.

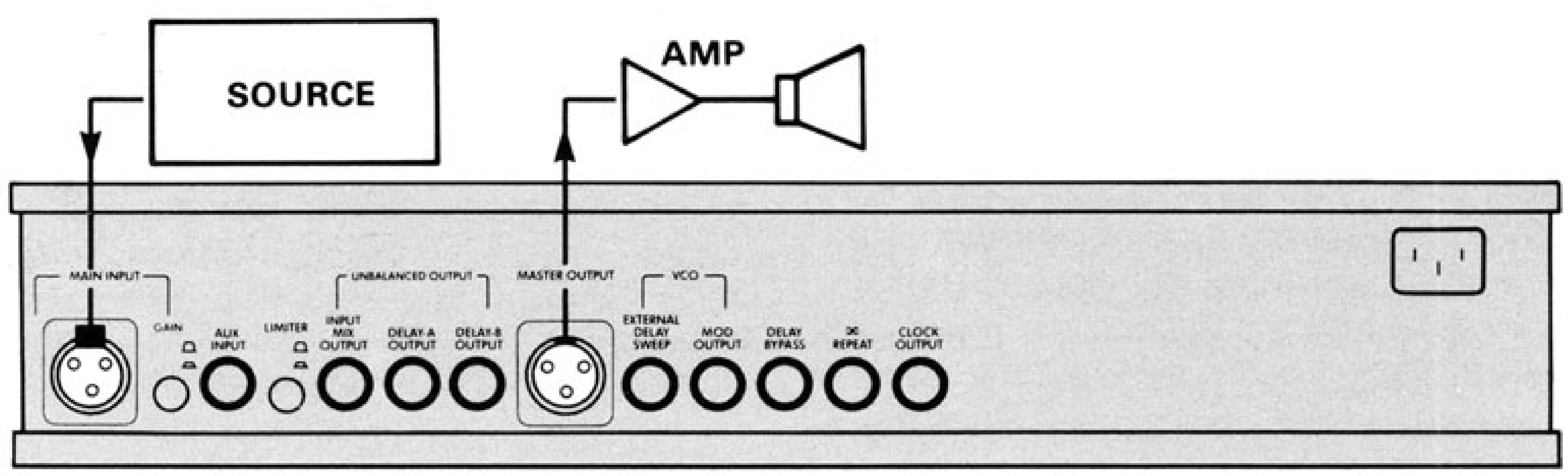


FIG. 2.8. Basic connection.

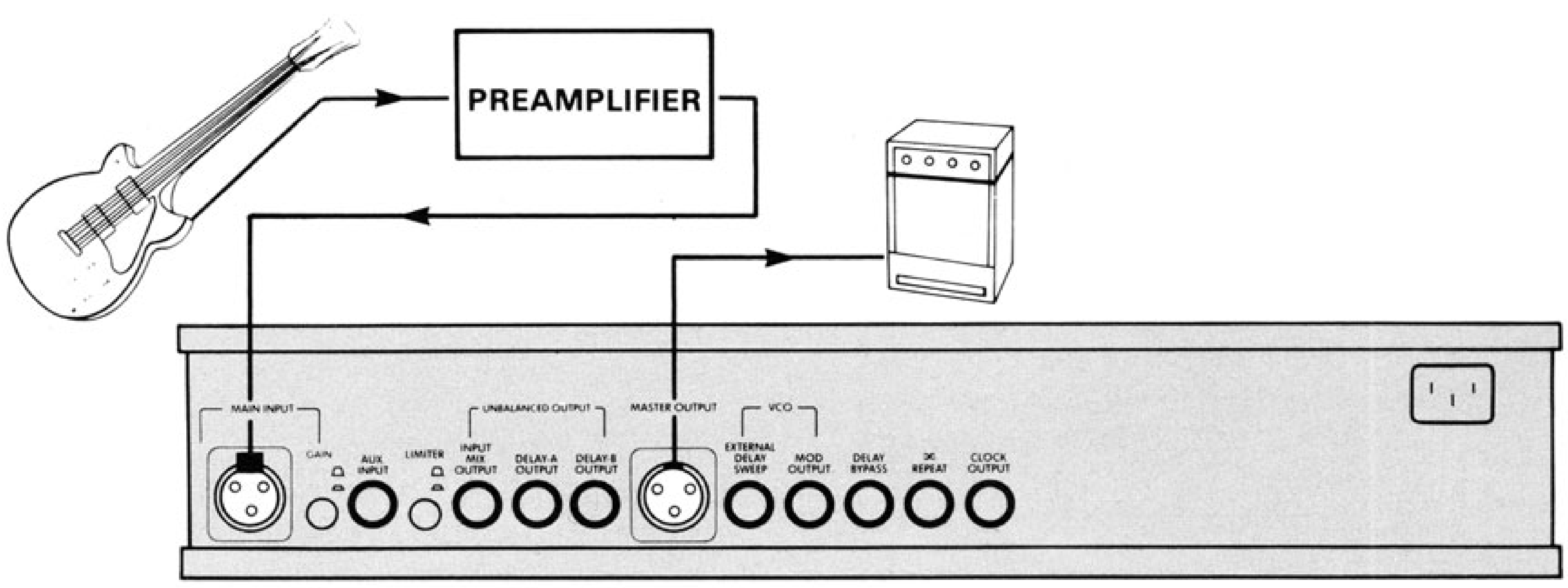


FIG. 2.9. Connection to guitar amplifier, using external preamp.

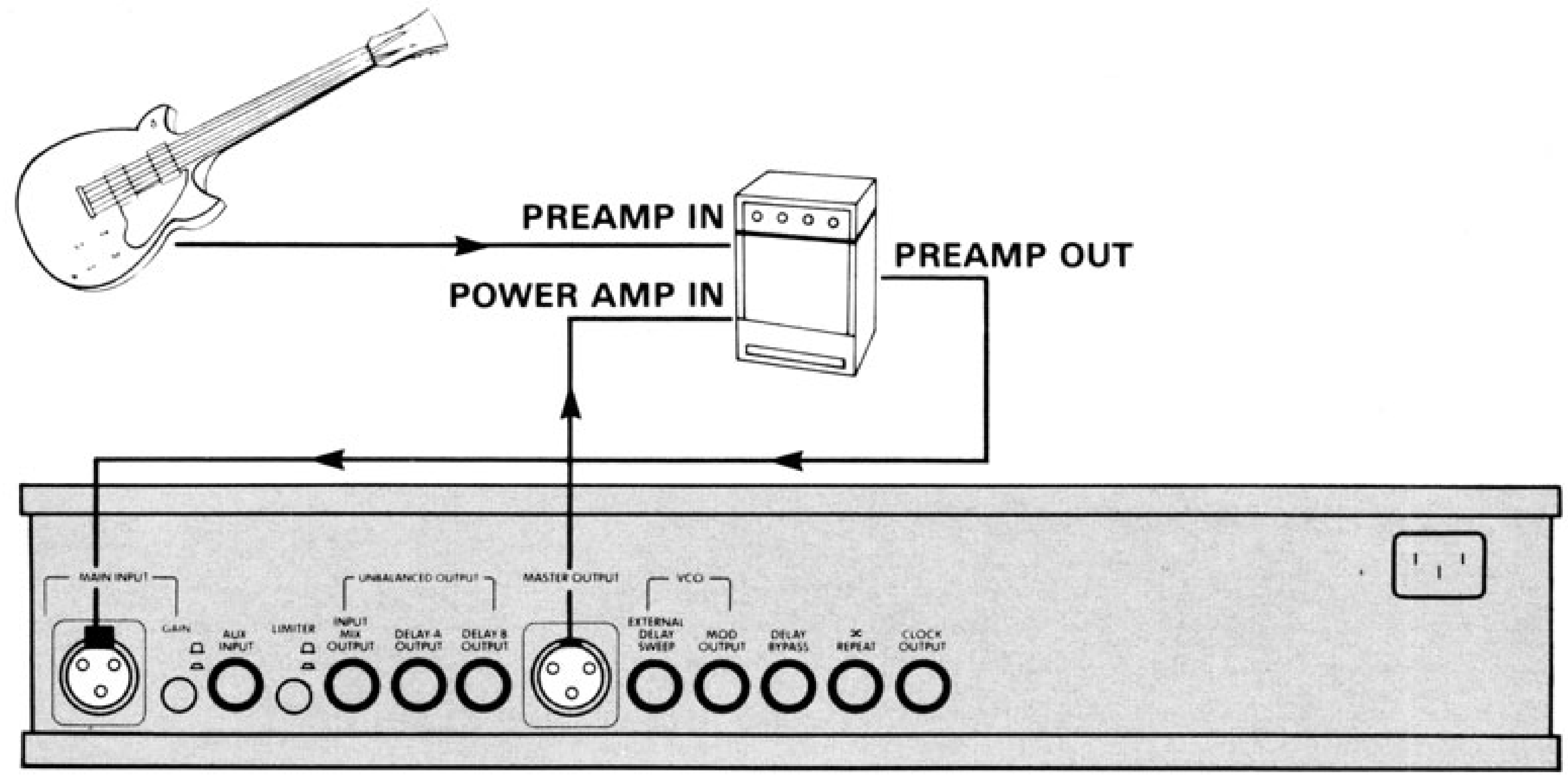


FIG. 2.10. Connection to guitar amplifier, using effects loop.

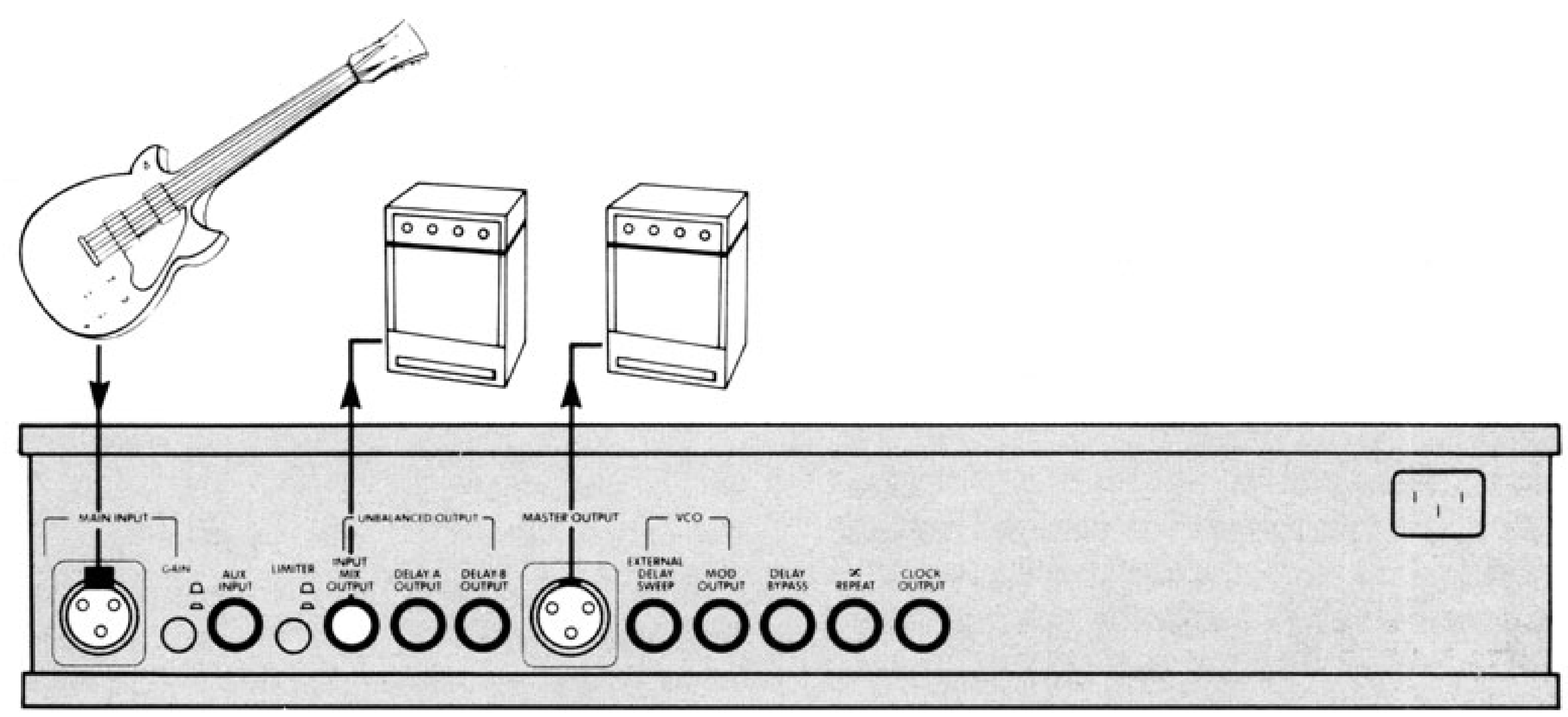


FIG. 2.11. Connection to dual amplifiers.

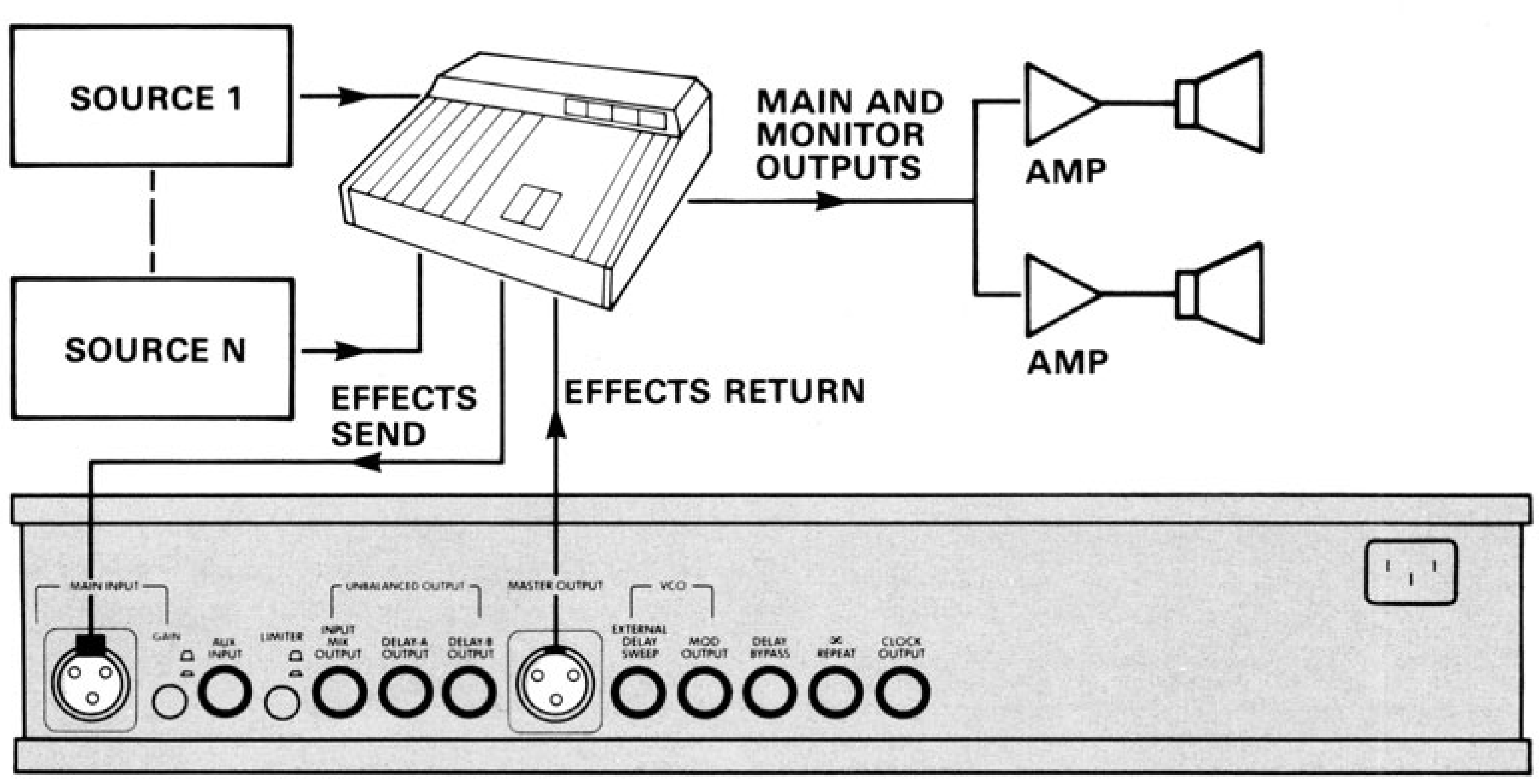
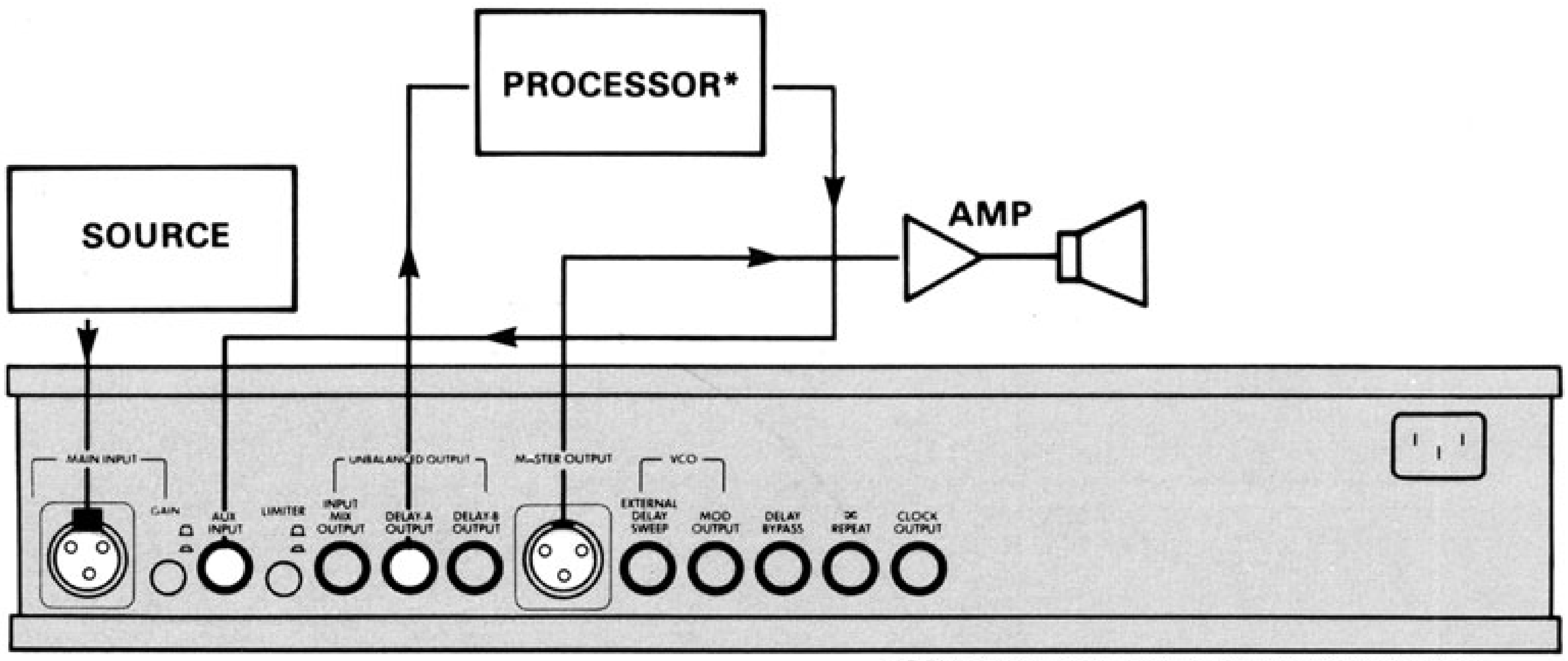


FIG. 2.12. Connection with mixing console.



*DELAY LINE, EQUALIZER, PITCH CHANGER, ETC.

FIG. 2.13. External processor connected in series.

3
Basic Operation—How To Get Started

3.0 Introduction

In this section of the manual, we provide a guided demonstration of the controls and functions of the Model 95. The novice or experienced user is led step by step through each section of the 95. More general information on selected applications is found in Section 4.

As we introduce the various functions, we illustrate each demonstration with front-panel setup sheets and block diagrams. Block diagrams are a common device for illustrating audio signal flow and processing—using lines to show the signal path, and symbols such as rectangles, circles, and triangles to represent delays, mixers, amplifiers, etc. In the course of our "tour," we develop the entire Model 95 block diagram, one piece at a time.

3.1 Front and Rear Panels

3.1.1 Initial Setup

To begin, place the Model 95 on a table or other support where you can conveniently look at the front and rear panels. Plug the AC power cord into the rear panel, and connect the unit to house current. Initially, set the front-panel controls as shown in Fig. 3.1.

Now, apply power to the unit by pressing the POWER switch (lower left corner of the front panel). All of the indicator lights on the panel, except the BYPASS LED, will light for a few seconds while the display shows "95 — —". The first two digits are the model number (95), and the second digit indicates the revision level (more about this in Section 7.6). The last digit indicates the memory option installed (Option 0, i.e., standard unit, = 1.92 seconds maximum delay; Option 1 = 3.84 seconds; and Option 2 = 7.68 seconds). After a few moments this display will disappear, to be replaced by the normal readout of delay times. **Note:** If an error indication appears ("Err" followed by a three-digit diagnostic code), refer to Section 7.2.

3.1.2 Control Groupings

3.1.2.1 Front-panel groupings

Take a moment to examine the layout of the front panel. As shown in Fig. 3.2, the array of controls breaks roughly into two groups: Delay Control and Audio Control. The left portion of the panel (Delay Control) has a number of rotary controls and momentary pushbuttons, with a display window containing six digits and several indicator lights. The right side of the panel (Audio Control) has ten slide controls (in two groups of five), a seven-step level indicator, a few toggle switches, and a couple of separate indicator lamps.

Delay Control Section — Figure 3.3 shows the Delay Control section. This group of controls selects delay time, modulation functions, and bandwidth (extended delay) and enables the Infinite Repeat mode and the

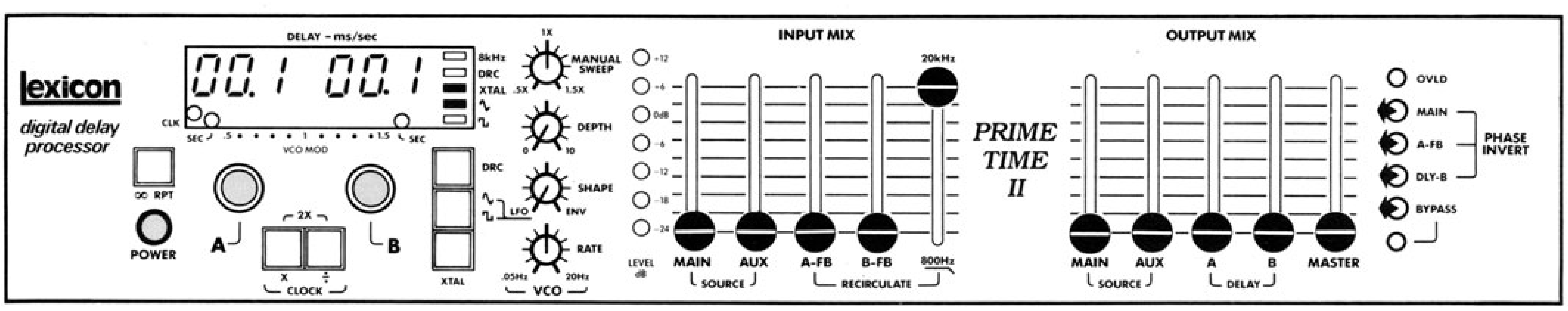


FIG. 3.1. Initial front-panel setup.

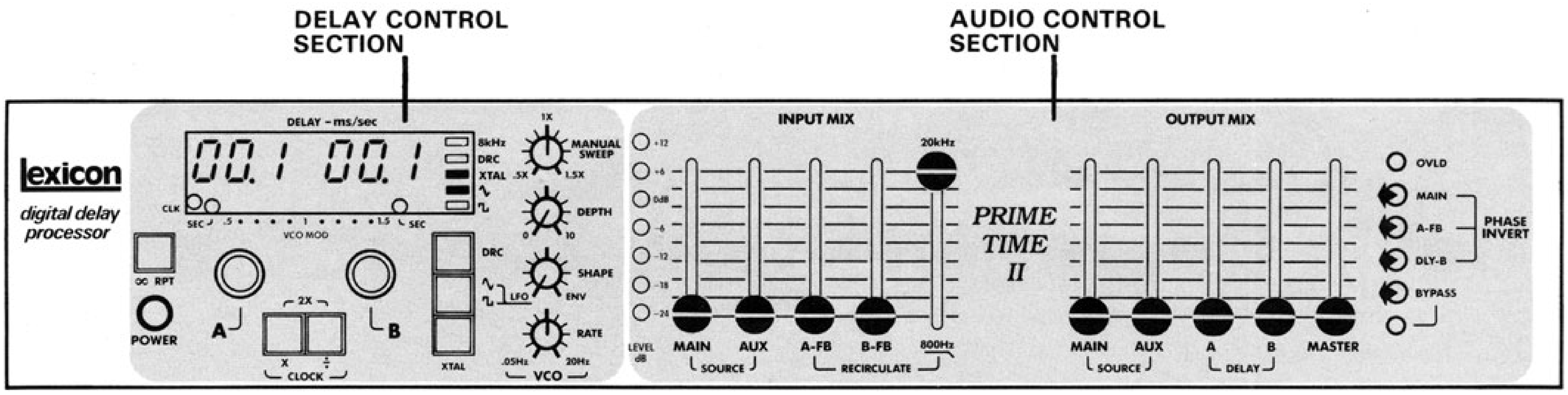


FIG. 3.2. Front-panel control groupings.

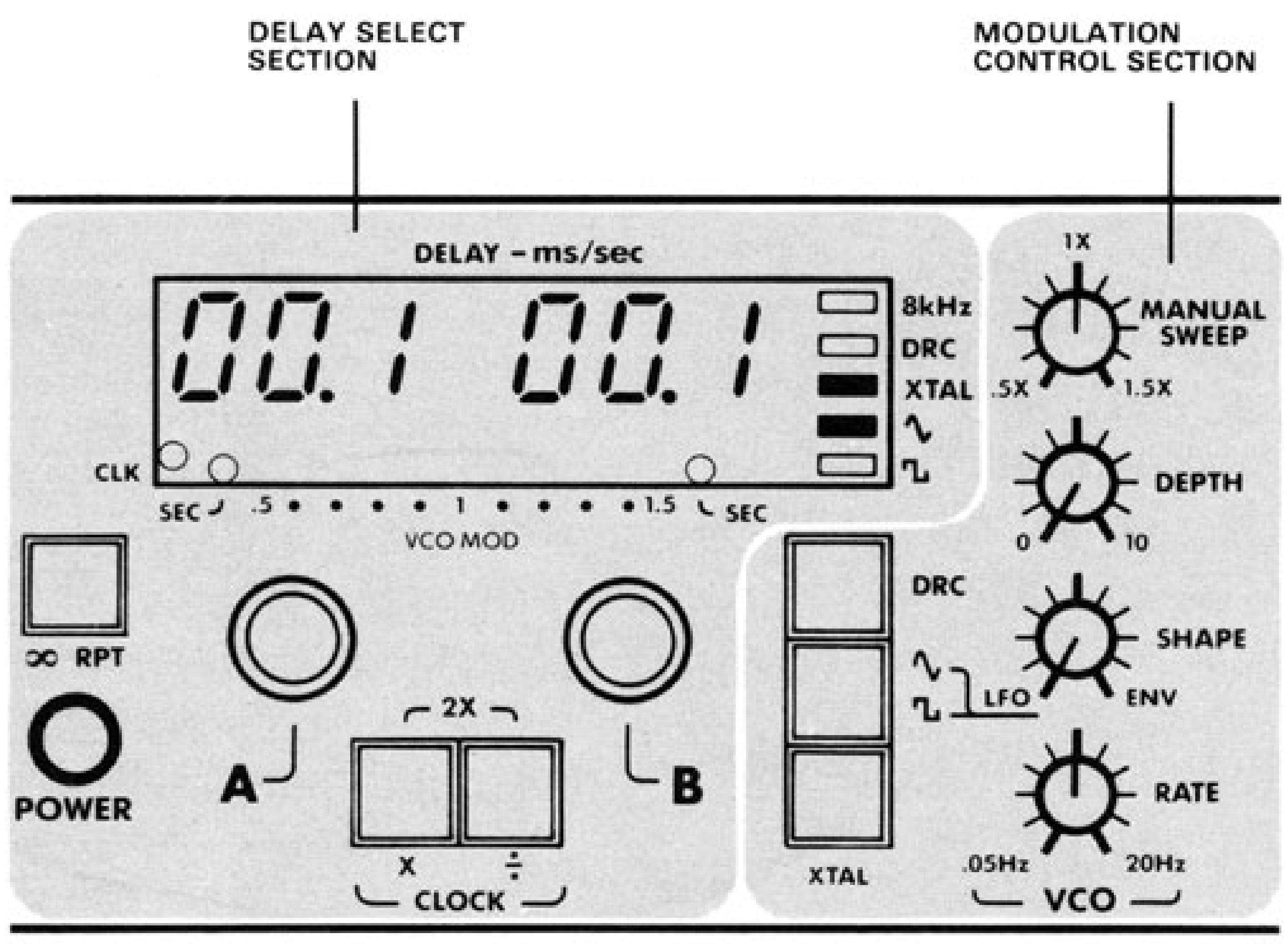


FIG. 3.3. Delay control section.

Clock functions. The slide controls and their associated functions in the Audio Control section regulate the flow of audio signal within the device.

The Delay Control section can be broken down into two smaller divisions: Delay Select and Modulation Control.

The Delay Select section has two large knobs that control individual delay times for Delay-A and Delay-B. In addition, this section contains pushbutton switches that control the Infinite Repeat function, delay/bandwidth range, and the programming of the Clock Output function. It also includes the POWER switch.

The Modulation Control section is discussed in Section 3.2.6.1, "Delay Modulation," after more fundamental information about the delay line has been presented.

Audio Control Section — The Audio Control section is conveniently divided into the Input Mix and Output Mix sections, as indicated by the panel graphics (see Fig. 3.4). The Input Mix section, as the name implies, combines signals to be input to the delay line (including recirculated Delay-A and Delay-B signals), and regulates their levels. This section also permits adjustable equalization of the recirculated Delay-A and Delay-B signals and includes indicators that show the level of the signal at the Input Mix stage.

The Output Mix section controls the relative level of signals (as well as the overall level) at the MASTER OUTPUT. Control of the polarity (phase inversion) of

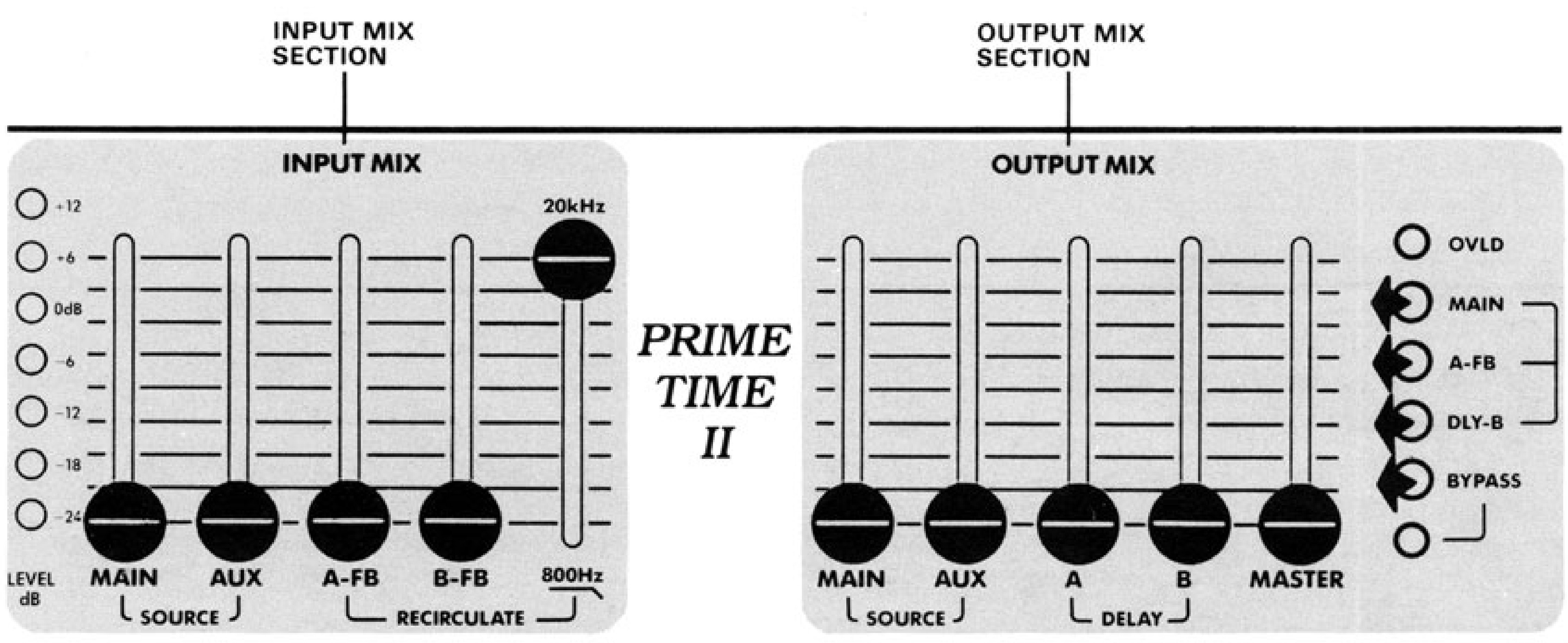


FIG. 3.4. Audio control section.

certain signals is provided by three toggle switches. One of these switches (A-FB) actually affects the Input Mix section, but is grouped with the other phase switches for convenience. A toggle switch is also provided to bypass the delay line and route "dry" signal (direct source signal that is not delayed by the 95) directly to the MASTER OUTPUT. Note that the Output Mix section affects only the MASTER OUTPUT.

Now go to the rear panel of the unit and examine the connections from left to right.

3.1.2.2 Rear-panel groupings

Input Group — As shown in Fig. 3.5, the first two connectors on the left and the two pushbuttons are audio inputs with associated control functions. The female XLR connector is the main input to the unit. The pushbutton switch next to it can enable a 20-dB gain in the MAIN INPUT so that it can be used with either microphone, instrument, or line-level signals. The 1/4″ phone jack labeled "AUX. INPUT" is a second audio input that is very useful for interconnecting the Model 95 with equalizers or other signal processors. This input accepts line-level signals only. The pushbutton to the right of this jack can engage a two-stage compressor/limiter in line with both audio inputs. This function is extremely useful to prevent audible distortion if the input to the A/D (analog/digital) converter is inadvertently overloaded.

Output Group — The Output Group of rear-panel jacks carries all of the audio signal outputs from the 95. The INPUT MIX OUTPUT carries the combination of signals developed by the Input Mix section of the front panel. This is a line-level signal, unbalanced, with a 600-Ω source impedance. The next two jacks carry unmixed outputs from Delay-A and Delay-B. These are unbalanced line-level outputs. The male XLR connector labeled "MASTER OUTPUT" represents the main output of the 95. This electronically balanced output carries the mix of signals created by the Output Mix section of the front panel, at the level set by the MASTER level control.

Control Input and Output Section — The remaining jacks are grouped together as the Control Input and Output section. These jacks are used for interfacing foot controls to the unit, as well as for interfacing the Model 95 with certain types of other devices. The first two jacks in this group are labeled "VCO" and are used to remotely control sweep functions. The next two jacks control the Bypass and Infinite Repeat functions. The last jack carries a pulse-train output that can synchronize other devices to the 95's delay loop.

3.2 Using the Model 95

3.2.1 Setting Up

Now we are ready to hook up the 95 and make some sound! As shown in Fig. 3.6, connect the sound source of your choice to the MAIN INPUT connector. (If you do not have the right cable or cable/adapter, refer to Section 2 for wiring instructions.) Connect the MASTER OUTPUT to the input of your monitoring system or amplifier, and adjust to your accustomed level.

Check the rear panel to see that the GAIN pushbutton is "out" and the LIMITER pushbutton is "in"; then play your source material into the 95. Slowly raise the MAIN slider in the Input Mix section until you see activity on the LEVEL dB indicator lamps. Raise the input until the average level is around 0 dB with occasional peaks of +6 or +12 dB. If you can't get to this level, reach around the back and press in the GAIN switch, and then set your level.

Now, go to the Output Mix section and raise the MAIN slider to the sixth gradation line from the bottom. Go to the MASTER slider and raise it slowly until you have a satisfactory listening level, without overloading the monitoring input. The front-panel setup and block diagram we have so laboriously established is shown in Fig. 3.7.

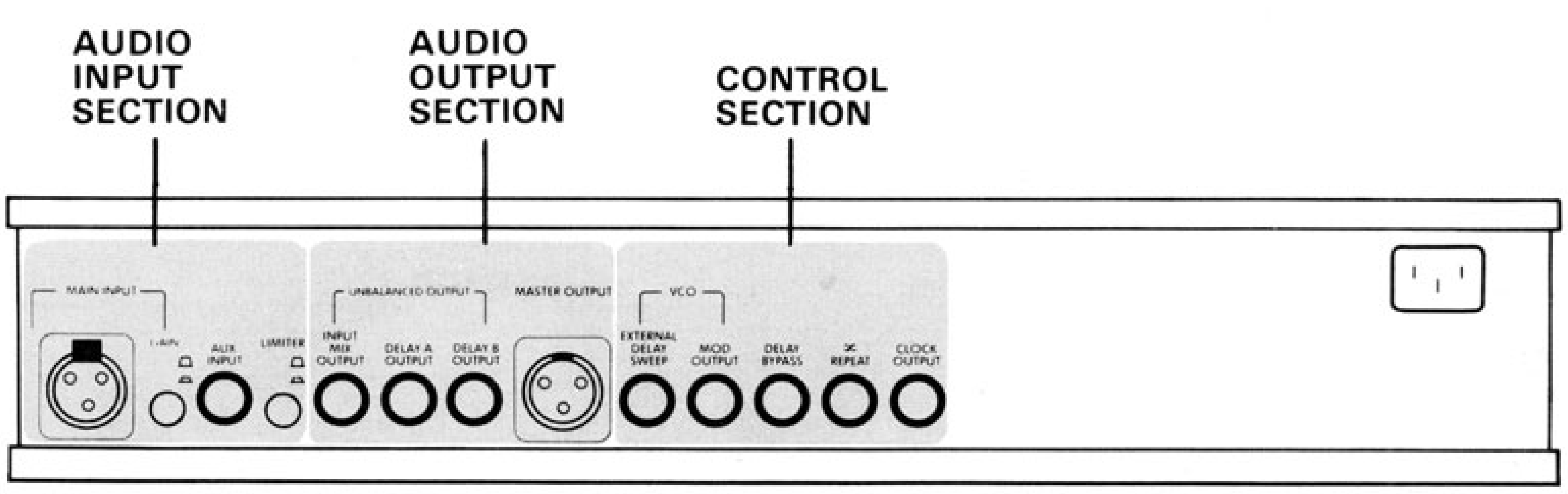


FIG. 3.5. Rear-panel jacks and controls.

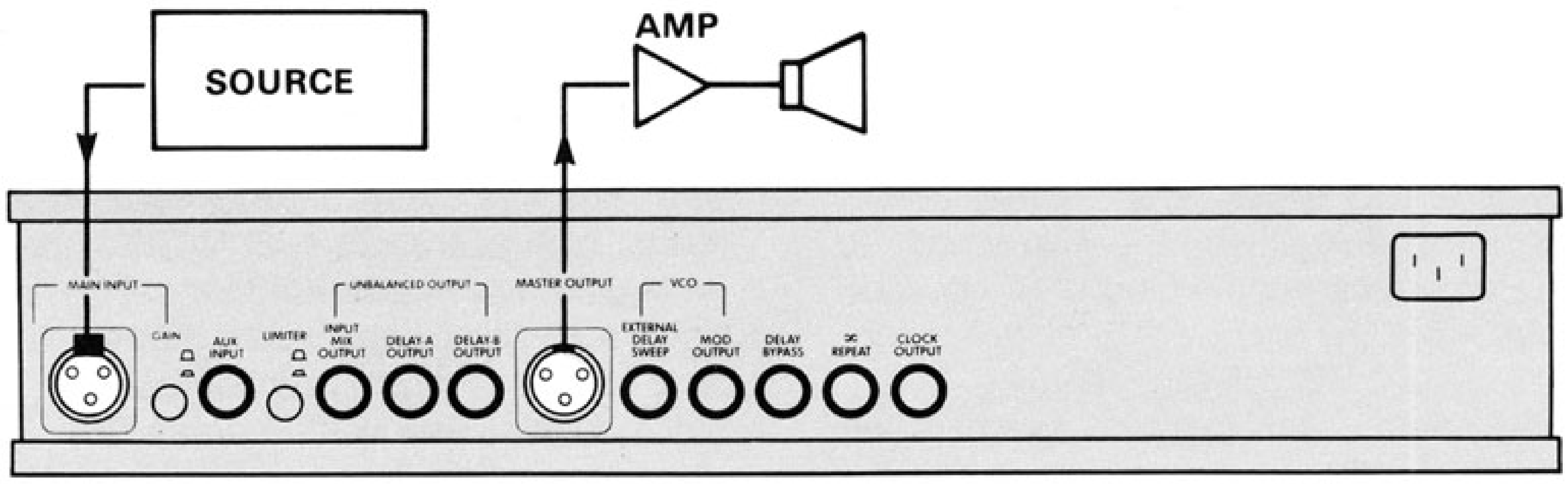
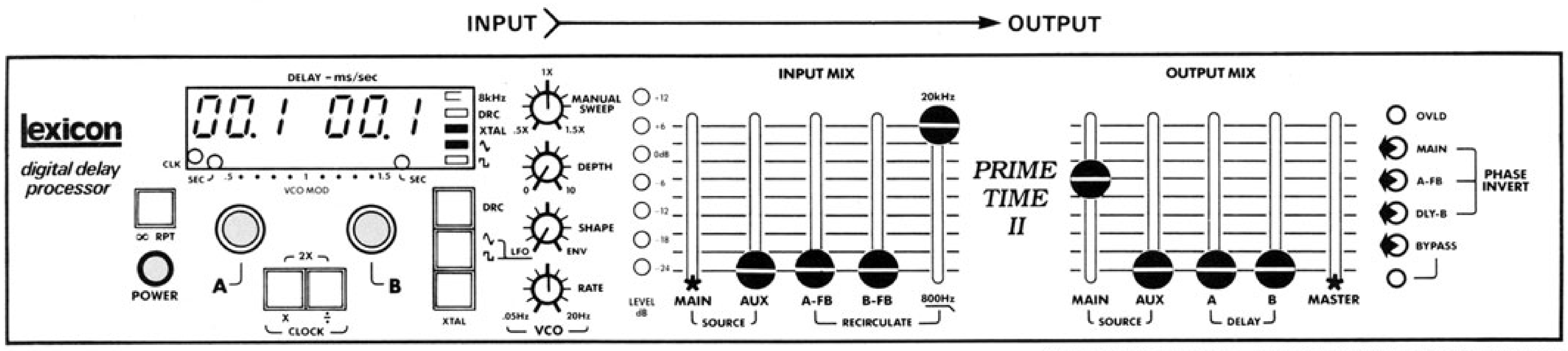


FIG. 3.6. Basic hookup.



*ADJUST LEVEL APPROPRIATE TO INPUT SOURCES AND OUTPUT DEVICES USED.

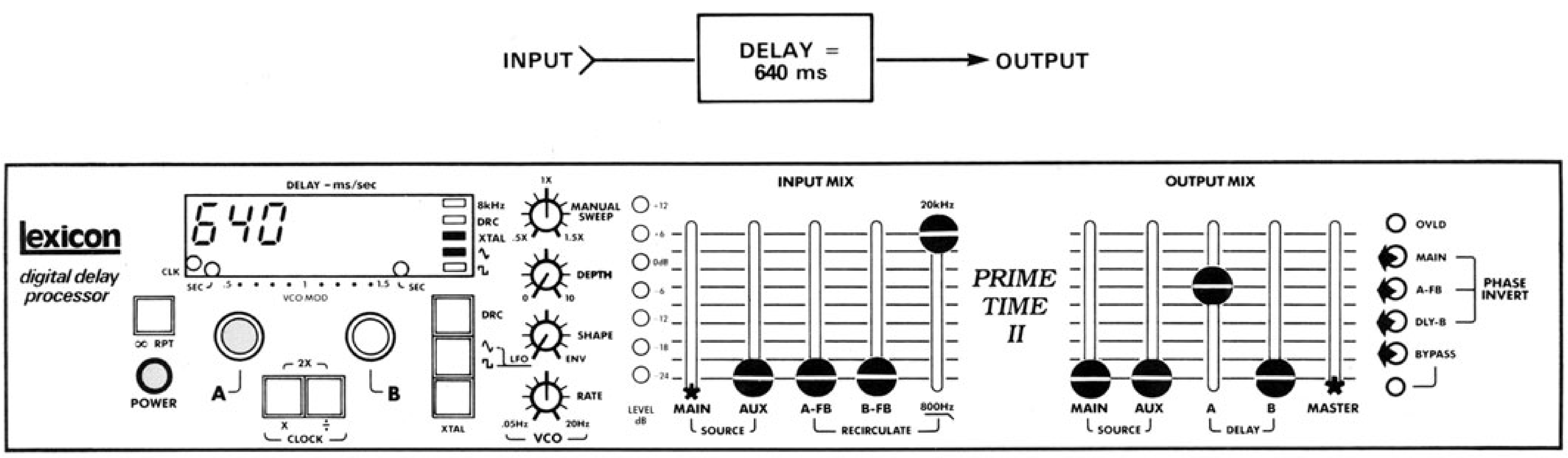
FIG. 3.7. Straight wire.

So far, not very impressive. Now go to the two Delay Select rotary controls and adjust the A control for 640 milliseconds as shown on the digital display. Go to the Output Mix section again, lower the MAIN slider all the way down and raise the DELAY A slider to the sixth line. What you hear now should be essentially identical to the dry signal you heard before. This is a simple delay, which when represented as a block diagram looks like Fig. 3.8.

3.2.2 Simple Delay

Actually, we have level controls in this setup. With these added, the block diagram becomes Fig. 3.9.

Raise the MAIN slider again to the same level as the DELAY A slider. You should now hear the original signal followed by a distinct echo of itself. Rotate the Delay Select A control slowly counterclockwise and listen to



*ADJUST LEVEL APPROPRIATE TO INPUT SOURCES AND OUTPUT DEVICES USED.

FIG. 3.8. Simple delay.

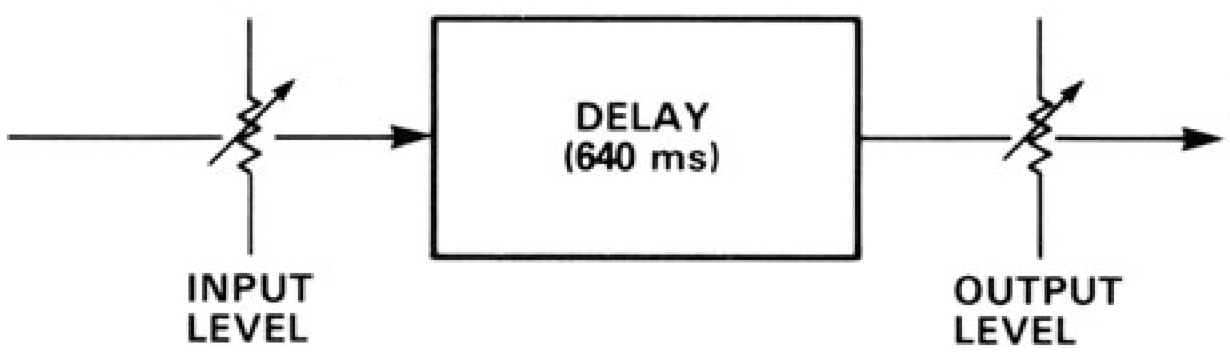


FIG. 3.9. Simple delay with level controls.

the change in delay time, then return to the original position. If your unit has the standard memory, the 640 milliseconds will be the far extreme position of the knob. If you have Memory Option 1 or 2 installed, the position will be somewhere in the middle of the dial and you will have more delay beyond that point (1.28 seconds total delay for Option 1; 2.56 for Option 2). Now we have established the setup shown in Fig. 3.10.

We can also recirculate the delayed signal back into the input by first raising the "800 Hz–20 kHz" slider in the Input Mix section to the top (more on this later), then raising the A-FB slider to the sixth line from the bottom. Now you should hear several repetitions of the input, fading out over time. If this is not completely apparent with the source you are using, cut the input off suddenly and you will hear the decay very clearly. At this point we have established a more complex block diagram with complete control of several parameters of the sound. This is the classic arrangement of a simple echo unit. By varying the delay time, the blend of direct and delay signals, and the amount of delay recirculation, quite a broad range of sounds can be achieved. Our block diagram and panel setup now looks like Fig. 3.11.

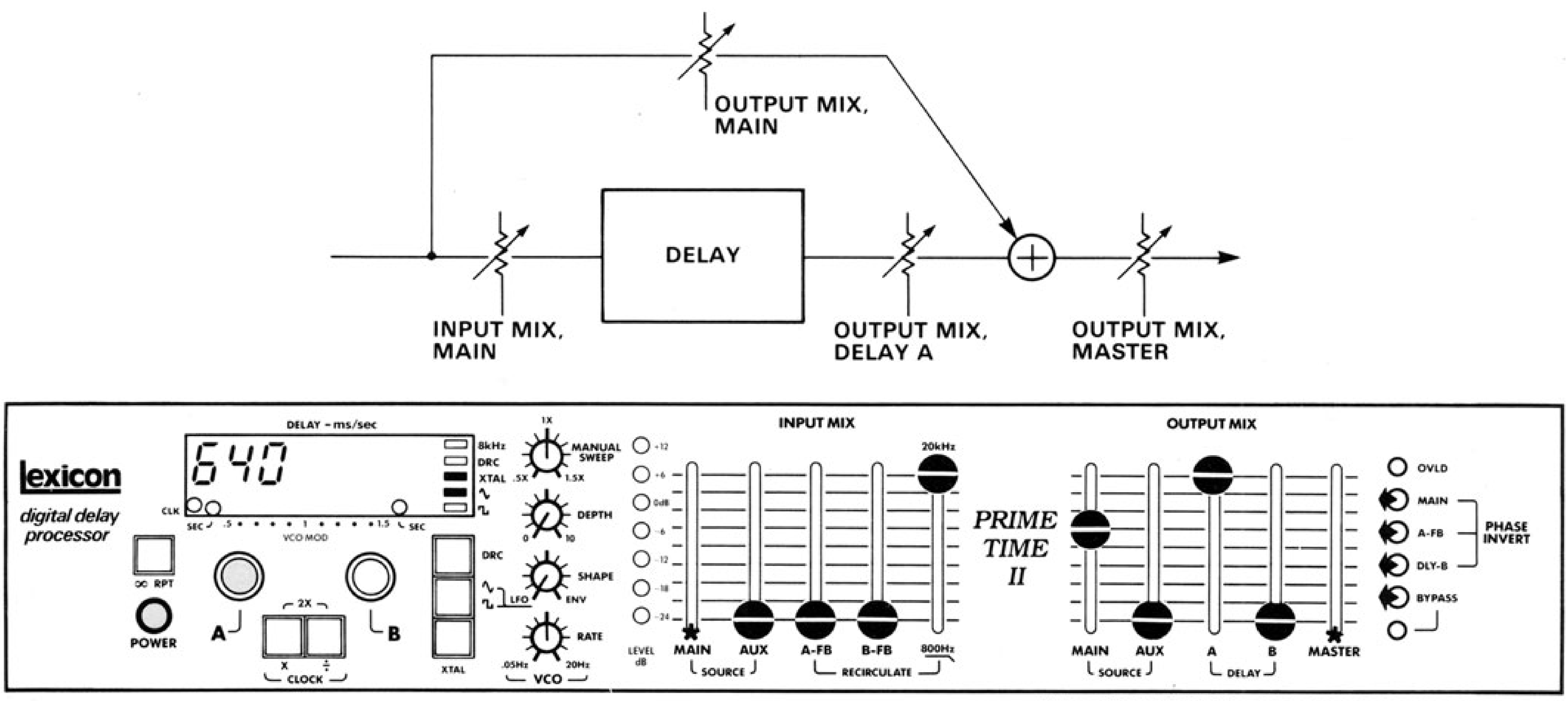
Now that we know what level controls are in the setup and their functions, we can clean up our block diagram by removing them as shown in Fig. 3.12.

At this point, take some time and experiment with the different settings of the controls we have discussed so far. Be systematic, starting with very small settings of delay time and varying the output blend and the recirculation over their full range. Do this at a number of delay settings from minimum to maximum, and you will form a reasonable idea of the range of sounds available from this simple setup. All of the effects setups we discuss in this manual have this basic configuration at their core, with additional features applied in different ways to achieve the full range of delay effects.

3.2.3 Dual Delay

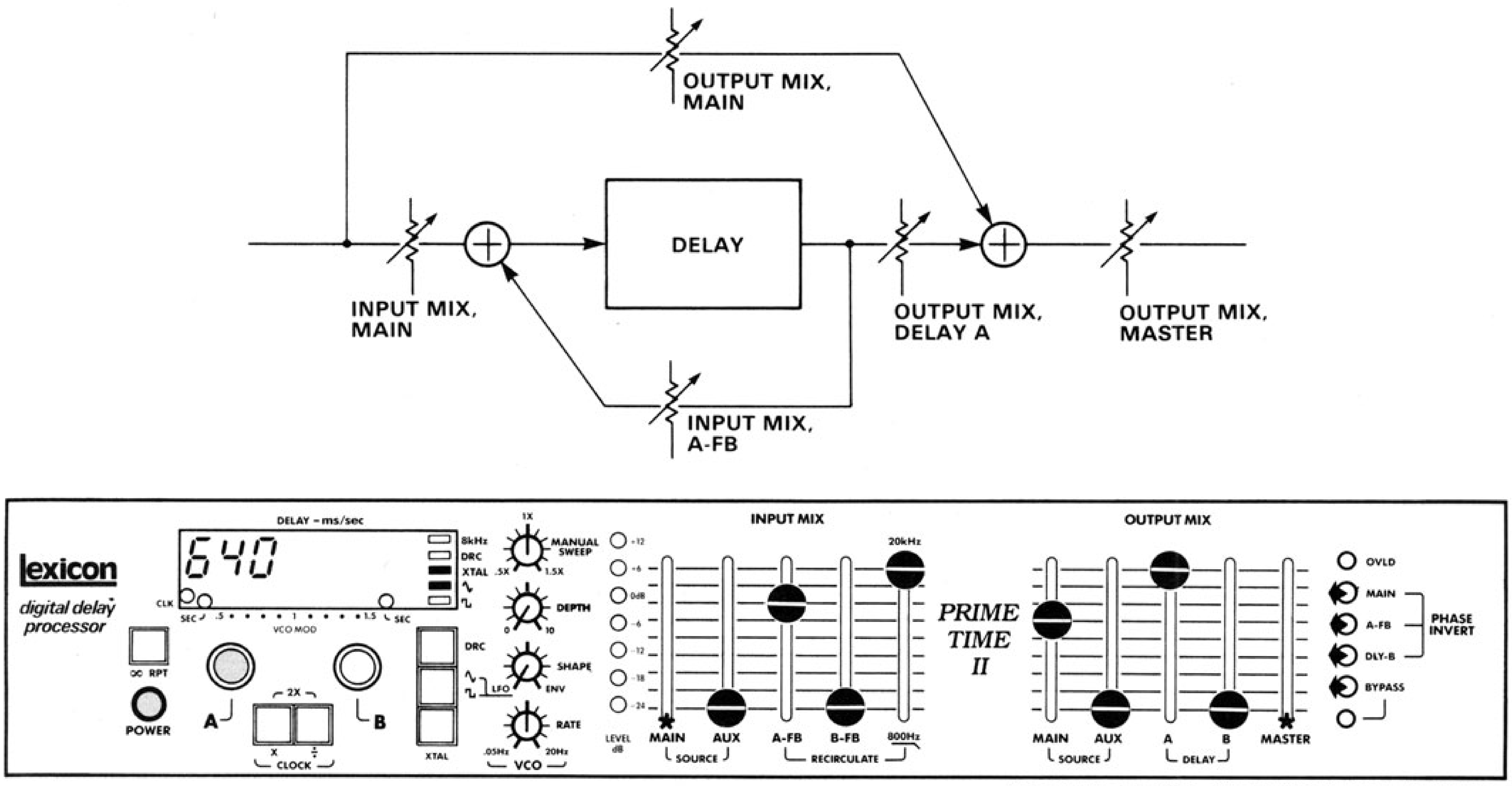
Now we can fill in the picture of the audio path and the controls of the Model 95. A key feature of the Model 95 Prime Time II is its dual delay outputs. This means that two separate delay times can be set up, and mixed into the Input and Output Mix sections. This capability allows the user a great degree of flexibility in setting up audio effects. To experience the dual-delay feature, set up the front panel as shown in Fig. 3.13.

Listening, you can hear the two different repeats at separate times. Vary the settings of the Delay Select B rotary control and listen to the changes in the relative delay times, then return to the 180-millisecond position. Now, lower the DELAY B slider in the Output Mix sec-



***ADJUST LEVEL APPROPRIATE TO INPUT SOURCES AND OUTPUT DEVICES USED.**

FIG. 3.10. Delay with Output Mix.



***ADJUST LEVEL APPROPRIATE TO INPUT SOURCES AND OUTPUT DEVICES USED.**

FIG. 3.11. Standard delay with Output Mix and recirculation.

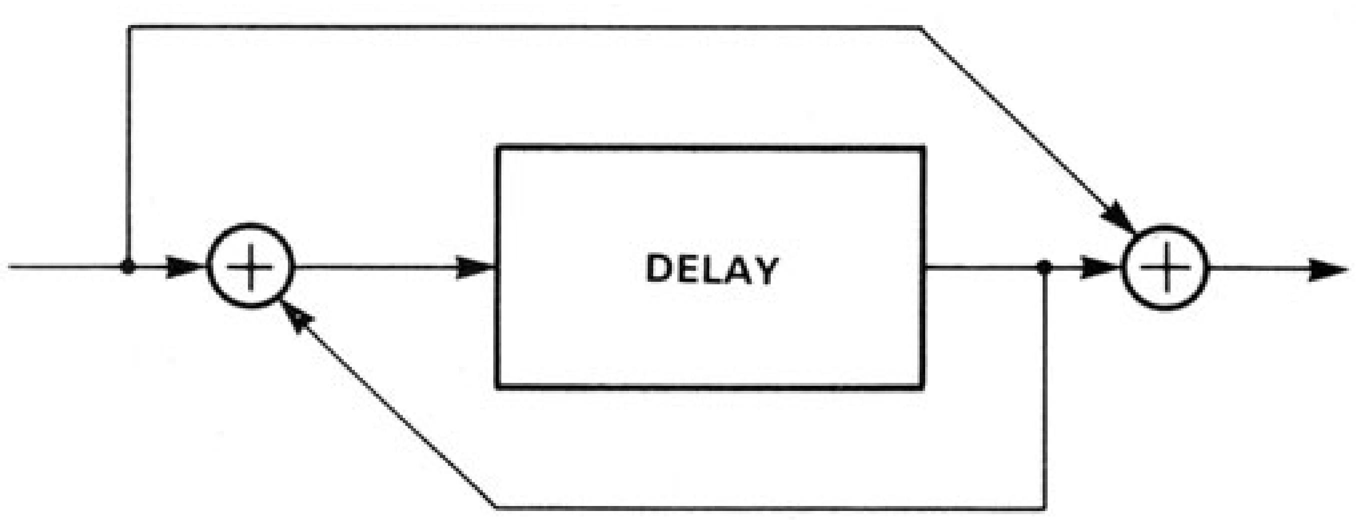
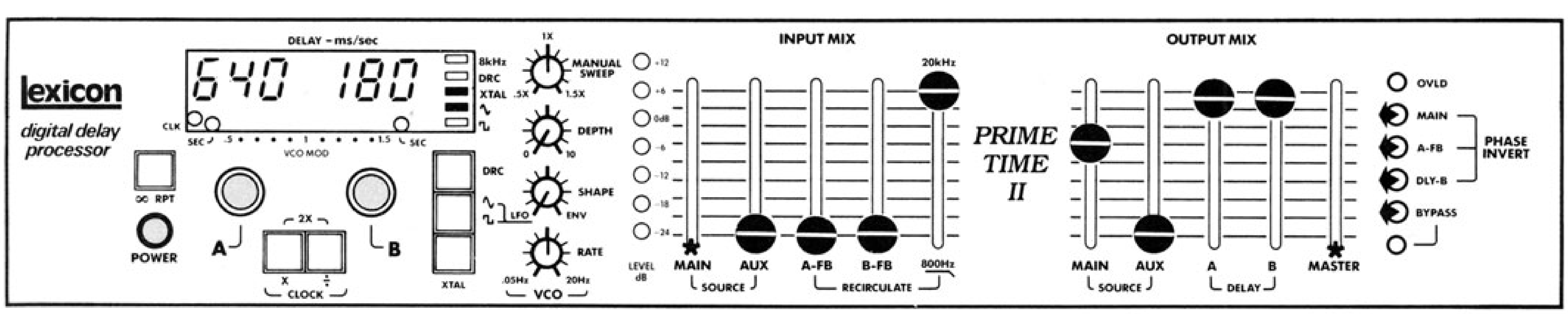


FIG. 3.12. Standard delay.

tion all the way down and raise the B-FB slider in the Input Mix section to the sixth line. The effect now is a delay of 640 milliseconds before the first repeat with several repeats spaced at 180 milliseconds following it. This is just one possible use of the two delay taps. Now our block diagram looks like Fig. 3.14.

Now, *very slowly* raise the A-FB slider in the Input Mix section. A point will be reached at which the delayed audio no longer fades out, but increases in level until the input overloads. This "runaway" results when the total gain of recirculation for Delay-A and B together exceeds a value of "1." Whenever you use A-FB and B-FB together, be very cautious that you do not inadvertently produce runaway. Use of the "800 Hz–20 kHz" slider can reduce the potential for runaway by filtering out many high-frequency components whose contributions to the total gain could result in a value exceeding unity gain.



***ADJUST LEVEL APPROPRIATE TO INPUT SOURCES AND OUTPUT DEVICES USED.**

FIG. 3.13. Dual-delay panel setup.

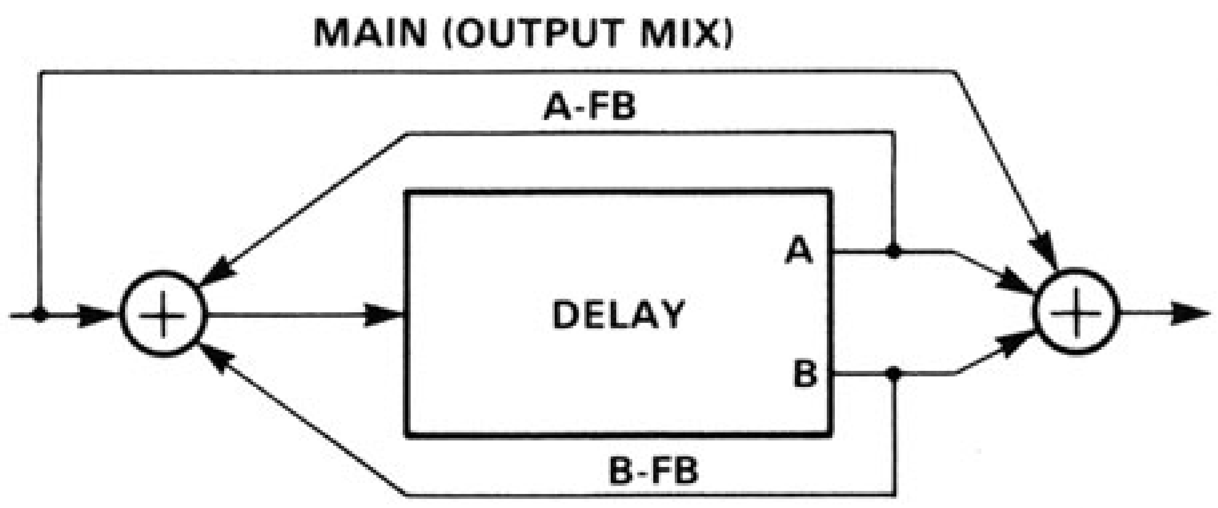


FIG. 3.14. Dual-delay block diagram.

Take some time now to experiment with the delay times for A and B, the recirculation controls (observing the caution noted before) and the output blend. You will notice that the three controls we just introduced (B rotary control, B-FB, and DELAY B) increase the number of possible configurations to the point that the type of systematic exploration we recommended before would require a large investment of time. However, it is important that you make a point of doing this type of exploration regularly, working for a comfortable length of time, then picking up later where you left off. If you work slowly through all of the features that we have introduced here, your knowledge of the Model 95's possibilities will grow enormously. Even very experienced users can benefit from this concentrated exploration of individual features.

3.2.4 Phase Inversion, Bypass Mode, Recirculation Filter, and DRC

A few control functions in the audio path still require explanation. Look now at the group of toggle switches at the right-hand side of the front panel. The upper three of these can be used to switch the polarity of particular channels in the Input and Output Mix sections. A polarity (phase) invert function is symbolized in a block diagram by a triangle with a minus sign, and our block diagram makes clear where these switches are inserted in the signal path (see Fig. 3.15).

These functions mainly come into play when using short delay times for flanging, resonance, chorusing/doubling,

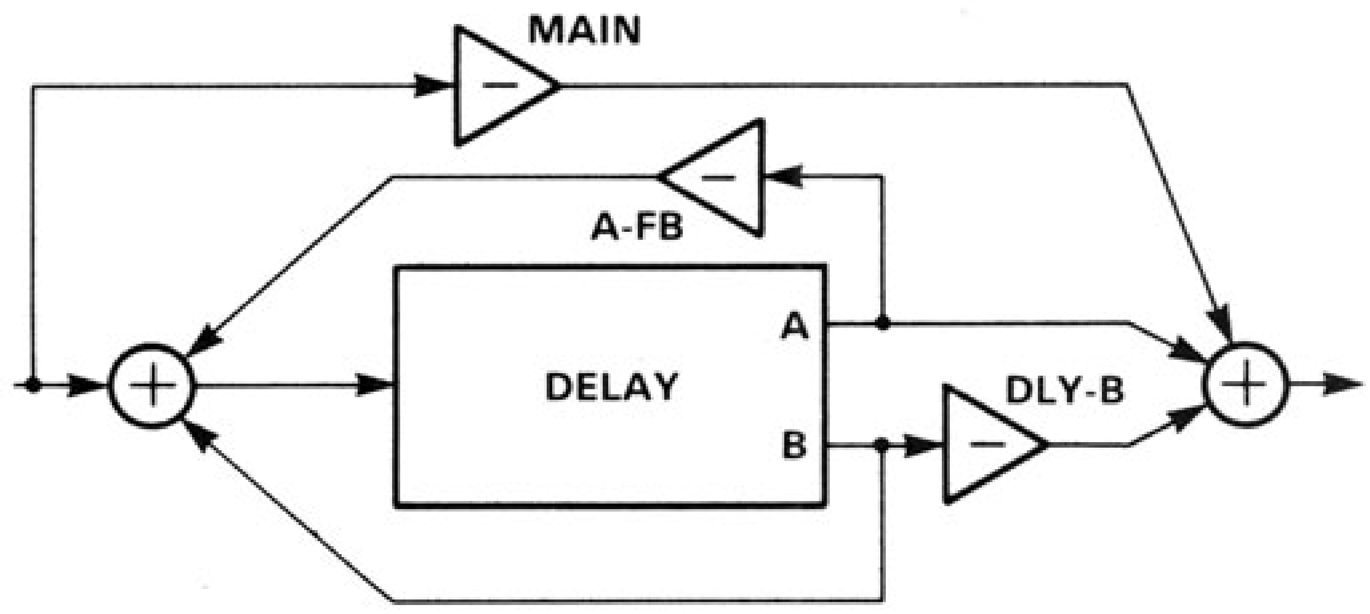


FIG. 3.15. Phase-invert block diagram.

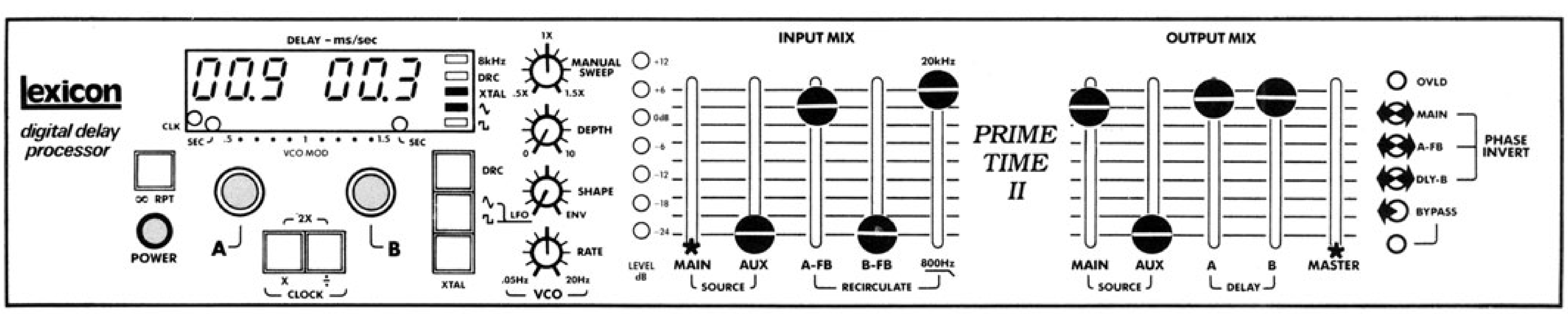
and other similar applications. To experience the effect these switches can have on the audio signals, position the front-panel controls as shown in Fig. 3.16.

This setup has a distinct effect on the source material, to the point of an extreme coloration. Now, toggle the PHASE INVERT switches one at a time, and then in each possible combination, listening at each point to the sound that is produced. You may wish to try this with different types of source material to bring out the qualitative changes in different ways. This type of control is used, though normally less extremely, to create different shades of coloration in flanging and chorusing setups as well as resonance effects like this one.

Now try the bottommost switch. This is used to completely cut out the processing, allowing the user to select when an effect occurs or to compare the processed and unprocessed sounds.

Additionally, you can equalize the high end of the recirculation path to temper the tonal characteristics. The "800 Hz–20 kHz" slider in the Input Mix section sets the cutoff frequency of a 6-dB/octave low-pass filter, as shown in Fig. 3.17 (note the BYPASS switch line as well).

Move the "800 Hz–20 kHz" slider slowly downward to hear the effect on the resonance setup from the last example. The sound will become progressively duller as



***ADJUST LEVEL APPROPRIATE TO INPUT SOURCES AND OUTPUT DEVICES USED.**

FIG. 3.16. Phase-invert demonstration.

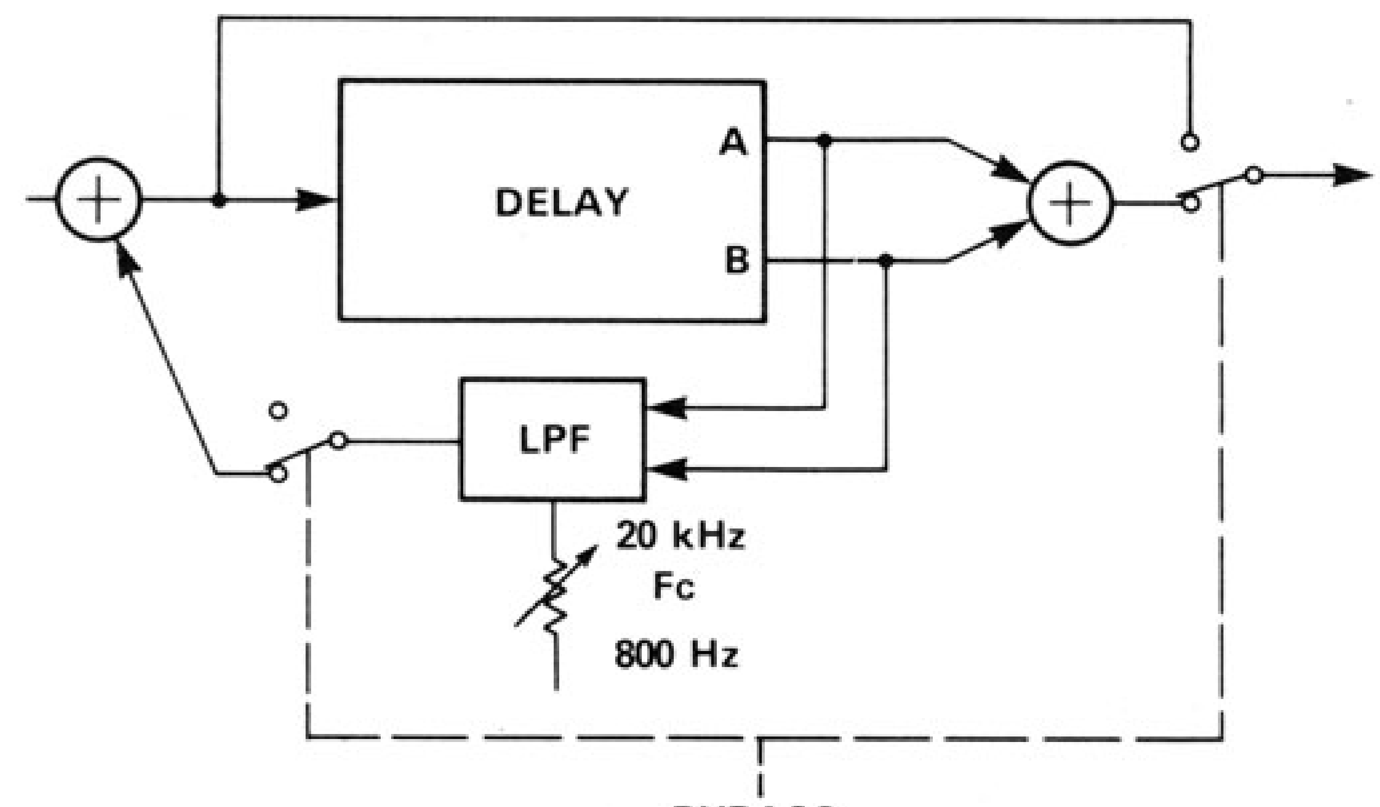


FIG. 3.17. Block diagram with recirculation filter and bypass line.

the control is lowered. Now, change the delay settings to 480 and 260 milliseconds, respectively, on A and B. Vary the cutoff frequency control and listen to the difference in the quality of the echo "tail." This type of control is often used in echo setups for more realism in simulating natural echoes, which have a tendency to lose some of their high-frequency components as they bounce and travel through the air.

There is yet another processing option that can be applied to the recirculation path. This function is referred to as Dynamic Recirculation Control (DRC). It has no adjustments and is simply engaged or disengaged by the pushbutton labeled "DRC." An indicator lamp in the display window lights to show when the DRC function is engaged.

The DRC function produces a dynamic effect in which the amount of recirculation allowed to pass through the A-FB and B-FB controls varies inversely with the level of signal input into the 95. This function allows the user to create echo setups in which the output remains comparatively dry and clear, but has long, multi-repeat tails between phrases. To experience this effect, set up the panel as shown in Fig. 3.18.

Now select input material such as a lead vocal or an instrumental solo that has a somewhat continuous flow with some distinct breaks between phrases. This type of material will bring out the desirable qualities of the DRC feature—that of clarity and articulation coupled with long, decaying echoes.

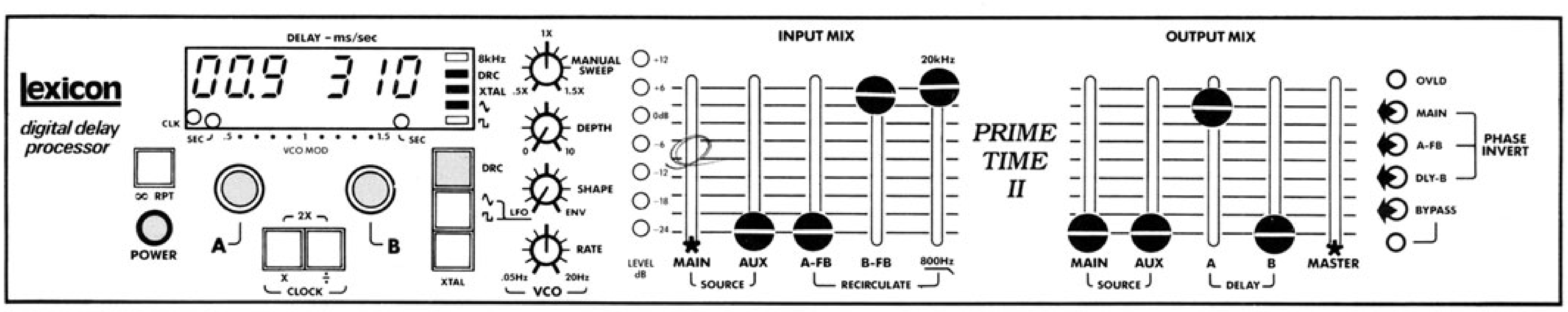
3.2.5 Auxiliary Input and Secondary Outputs

Besides the MAIN INPUT, the Model 95 has an auxiliary input. This separate input jack (labeled "AUX. INPUT") can be used to mix a second input signal into the delay line and/or the MASTER OUTPUT. This is very useful for interfacing the 95 to another delay unit or other types of audio processing gear, as shown in Fig. 3.19.

In this setup, the user can selectively blend the double-processed signal into the output, or recirculate it through the AUX slider in the Input Mix section. Note that this input is set up for line-level signals, and cannot be driven adequately by lower-level signals.

Besides the MASTER OUTPUT, the Model 95 has three secondary outputs. An INPUT MIX OUTPUT carries the combined input and recirculation signals established by the Input Mix section controls. This can be used for effective "stereo" outputs. If you have a second channel of sound available, set the controls as shown in Fig. 3.20 and connect the other speaker to the INPUT MIX OUTPUT. **Note:** This is a line-level signal only. If you are using a guitar amplifier or other system requiring a low input level, you must insert a pad into the line to protect the input.

The setup shown in Fig. 3.21 produces a long echo with a pronounced stereo "bounce." This type of connection is also very effective for adding a sense of dimension to a sound using shorter delays.



***ADJUST LEVEL APPROPRIATE TO INPUT SOURCES AND OUTPUT DEVICES USED.**

FIG. 3.18. Dynamic recirculation control demonstration.

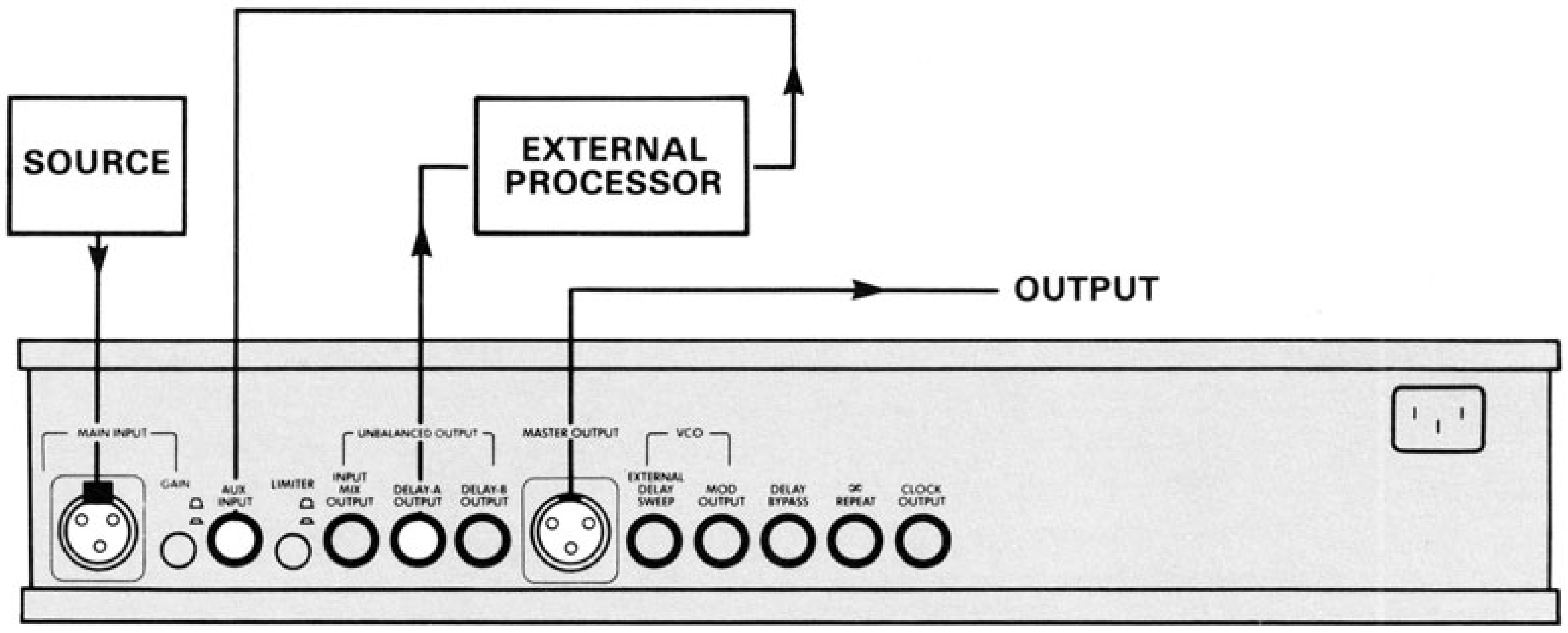
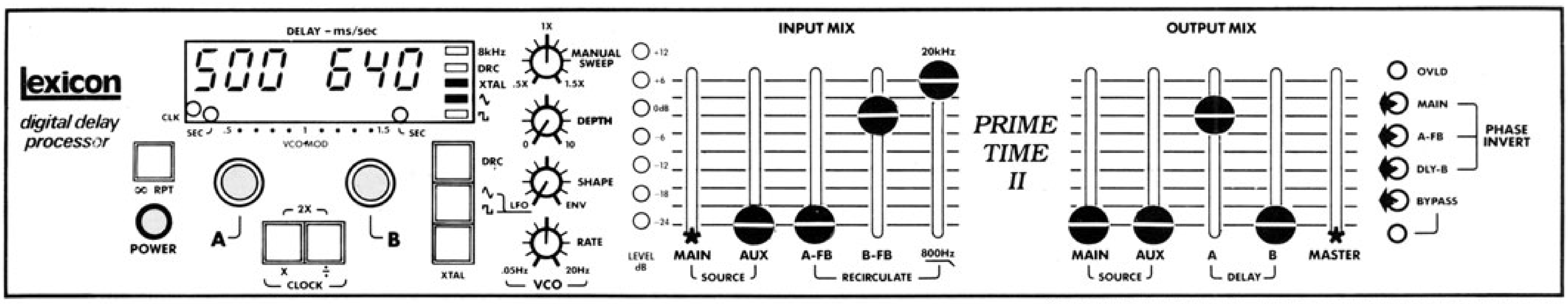


FIG. 3.19. Application of auxiliary input.



***ADJUST LEVEL APPROPRIATE TO INPUT SOURCES AND OUTPUT DEVICES USED.**

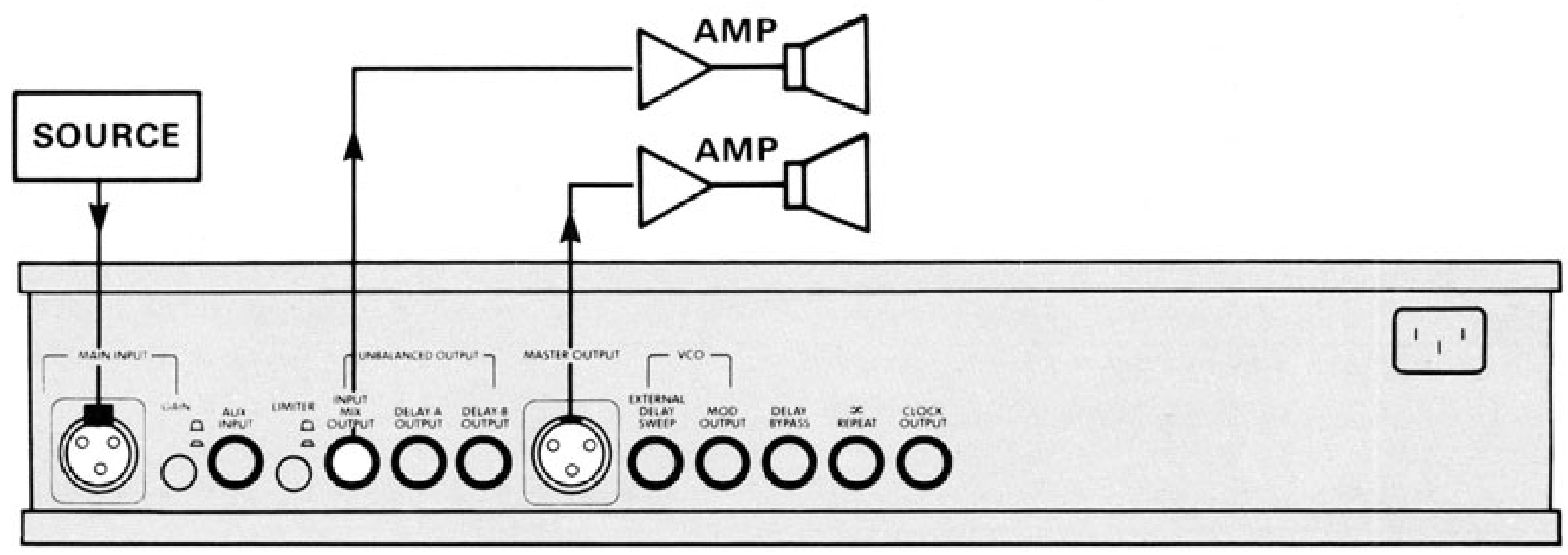
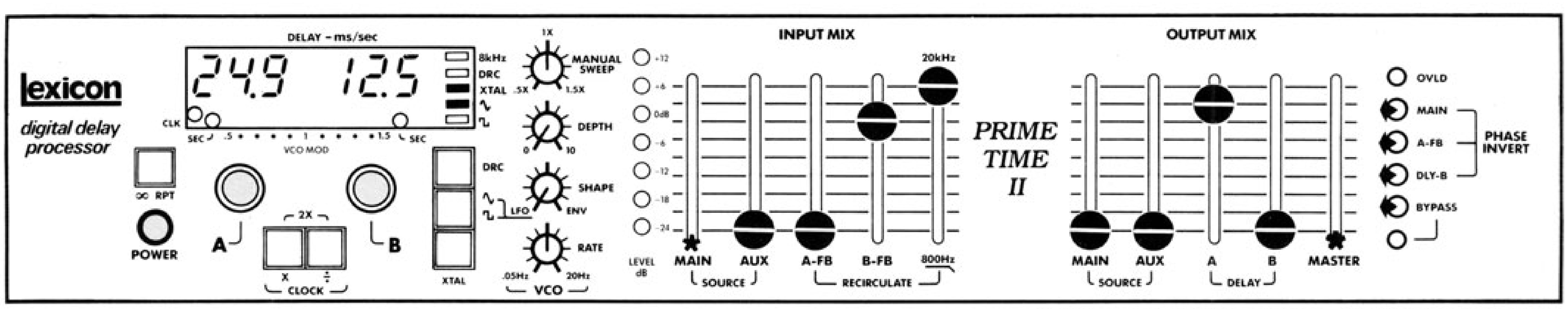


FIG. 3.20. Application of INPUT MIX OUTPUT.



***ADJUST LEVEL APPROPRIATE TO INPUT SOURCES AND OUTPUT DEVICES USED.**

FIG. 3.21. Stereo enhancement.

The DELAY-A OUTPUT and DELAY-B OUTPUT carry the unmixed outputs of the respective delay settings at line level. One might normally connect these outputs when using the Model 95 with a mixing console for full control of a stereo mix.

We have now completed the full block diagram of the audio, or "analog," signal path in the Model 95 Prime Time II (see Fig. 3.22).

We can delete many of the secondary functions from the diagram, unless we are specifically making use of a particular feature. Our basic diagram of the audio signal path can therefore be simplified as shown in Fig. 3.23.

3.2.6 Delay-Line Controls

Up to this point, we have chiefly explored the path of the audio signal around and through the delay line itself, while regarding the delay line as a relatively simple "block" of delay with an input, two outputs, and adjustments only for the delay time of each output. In this section, we develop an understanding of the remainder of the delay-line controls and how they affect operation of the entire unit.

Infinite Repeat — The Infinite Repeat feature is one of the most fascinating functions of a digital delay device, particularly when coupled with the advanced metronome clock feature of the Model 95. The Infinite Repeat function is triggered whenever the ∞ RPT pushbutton is engaged. The segment of sound that is captured is equal to the entire contents of the delay memory at that point in time, so the portion captured is that which entered the line just *before* the Infinite Repeat pushbutton was pressed. The length of the segment is *not* affected by the setting of the delay rotary controls, but is equal to the maximum delay time that can be selected by them. These delay times equal 640 milliseconds, 1.28 seconds, or 2.56 seconds, depending on the memory option installed. Later we will see how other periods can be selected. Set the controls for a simple long delay as shown in Fig. 3.24.

Play some material into the 95, and press the Infinite Repeat pushbutton. A section of sound will be trapped in memory and will continue to repeat over and over again. Adjust the delay time and note that there is no change in the length of the delay segment. Now raise the DELAY B output fader, and select a Delay-B time different from Delay-A. Notice the staggered, syncopated effect of listening to the two taps. Turn the Delay-A rotary control fully clockwise, raise the A-FB slider to the fourth line and press the Infinite Repeat pushbutton again. The segment captured will be released and will decay away at the rate set by the level of recirculation. We will return to the Infinite Repeat mode later in connection with other features.

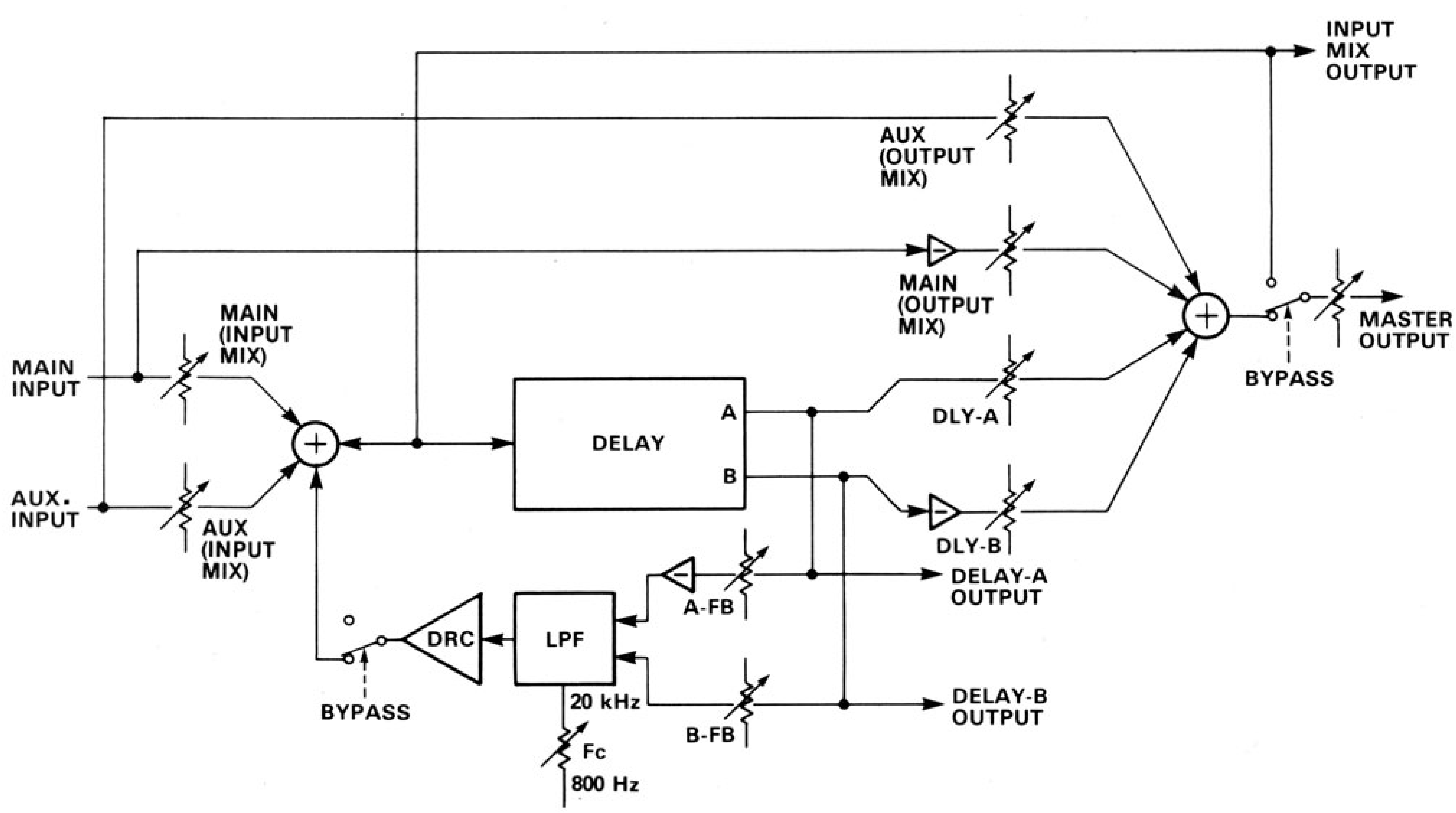


FIG. 3.22. Model 95 audio block diagram.

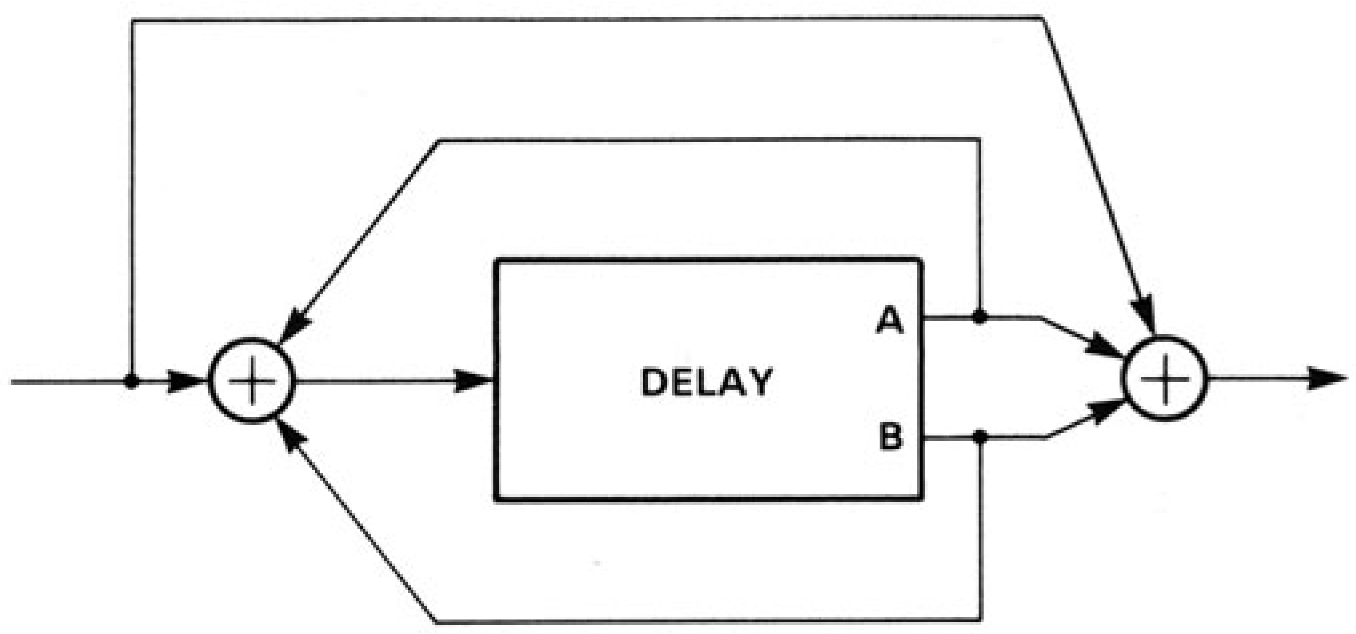


FIG. 3.23. Simplified audio block diagram.

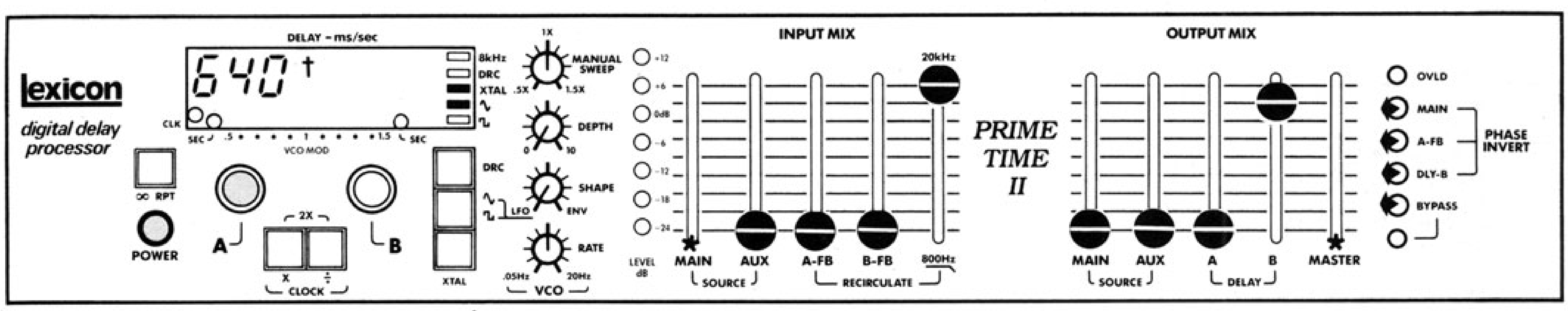
2X/CLOCK Pushbuttons — The 2X/CLOCK pushbuttons serve dual functions. For now, we discuss only the application of these pushbuttons to select two different delay/bandwidth ranges of the Model 95. Set the unit up for a simple long delay setting as shown in Fig. 3.25.

Now press the two 2X/CLOCK pushbuttons simultaneously for X2 (or double-delay mode), while monitoring the audio through the unit. You will hear a shifting of pitch down one octave, which will resolve into the normal pitch of the input signals in a few seconds. You will see that the delay indication has now doubled to 1.28 seconds. Depending on the source material, you will likely also hear the reduction of bandwidth down to 8 kHz.

This is the meaning of the indication of the 8 kHz lamp in the display window. This mode of operation is intended to provide more delay time for the amount of memory installed by passing data through at a slower rate than normal. This also means that the higher frequencies cannot be reproduced properly, hence the reduction in bandwidth. In this mode, you now have 1.28, 2.56, or 5.12 seconds of delay available, depending on the memory option. If you try the Infinite Repeat function now, you will find that the segment captured is twice as long as before.

3.2.6.1 Delay modulation

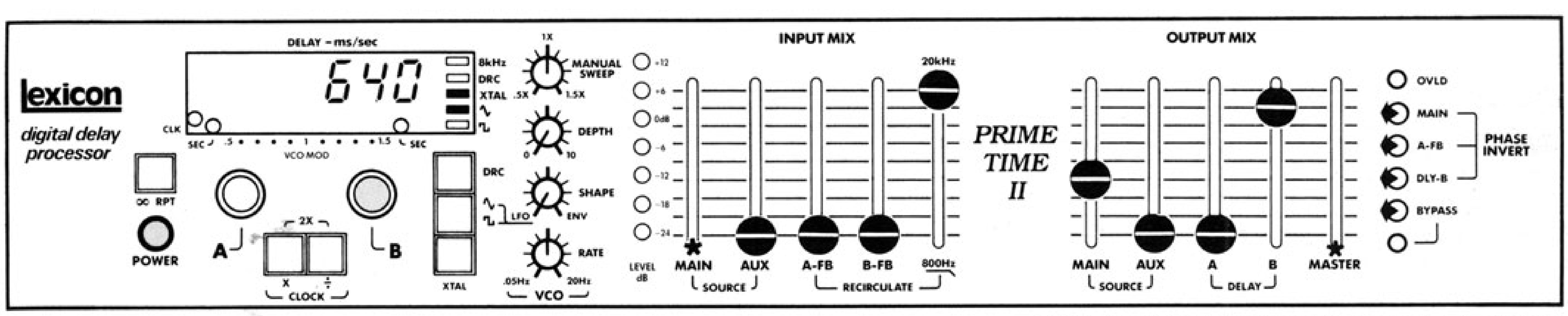
Up to this point, we have used the rotary Delay Select controls (Delay-A and Delay-B) to change delay time, with the X2 function introduced last as a way of selecting delay/bandwidth range. There is another way of affecting delay time that has particular application for us. By using the controls in the Modulation Control section, we can alter delay time in a smooth, sweeping fashion. This is fundamentally different from selecting delay with the rotary controls. The Delay Select controls change the delay time in jumps as they address different portions of the delay memory. The Modulation Control section controls operate by affecting the rate at which data moves through the delay device. This is analogous to changing the motor speed or head separation of a tape echo device, and produces a characteristic "Doppler pitch shift" effect. Delay time modulation is what puts the sweep into flanging and chorusing, and the warble into double tracking and vibrato setups. It can also be



† 1.28 sec FOR MEMORY OPTION 1
2.56 sec FOR MEMORY OPTION 2

***ADJUST LEVEL APPROPRIATE TO INPUT SOURCES AND OUTPUT DEVICES USED.**

FIG. 3.24. Setup for Infinite Repeat demonstration.



***ADJUST LEVEL APPROPRIATE TO INPUT SOURCES AND OUTPUT DEVICES USED.**

FIG. 3.25. Setup for X2/8-kHz demonstration.

used to produce some dramatic pitch coloration at longer delay times.

To begin our experimentation with the Modulation Control section controls, set up the front-panel controls as shown in Fig. 3.26.

XTAL and MANUAL SWEEP Controls — Now, input audio source signals through the 95 and press the pushbutton labeled "XTAL" (changing control of the time base over from the crystal oscillator to the voltage-controlled oscillator). Very little, if any, change will be heard in the output, but we have now enabled all of the controls in the Modulation Control section. Locate the MANUAL SWEEP control and rotate it slowly throughout its range. You should hear a sort of flanging sound as you move this control. Then move the control more quickly and listen to the difference. Faster motion of the control causes an audible bend in pitch to occur. This is because audio material enters the delay at one rate, and leaves it at another rate. If the amount of initial delay time is increased, the effect will become much more pronounced.

Notice that as you turn the control, the delay times indicated on the digital display change, and a small bar of light slides back and forth beneath the digital display. Rotate the Delay Select A control fully clockwise and then vary the MANUAL SWEEP control. The delay indicated on the digital display is equal to the length of the segment that will be captured in memory if the Infinite Repeat pushbutton is engaged. Using the MANUAL SWEEP and the X2 controls, segment lengths can be controlled over a range of 6 to 1. Now capture a segment in memory (by engaging the Infinite Repeat pushbutton) and then change the MANUAL SWEEP setting. The captured material will change its length and pitch just as though it were recorded on tape that was being sped up and slowed down.

DEPTH and RATE Controls — Return to the initial modulation setup (Fig. 3.26) and rotate the DEPTH control fully clockwise. The 95 will now automatically perform the sweep function that we just created manually. This setup is the classic flanging effect; i.e., a simple audio comb filter being periodically swept up and then down. Jump down to the RATE control at the bottom of the Modulation Control section. As you vary this control, you will hear the sweep function speed up until it becomes a rapid vibrato. Notice that as the rate increases, the amount of pitch bend heard increases. To counteract this effect, reduce the setting of the DEPTH control. Generally speaking, in any swept type of setup, the depth of modulation should be reduced as the rate is increased or as longer delay times are selected, provided that the intent is to maintain a similar amount of pitch coloration. If exaggerated pitch coloration is the intent, though, this rule would not apply.

LFO Control — Now press the pushbutton labeled "LFO" (low-frequency oscillator) once and listen to the difference in the sweep. This is a square-wave modulation that changes in large steps up and down periodically as set by the RATE control. Select a longer delay time for the initial delay, and listen to the large shifts in pitch as the delay time changes. Then return to the approximate original delay and press the LFO pushbutton again to return to normal sine-wave sweep.

SHAPE Control — Turn the rotary control labeled "SHAPE" to its full counterclockwise position, and play material with some percussive dynamics. What you hear is a flange type of setup in which the sweep is generated by an envelope-follower circuit on the Model 95's input. This causes the delay time to sweep in response to changes in the level of the signal from the Input Mix Control section. If the SHAPE control is rotated to its middle position, a combination of equal amounts of waveform and envelope-follower modulation will be heard. Experiment with some longer delay times and feedback for other effects. A small amount of mixed modulation at a medium delay produces some very striking chorusing effects. An example of this is shown in Fig. 3.27.

A block diagram of the complete Modulation Control section of the 95 appears in Fig. 3.28.

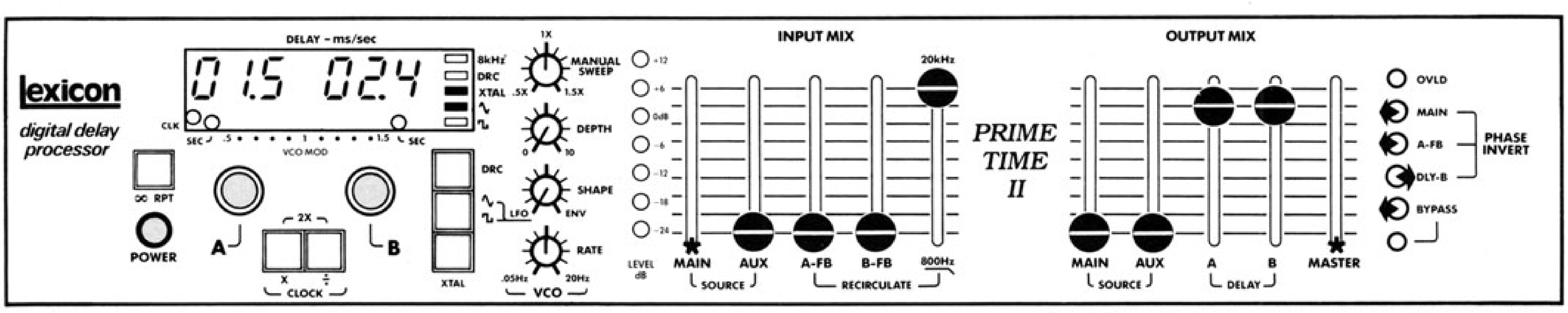


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FIG. 3.26. Setup for delay-time modulation demonstration.

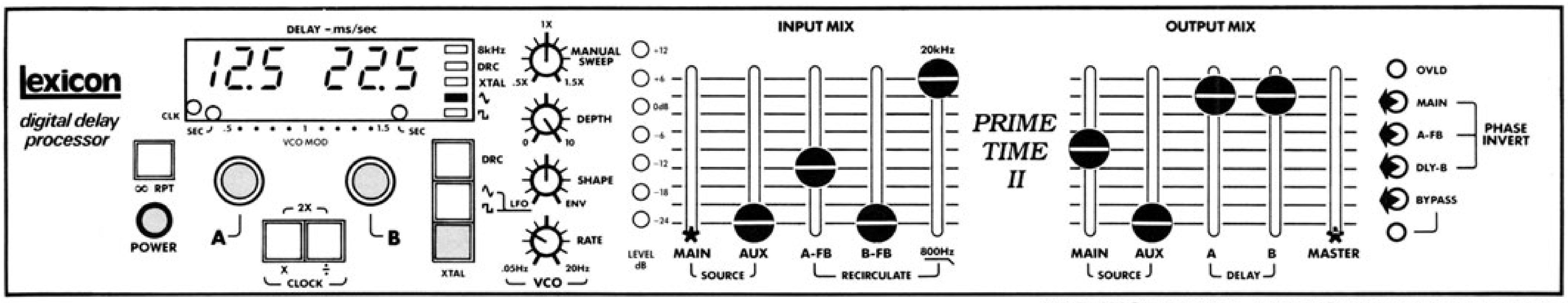

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FIG. 3.27. Mixed-modulation chorusing.

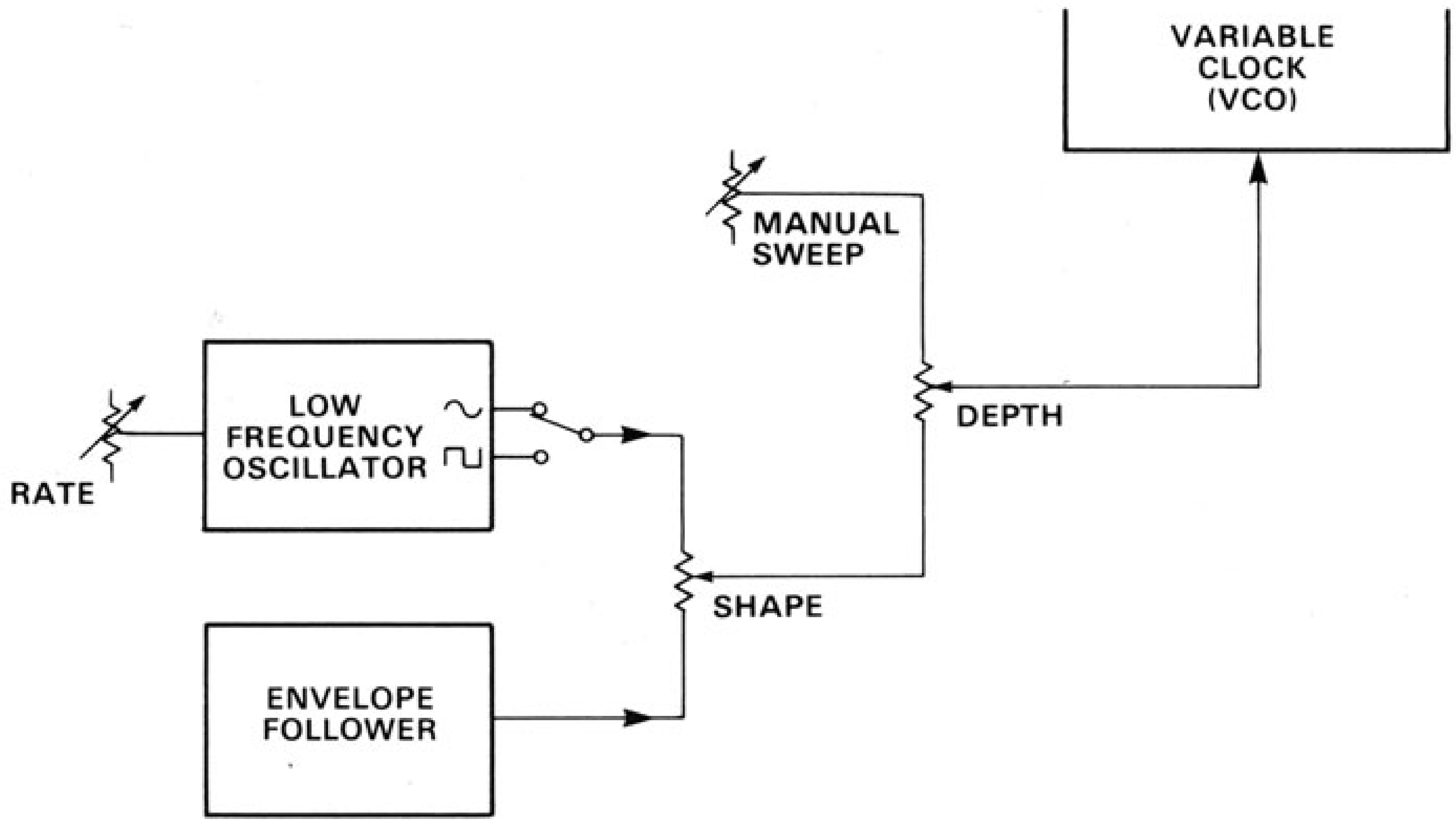


FIG. 3.28. Modulation control section block diagram.

The output of this block enters the delay block shown in previous diagrams. The basic signal diagram, with the delay controls and modulation functions, is shown in Fig. 3.29.

The Modulation Control section can also produce more exotic types of effects by applying various sweeps to longer delay times. The pitch-change artifacts of the modulation become much more exaggerated, and many unusual transformations of the input signal can be produced by combining sweep functions with recirculation, etc. These types of settings are referred to generically as *pitch twisting*. Figure 3.30 shows a starting panel setup for exploring these effects.

At this point, we have introduced all of the basic functions of the Model 95 and have developed the full block diagram of audio flow and effects processing. (The Clock function has been omitted because it is a specialized feature that interacts with the Infinite Repeat function and/or outboard devices for overdubbing, rythmic backing tracks, and other specialized features. The Clock function is explained fully in Section 5 in the discussion of the 2X/CLOCK pushbuttons, and an application of the Clock function is described in Section 4.)

The user should now be well equipped to understand the operation of the various controls and their combinations. Section 4 discusses in more detail the theory of time-base processing with illustrations of many practical settings and suggestions for further exploration.

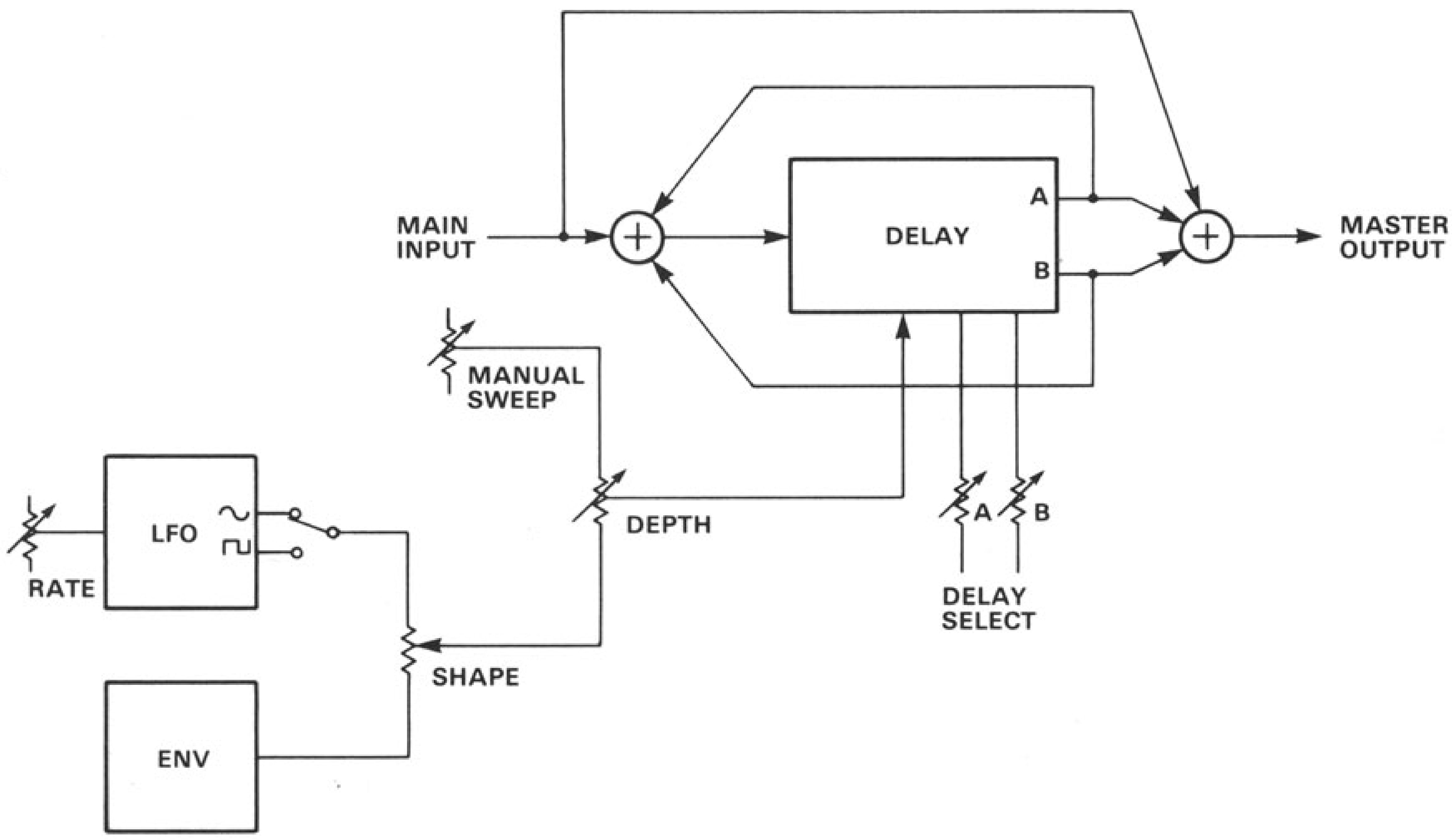
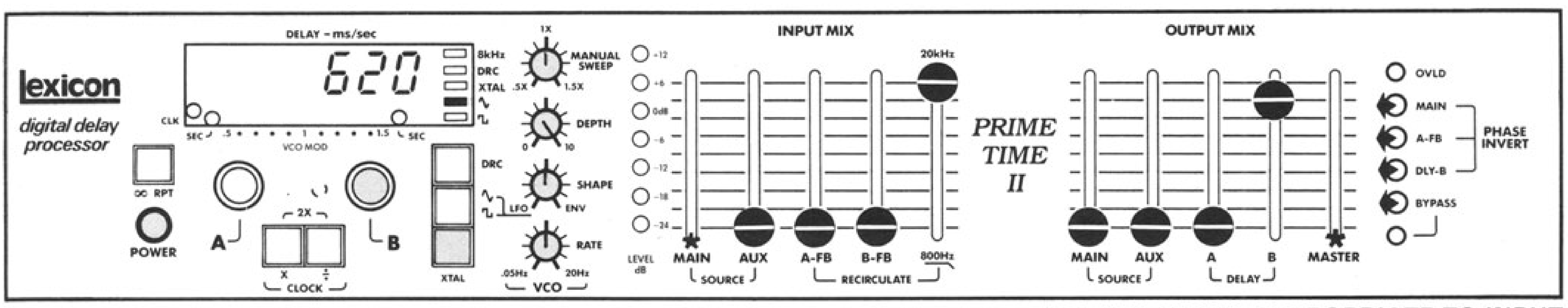


FIG. 3.29. Audio block diagram with delay control/modulation.



*ADJUST LEVEL APPROPRIATE TO INPUT SOURCES AND OUTPUT DEVICES USED.

FIG. 3.30. Panel setup for pitch-twisting experiments.

4
Applications

4.0 Introduction

This section of the manual is aimed at users who already have a general knowledge of the operation of the controls and features of the Model 95, either from previous experience or from studying Sections 3 or 5. In this section we discuss several areas of application in terms of their general parameters, with specific illustrations in the form of panel setup sheets. We also discuss the use of the Model 95's unique features in some detail.

4.1 Theoretical Background

The capability of audio time delay functions to produce perceptually interesting processing stems from both the physics and psychology of sound and music perception (psychoacoustics). All of our lives we experience time delay effects, which we generally integrate into our perception without conscious thought. Our hearing mechanism (ear and brain together), for instance, can locate the source of a sound largely from minute differences in the arrival time of the sound at each ear. Automobiles speeding past with their horns blowing exhibit "Doppler pitch shifting," a time-modulation phenomenon resulting from the continuously changing distance between the source and the listener.

Reflected sound produces many familiar acoustic effects. Jet airplanes passing overhead have a "flanging" sound because of the difference in path lengths between the direct sound to our ears and the strong reflections from the ground around our feet. Everyone is familiar with the long echoes found near mountain cliffs, but we are less likely to consciously notice the "comb filter" effects of ordinary rectangular rooms, or the shorter echoes that surround us on city streets and elsewhere. In fact, virtually every object, surface, or space we encounter in our daily lives reflects sound in some degree, with the delay time we hear dependent on the difference in path lengths between the direct and reflected sound (sound travels at the rate of approximately 1 millisecond per foot). Most people do not give thought to these delays constantly impinging on their hearing, but anyone who has had the experience of entering an "anechoic" chamber (a room specially constructed to have *no* reflected sound at all) can testify to the unnerving effect of having these phenomena suddenly cease.

In music production, we can use these various types of effects to enhance and alter sound in a creative, controlled fashion. Localization (time-delay panning), flanging effects, artificial double tracking, chorusing, and simulated room ambiences can be easily obtained using short delay times. Because we are using electronics to do these things, we can go beyond nature in some cases. For example, we can add recirculation to flanging and doubling setups, simulating a room with 99.9% reflective walls and no air absorption factor for bizarre, robotic "tuned resonance" effects, or we can rapidly modulate time delay as though a sound source were hopping about at an impossible speed.

At longer delay times, we hear the delayed signal as a distinct repetition of the original sound. That this phenomenon can have a highly pleasing effect has been testified to by over 30 years of common usage, ever since the advent of tape recording made artificial echoes practical. Artists from Les Paul to Pink Floyd have made effective use of discrete echoes in some of the most popular recordings of all time, and there is no sign that the usage of long delay in popular music is likely to decrease.

The reasons that this phenomenon has been found so desirable can be discovered from an examination of the nature of music itself. The primary elements of music are rhythm, melody, form, and timbre (created by harmony and orchestration). The repetitive nature of rhythm should be apparent, and the role of repetition and variation in melody is well known. Form can be said to be the organization of variation and repetition on a larger time scale, and timbre variations are normally used in the service of form and rhythm to provide nuance, shading, and color for enhancement of the overall sensory experience. The very art of music, then, might be said to be the interplay of intervals of repetition with variation. Clearly, our musical "sense" enjoys both being led to expect certain events to recur, as well as being "fooled" when something different occurs.

Interesting speculations about the nature of the human mechanism of perceiving repetitive audio material might be derived from a discovery made by researchers in the area of speech impediments. They discovered that certain kinds of speech problems can be ameliorated if the patient monitors his own speech through an audio delay of about 1/4 second—something that *induces* a speech impediment in most normal people (try it sometime). Perhaps most of us have a natural delay function in our hearing mechanism through which we unconsciously review our own speech, and these speech-impaired individuals might lack this mechanism.

The electronic medium again allows us to exceed the normal ranges of acoustic phenomena and produce repeats at intervals of several seconds, with multiple repetitions and little change in frequency content (natural echoes are always affected by the air they travel through and the surfaces they reflect from). Contemporary composers such as Steve Reich, Robert Fripp, and Brian Eno have exploited these characteristics to produce "loop music" in which the repetitive portion of the music is derived directly from the repetition of material played into a delay system with very long delay times and recirculation. The Model 95 has special facilities for exploring this realm (see Section 4.9).

4.2 Localization Effects

The *Haas effect* is the name given to the phenomenon of psychoacoustic localization of sound based on the relative delay of the incidence of a sound between each ear. Dr. Haas's research has shown that a great deal of the sonic cues that the brain uses to locate a sound laterally (that is, from left to right) is derived from the unconscious perception of the interval between the sound reaching the nearer ear and the farther ear. Since the maximum distance between the ears of the path to the ears is under 1 foot, the delays involved are less than 1 millisecond, showing our ear-brain mechanism to be very sensitive indeed. We can use this effect to our own advantage to create the illusion of location in sound reproduction (see Fig.4.1).

Ordinarily, recording engineers use the "pan pots" on recording consoles to set the location of each sound in the stereo field, but a similar effect can be obtained by delaying the sound fed to the speaker opposite the side the engineer wishes to localize the sound to. If headphones are used for monitoring, the effect is dramatic with even tiny amounts of delay. On speakers, the effect is confused by the combined speaker outputs reaching each ear independently, but becomes very apparent with a few milliseconds of delay.

Localization of sound in this fashion offers the advantage of having the energy in each channel remaining the same, yielding higher apparent volume than if the sound is panned in the conventional manner. Some or-

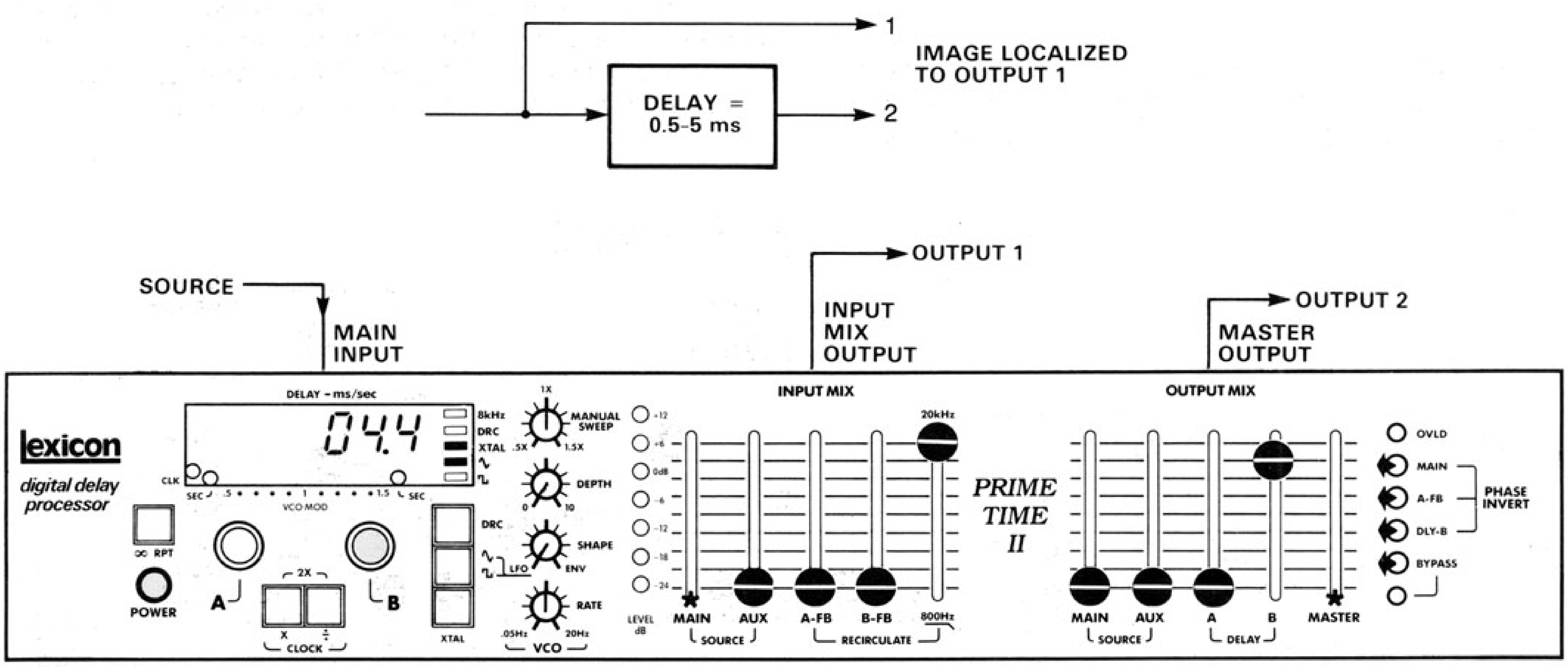


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FIG. 4.1. Hass-effect panning.

dinary amplitude panning can enhance the effect, working together as they do in nature. The disadvantage of this technique is that monaural compatibility may be compromised (the original and delayed signal form a "comb filter" when they are combined). If this technique is used, it is a good idea to listen to the "monoed" mix for undesirable changes in tone. Usually a time-delay value can be selected to produce a satisfying localization without excessive loss of color in mono.

A few milliseconds of delay between stereo channels also tends to "spread" the sound across the stereo field, creating a sense of an image larger than the speaker itself. This property can be used very effectively to create mixes that have a feeling of dimension in which the various instruments each occupy their own distinct spaces (see Fig. 4.2).

4.3 Comb Filters, Resonance, and Flanging

When a time-delayed signal is combined with its original source, a phenomenon occurs in which those components of the sound whose period has a direct relationship to the delay time are either boosted or cut in relation to the original spectrum. This effect is called *comb filtering* because of the comblike appearance of the frequency response curve caused by the alternating emphasis and canceling of relational frequencies across the frequency spectrum (refer to Fig. 4.4). This effect is most audible for delay times from a fractional millisecond to about 20 milliseconds at equal amplitudes. The nature of this frequency cancellation can be understood from an examination of Fig. 4.3.

If we start with a delay line whose delay we call T, input a sine-wave signal whose frequency equals $1/T$ (meaning the period it takes to complete one cycle equals T), and mix the output with the original signal, the total signal is twice the amplitude of either signal alone. However, if the frequency equals $1/2T$ (the period to complete one cycle equals twice the delay time), the resulting signal is 0, because the two signals exactly cancel each other.

If a complex signal is input, the frequencies at which signals are emphasized or canceled (nulled) are dependent upon the mathematical relationship of the frequencies to the delay time. The reciprocal value of the delay time determines the fundamental frequency at which a peak occurs. Peaks also occur at harmonics of that frequency, and nulls occur at 1/2 the fundamental frequency and at uneven multiples of 1/2 the fundamental frequency. As an example, a 1-millisecond delay produces a peak at the fundamental frequency of 1000 Hz ($1 \div 0.001 = 1000$) with peaks at its harmonics of 2000 Hz (1000 x 2), 3000 Hz (1000 x 3), etc. Nulls occur at 500 Hz, (1000 x 1/2), and at uneven multiples of 1/2, or every 1000 Hz; such as: 1500 Hz (1000 x 3/2), 2500 Hz (1000 x 5/2), 3500 Hz (1000 x 7/2), etc. If the delayed signal is phase inverted, the reverse occurs; the fundamental peak is at 500 Hz, with nulls at 1000 Hz and every 1000 Hz thereafter.

Without phase inversion, the delayed signal simply adds to the undelayed signal at low frequencies. This reinforcement of low-frequency components occurs because the short delay of 1 millisecond only represents a slight phase shift; however, the phase shift increases with frequency, until at 500 Hz it is 180 degrees and the first null is created. With phase inversion, the low-frequency components of the delayed and undelayed signals are out of phase, canceling much of the low-frequency content. This inverted comb filter acts as a low-frequency filter, which can be "tuned" to filter progressively higher frequencies as the delay time is shortened. (See Figs. 4.4 and 4.5 for examples of noninverted and inverted comb filters, respectively.)

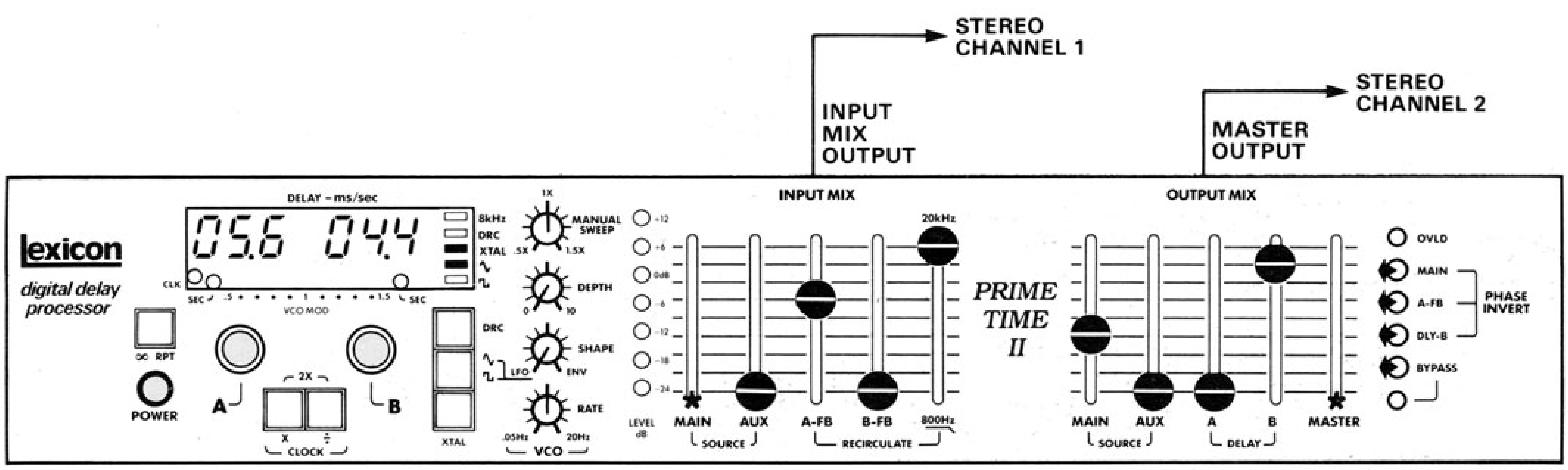

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FIG. 4.2. Stereo image enhancement.

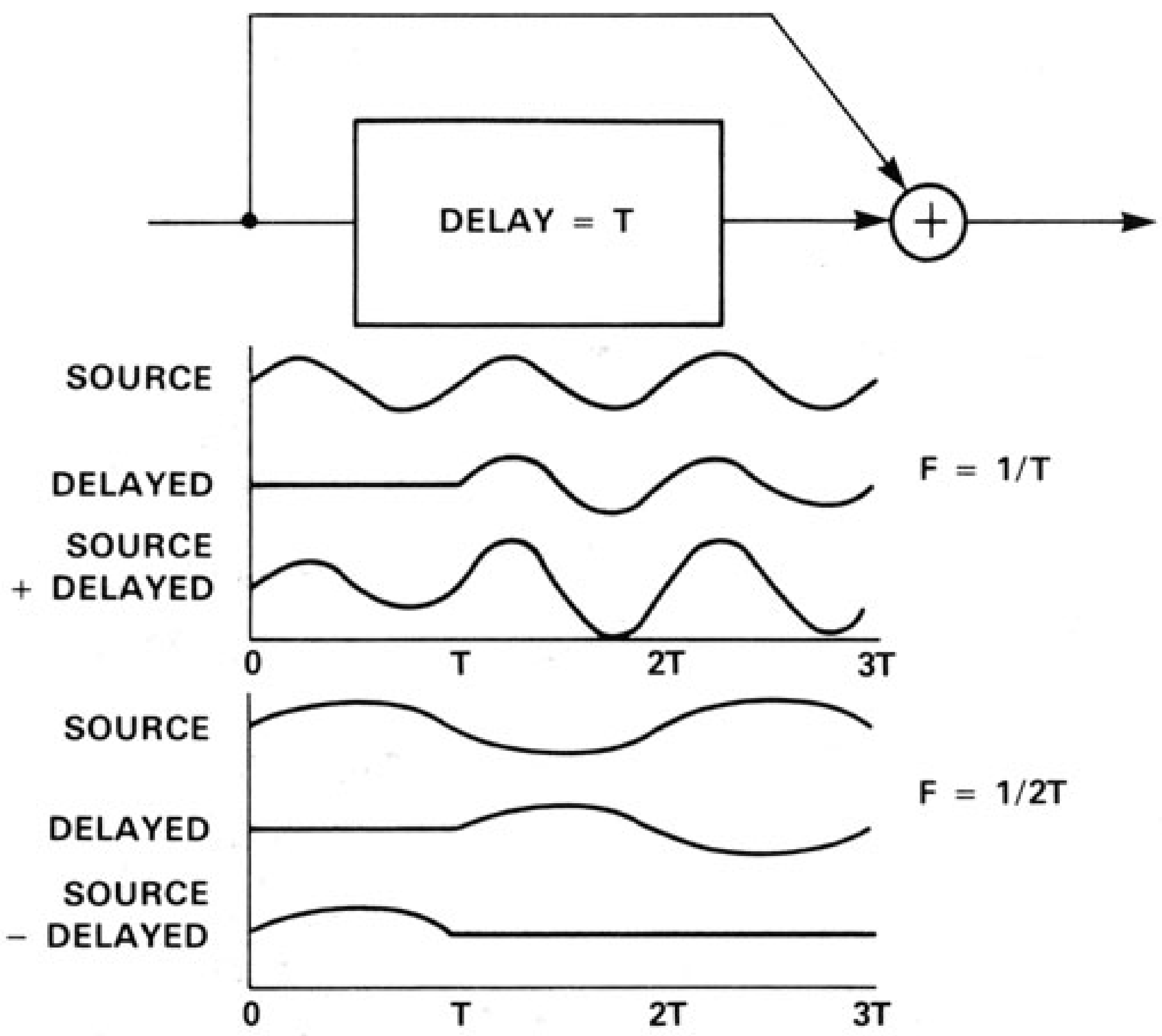

FIG. 4.3. Physics of comb filtering.

4.3.1 Hum Filtering

From an understanding of comb filters, it is easy to see that a single frequency and all of its harmonics can be eliminated in the nulls of a comb filter. This is particularly useful for eliminating hum from a source signal, because hum can consist of the line frequency plus all of its harmonics, and therefore cannot be entirely eliminated through the use of a high-pass hum filter because the harmonic products would not be affected.

As an example, to eliminate hum from a 60-Hz line, an inverted delay of 16.7 milliseconds (1/60 = 0.0167) can be added to the source signal, both signals at equal amplitudes. This creates nulls at 60 Hz and all of its harmonics. However, since the comb filter produces changes in timbre and therefore can result in coloration of the original source signal, experimentation with different delays can provide the delay that is both effective in eliminating hum and suitable for the characteristics of the original signal. For example, an inverted delay of

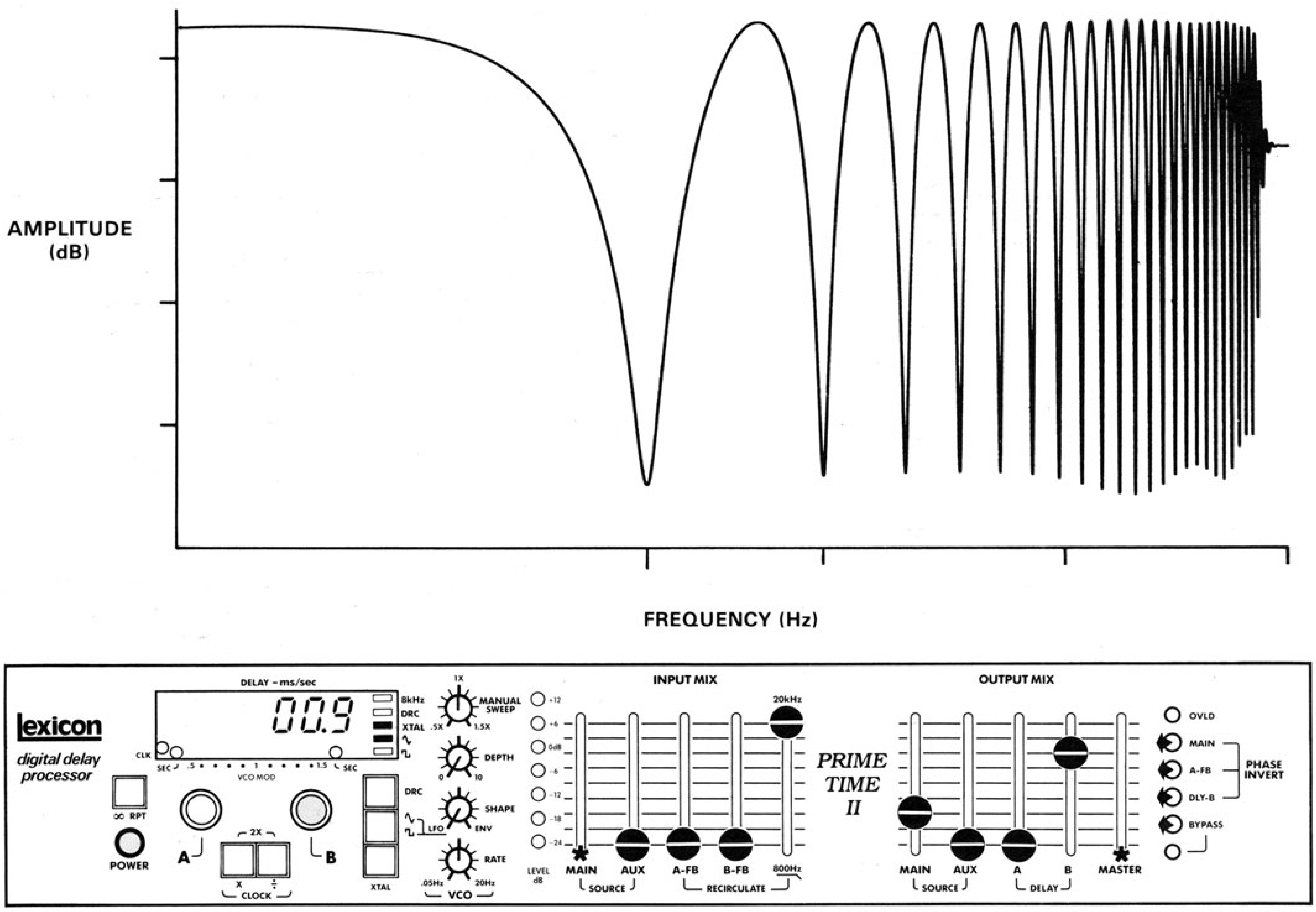

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FIG. 4.4. Simple comb filter.

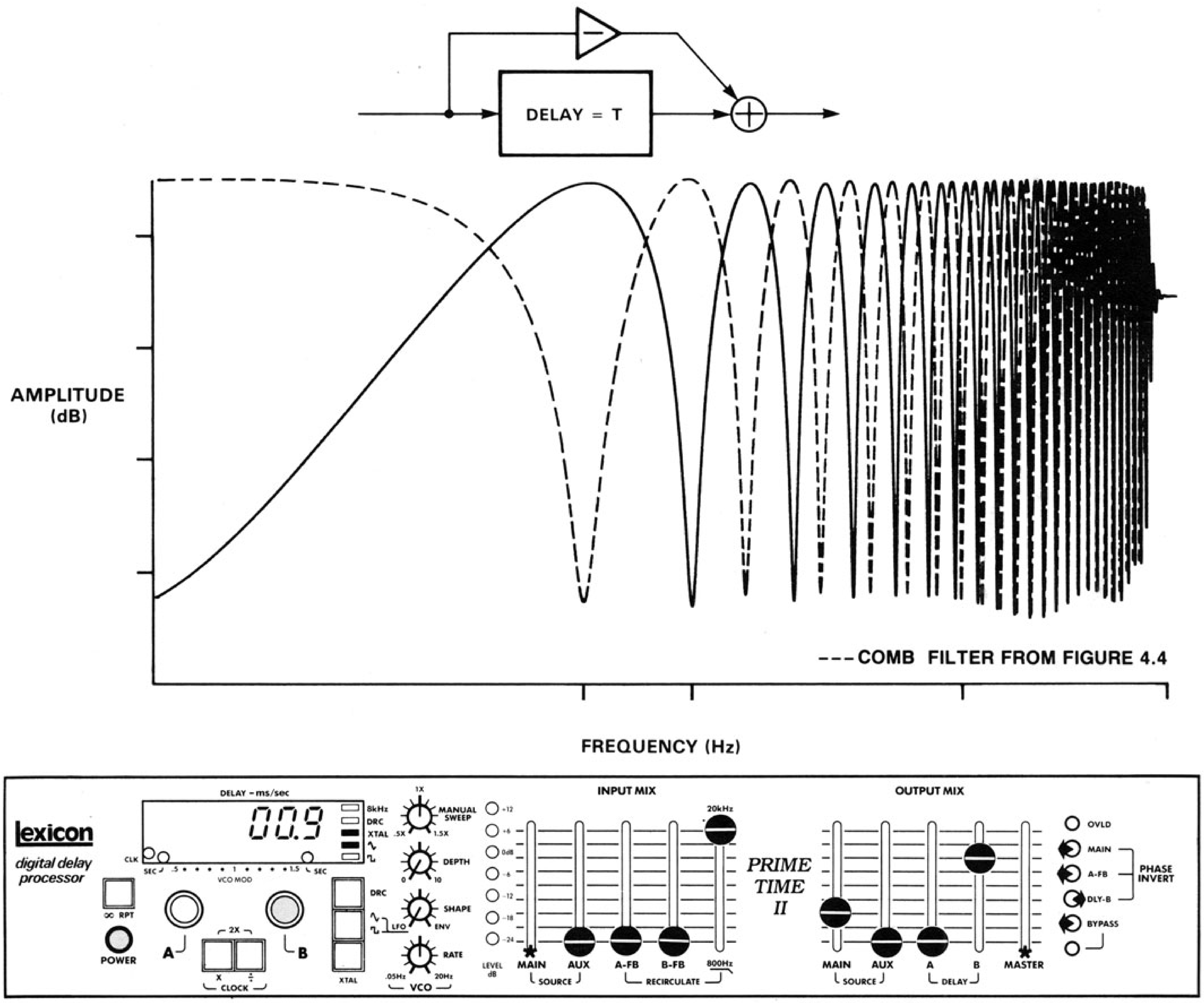

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FIG. 4.5. Comb filter with phase inversion.

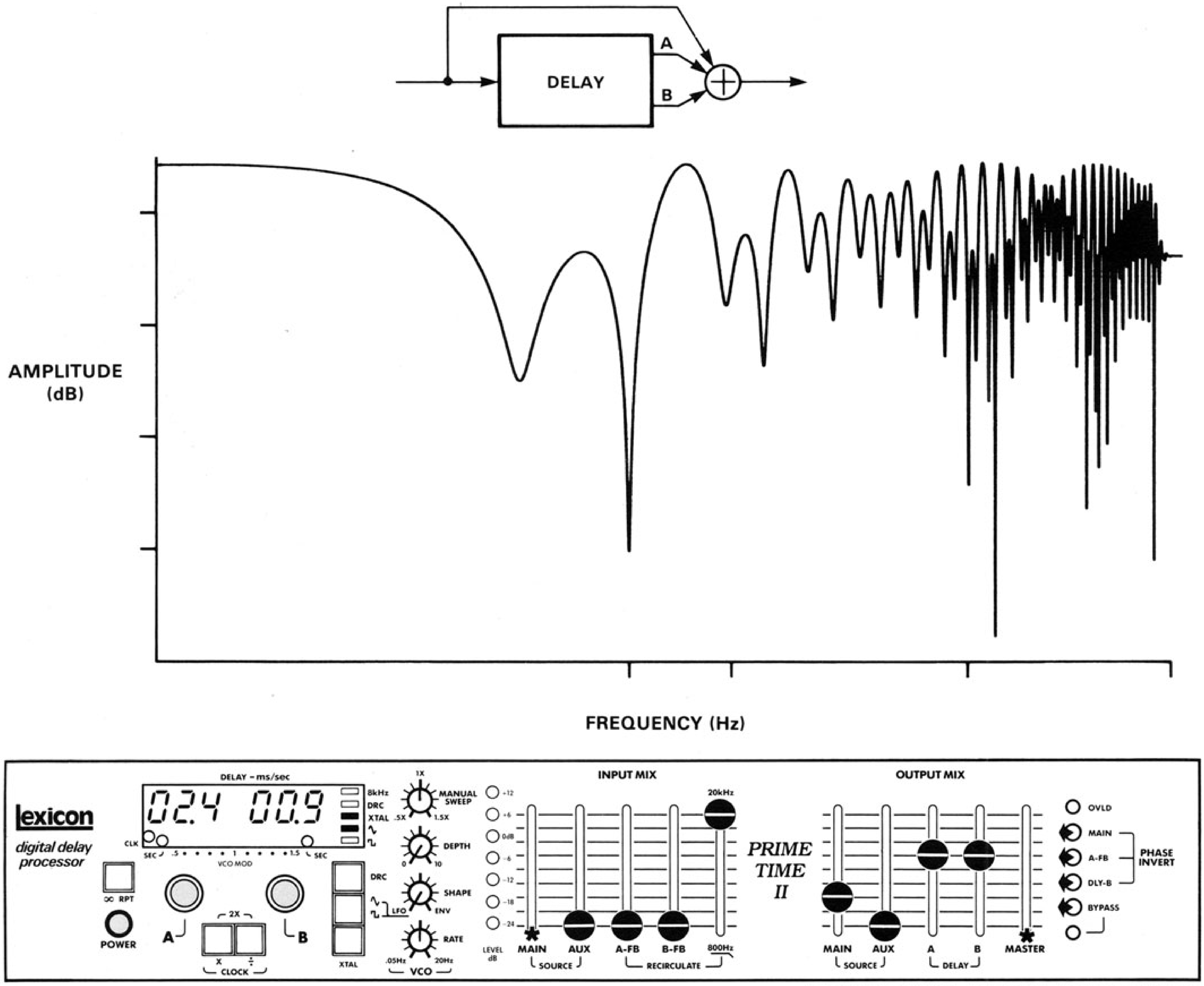
33.3 milliseconds (1/30 = 0.0333) eliminates 30 Hz and all its harmonics (including the offending second harmonic of 60 Hz and its related harmonics) and may ideally match the tonal characteristics of the source signals.

4.3.2 Resonance and Flanging Effects Using Comb Filters

With the possible variations of signal routing and recirculation available using the 95, comb filtering setups can become highly complex and require a great deal of experimentation to discover the vast number of effects possible. As a starting point, Figs. 4.6, 4.7, and 4.8 show some typical comb filter setups incorporating dual delays, recirculation, and resonance effects, respectively. Figures 4.9 to 4.14 show the use of comb filtering setups to produce various flanging effects.

A comb filter setup in which recirculation is turned up to almost (but not quite) unity gain, has very strong resonant qualities at the characteristic frequencies. This type of effect can be used to produce vocoderlike "robot" voices, to emulate resonances of acoustic instruments, or to produce actual tones from percussive inputs.

In real life, we deal with comb filters all the time as sounds reflect from surfaces in our environment and combine with the direct source signals. Most often these effects are a little more subtle than the setups we have



*ADJUST LEVEL APPROPRIATE TO INPUT SOURCES AND OUTPUT DEVICES USED.

FIG. 4.6. Comb filter with dual taps.

shown because of the disproportionate levels of the direct and delayed sound, but some situations can produce very strong acoustic comb-filter effects. Very hard and flat walls running parallel to each other, such as a tiled hallway, produce effects similar to the "resonant comb" shown in Fig. 4.8.

The "tuning," and thus the tonal quality of a comb filter is primarily dependent on the delay intervals involved, as well as the adjustments of levels and relative polarity of signals. So far, we have discussed the case in which the delay times are left fixed while signal is played through the unit. If the delay times are varied in a smooth fashion, as can be done with the Delay Modulation section, the nulls of the comb filter move through the incoming spectrum and produce the effect commonly known as *flanging*. Doppler pitch shifting effects (discussed in Section 4.4) are also introduced by the variation in delay time and contribute to the sense of motion that flanging can impart. Any variation of a comb filter can be "swept" (a setting that is continuously changed over a prescribed range) in this fashion, and a large number of different colorations of the basic effect can be obtained, particularly with a dual-tap delay line such as the Model 95. A few of the possible variations are shown in Figs. 4.9 through 4.13.

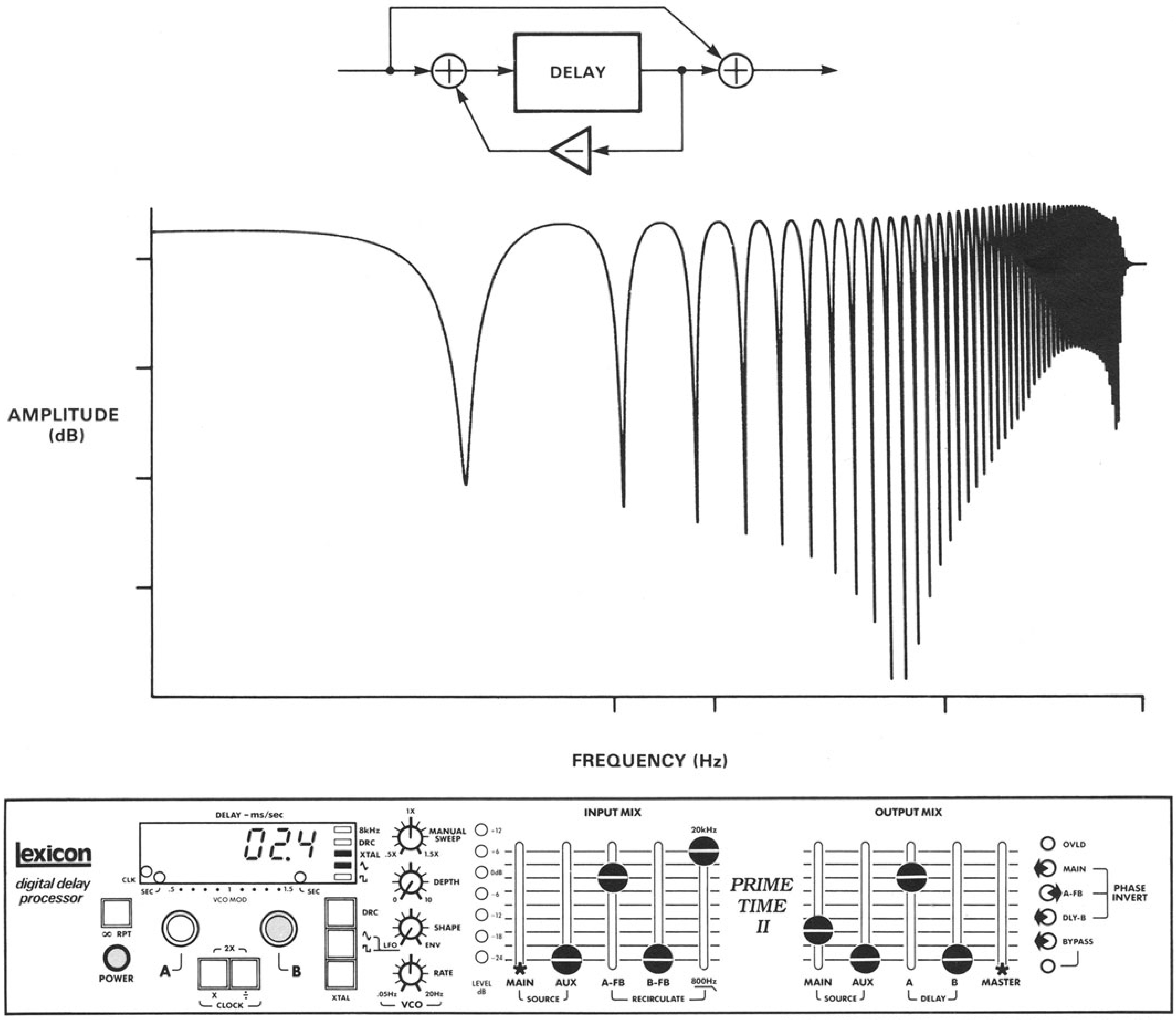


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FIG. 4.7. Comb filter with recirculation.

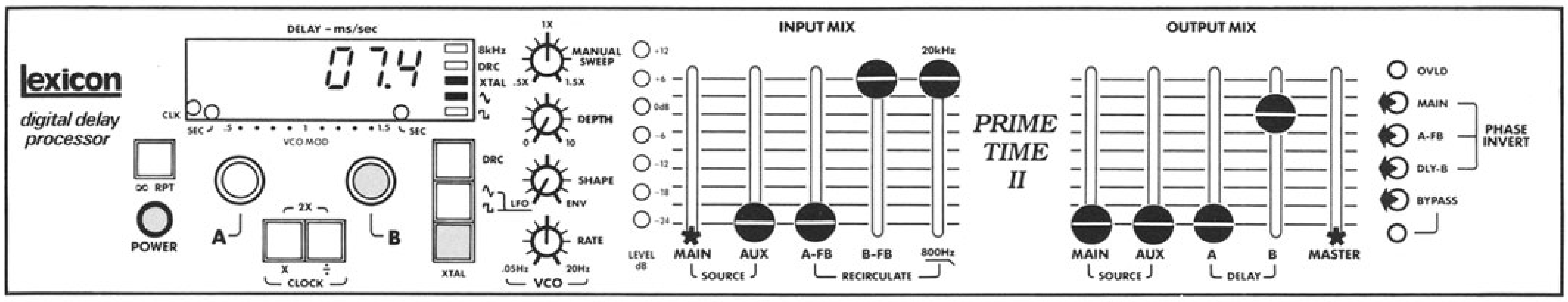

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FIG. 4.8. Resonant comb filter.

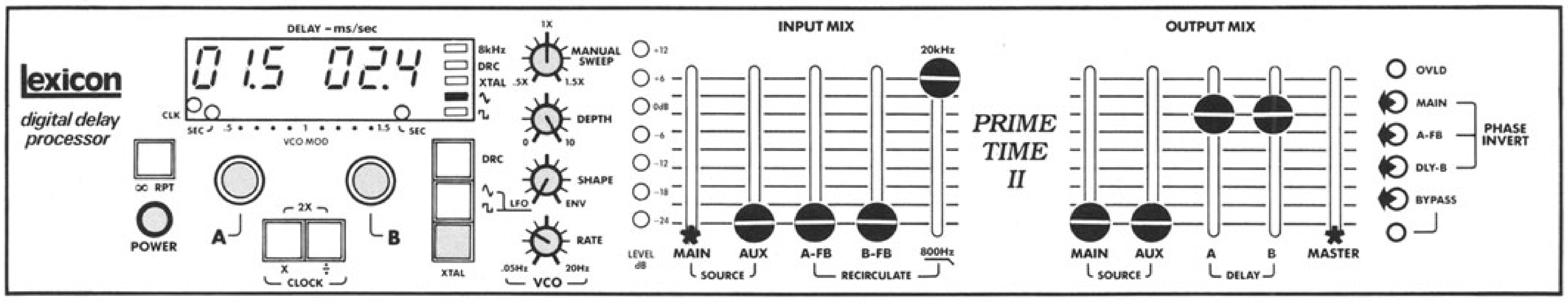

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FIG. 4.9. Basic flange.

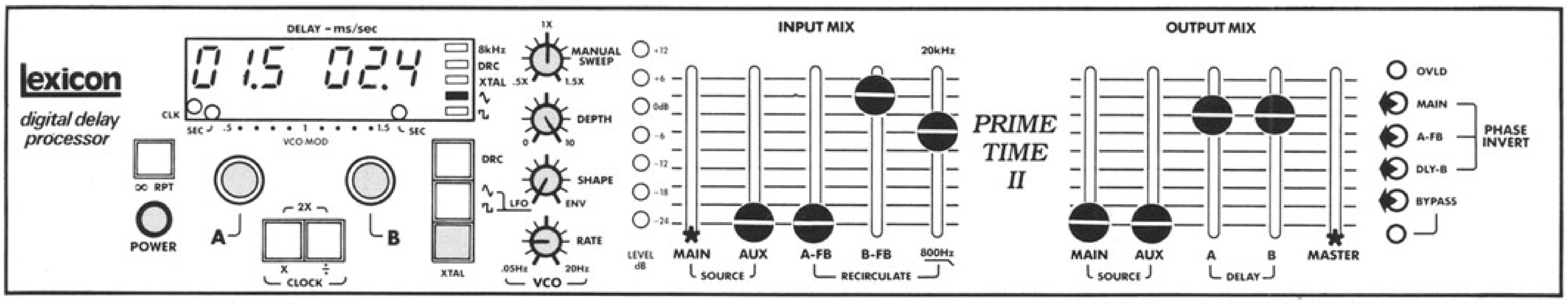

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FIG. 4.10. Resonant flange.

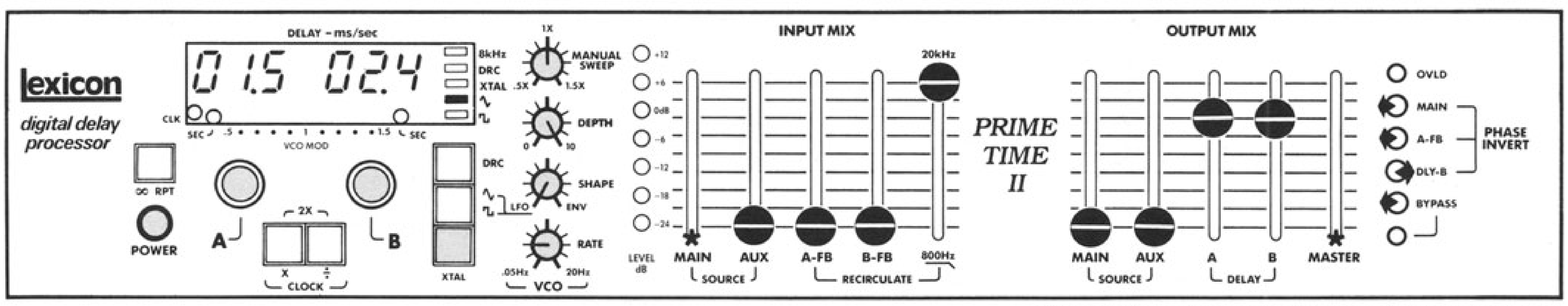

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FIG. 4.11. Negative flange.

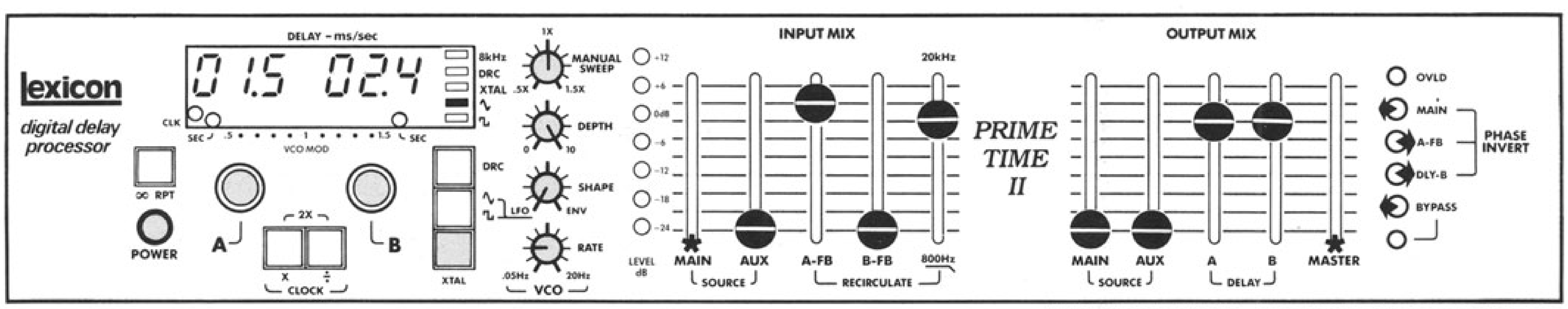

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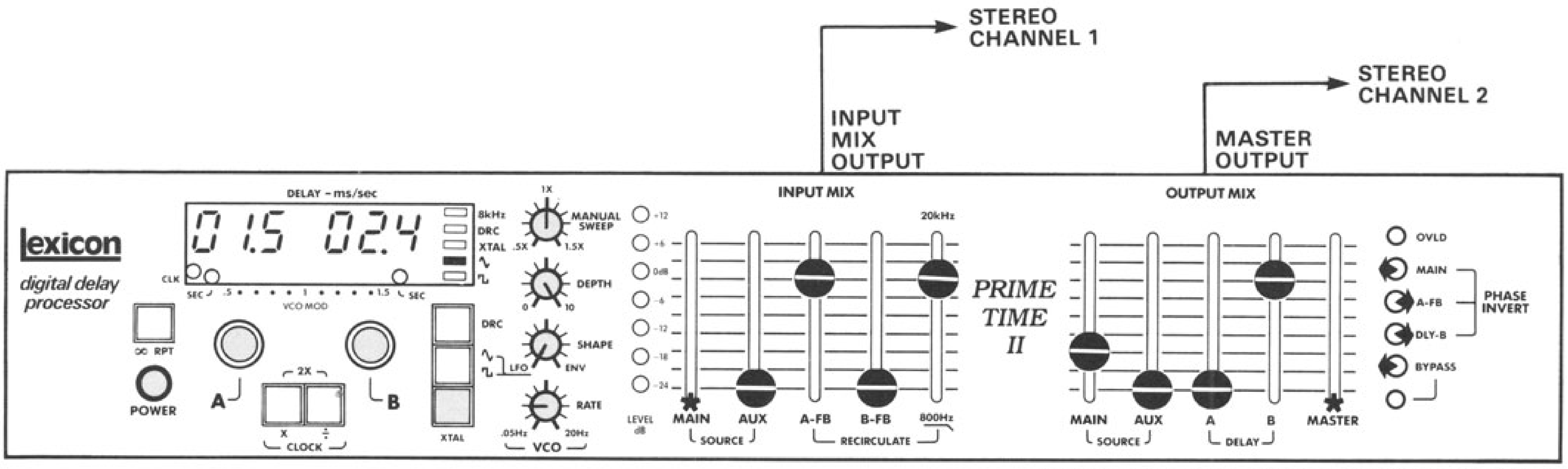
FIG. 4.12. Negative resonant flange.



*ADJUST LEVEL APPROPRIATE TO INPUT SOURCES AND OUTPUT DEVICES USED.

FIG. 4.13. Stereo flange.

The setup shown in Fig. 4.13 uses the dual-delay tap outputs to create separately tuned flanges on the left and the right sides. The time difference between the two sides also creates a sense of "spread" in the sound, which can be very effective.

In all of the previous flange setups, we used the same source of modulation, a very slow sine wave. We can vary the sound of any flanging setup by changing the rate and depth of sweep, and also by using other sources of modulation. The Model 95 has an envelope-follower circuit (which produces a voltage proportional to the audio signal level) available as an alternate sweep function. This signal can be blended with the signal from the LFO—or low-frequency oscillator circuit (which generates the waveform sweep functions)—or used alone. The term *talking flange* describes the effect generated when a resonant flange is swept by the envelope-follower function (see Fig. 4.14).

4.4 Doppler Pitch Shift Effects

In previous sections we introduced the idea of time "sweep" or modulation, and we mentioned a change of pitch resulting from that modulation. For flanging setups this shift is generally held to a subliminal level where it is heard as part of the general sense of motion that flanging imparts. However, we have the capability to create far more exaggerated effects, with a variety of modulation shapes. The amount of pitch shift produced is a function of how rapidly the delay times change. Since the sweep function on the Model 95 operates as a proportion of delay, this rate of change, or differential, becomes much larger when longer delay times are selected. To hear this, set up the front panel as shown in Fig. 4.15, and then slowly rotate Delay-A clockwise and listen to the effect at several positions. You will hear the pitch shift become more and more exaggerated as the delay times increase, becoming fairly outrageous at the longer settings. Then set the controls as in (b) and slowly decrease the DEPTH control (by rotating it counterclockwise), noticing that the amount of pitch shift decreases. Finally, set the DEPTH control as in (c), and vary the RATE control. You will notice that as the rate of modulation increases, the amount of audible pitch variation also increases.

From this demonstration, we can see that the amount of pitch variation produced is dependent on the delay tap selected, the depth of modulation, and the rate of modulation. By controlling these parameters, we can produce pitch effects ranging from subtle vibrato to extreme swings in pitch (over one octave). In addition, we can

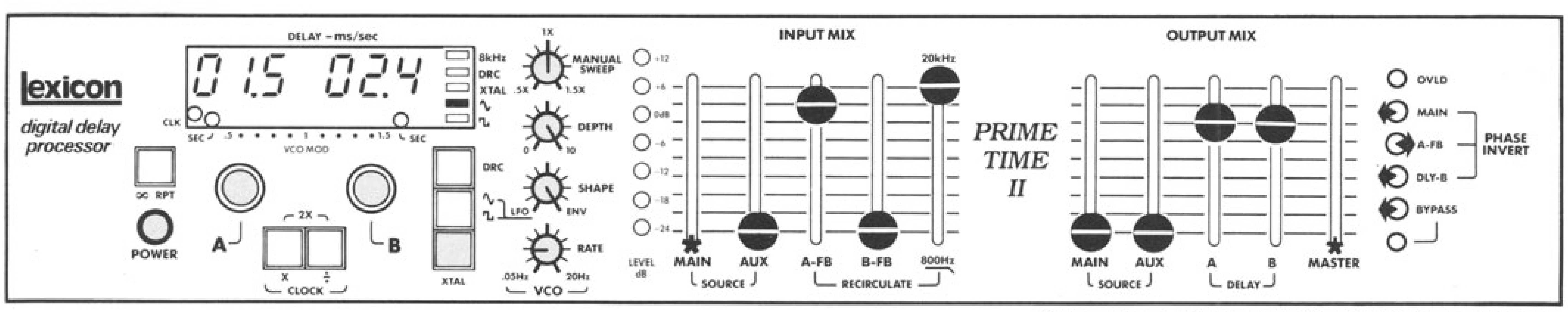


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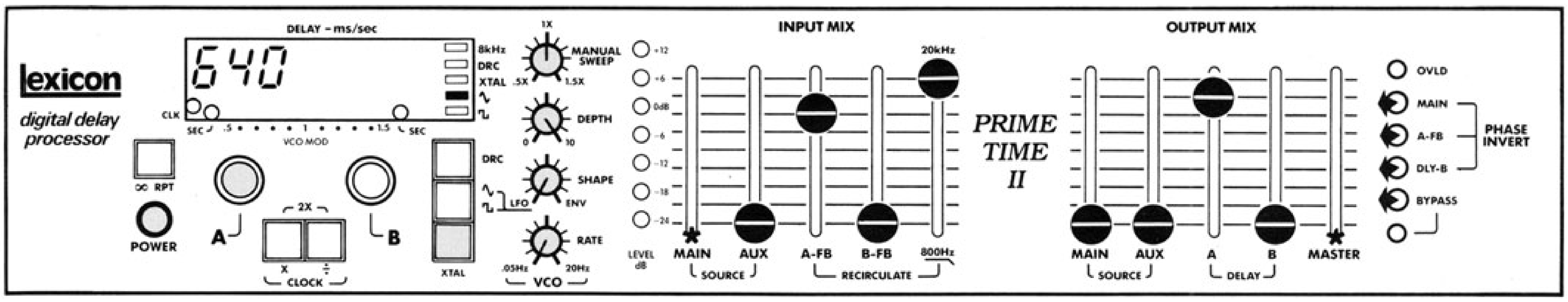
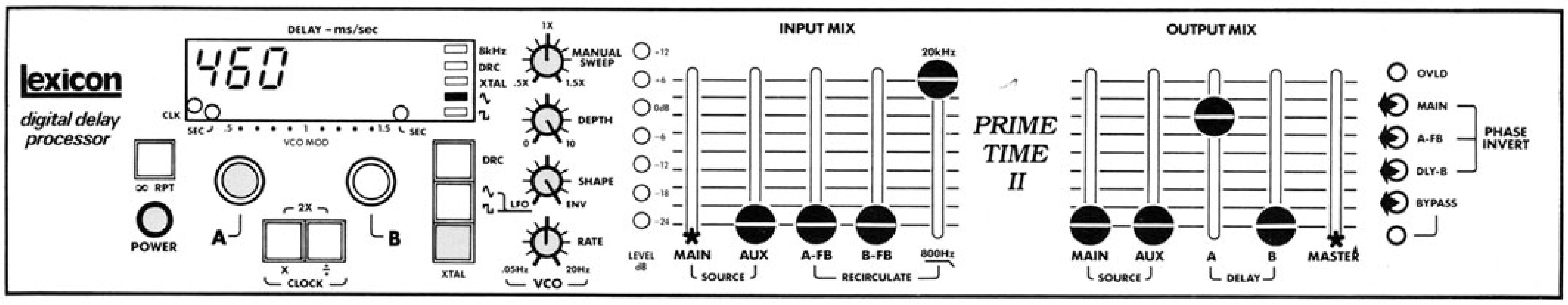
FIG. 4.14. Talking flange.

(a)



(b)



(c)

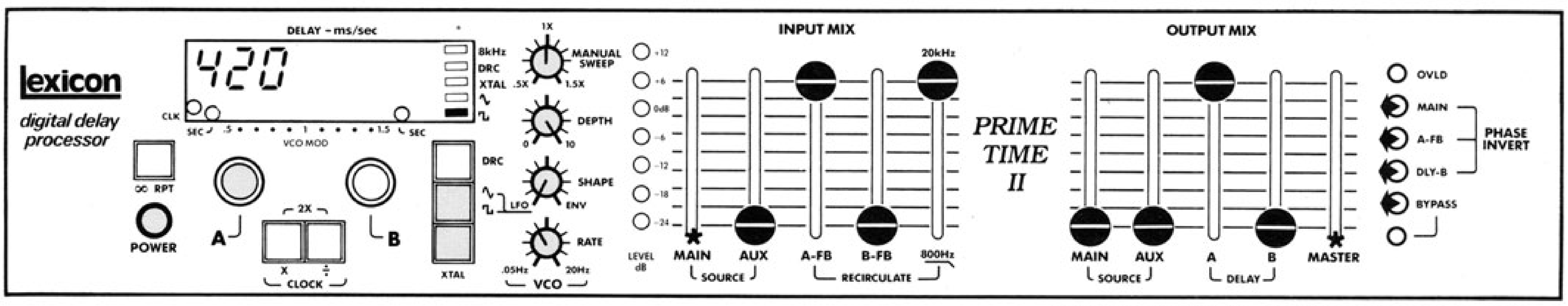


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FIG. 4.16. Pitch-twisting effects.

4.6 Simulated Room Ambience

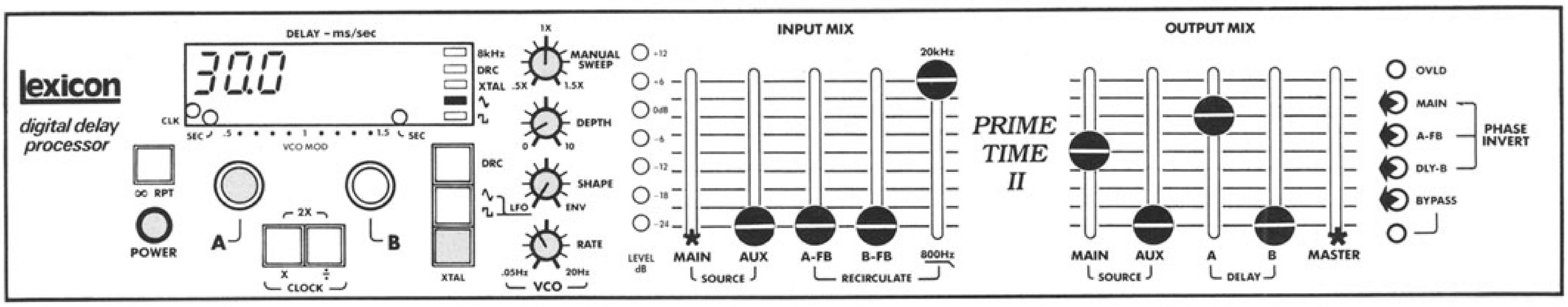
Time delay can be used to create convincing simulations of acoustical environments and room size, particularly environments that have some parallel and relatively hard surfaces. The setup in Fig. 4.19 can be used as a basic small-room setup, and varied to create a range of sizes and characteristics of the space being simulated.

The delay settings can be varied to create larger or smaller spaces. The longer the delay times, of course, the larger the space. The relative lengths of Delay-A and B correspond to the proportions of a rectangular shape. As these are varied, however, care should be taken to select delay values that are *not* related by some simple integer or fractional ratio, as this tends to create an unnatural flutter (a comb-filter effect that cancels certain related frequencies and boosts others) in the "echo return" to the mixing console (i.e., in the output of the 95).

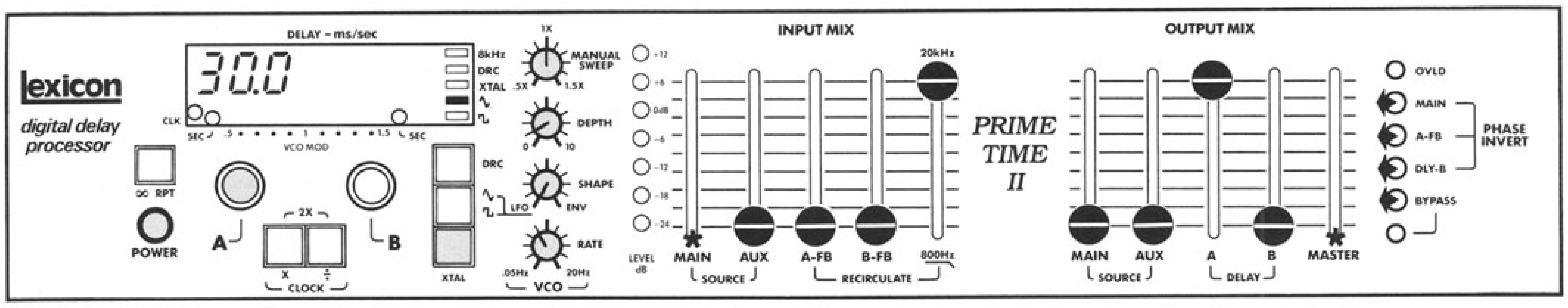
Variation in the amount of recirculation corresponds to the hardness of the (imaginary) reflecting surfaces, and use of the rolloff ("800 Hz–20 kHz") slider can emulate the effects of air absorption on the high end of the sonic spectrum.

The Model 95 can also be used to suggest reverberation in larger spaces, but it should be understood that there is a great deal of difference between single or dual tap time delay and the very high echo density of true reverberation. Nevertheless, delay lines can sometimes produce convincing illusions, dependent on the input material and the degree of masking introduced by the other elements in the mix.

(a)

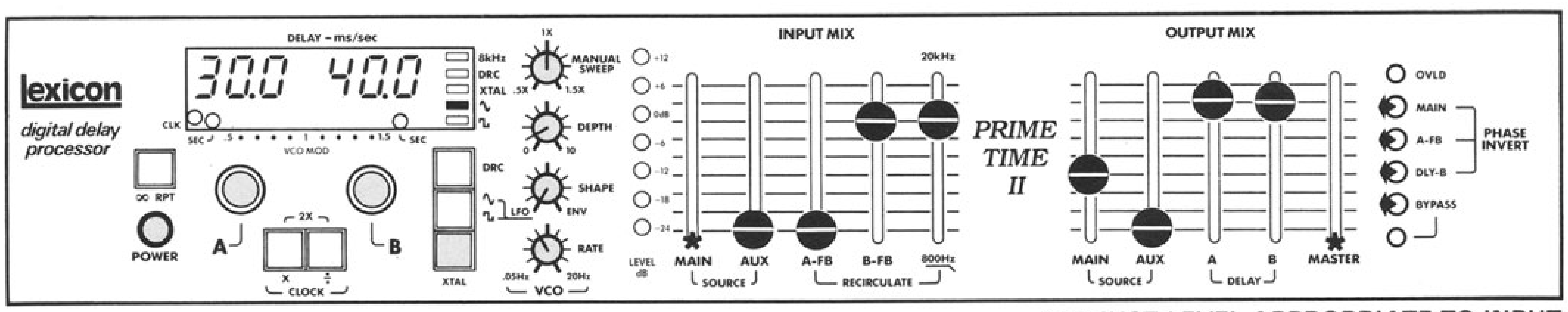


(b)



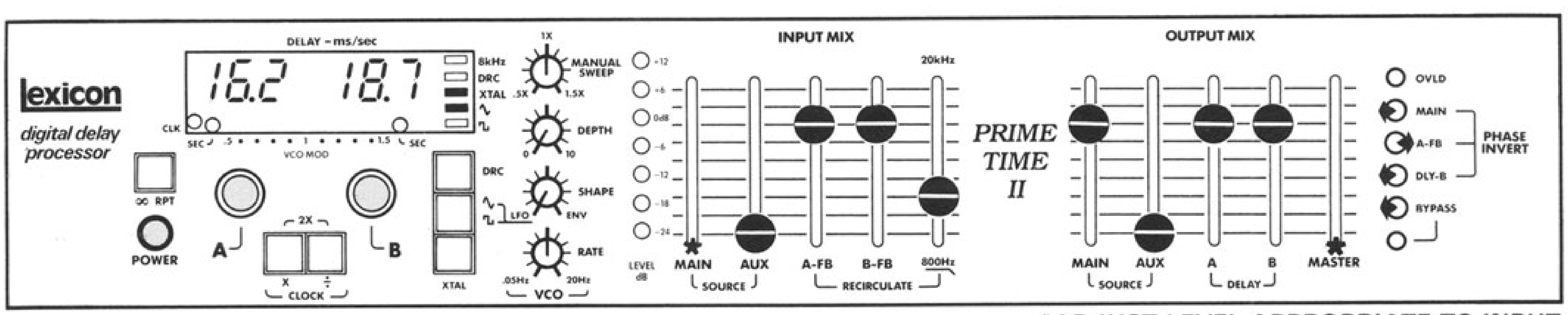
*ADJUST LEVEL APPROPRIATE TO INPUT SOURCES AND OUTPUT DEVICES USED.

FIG. 4.17. Mono and stereo doubling.



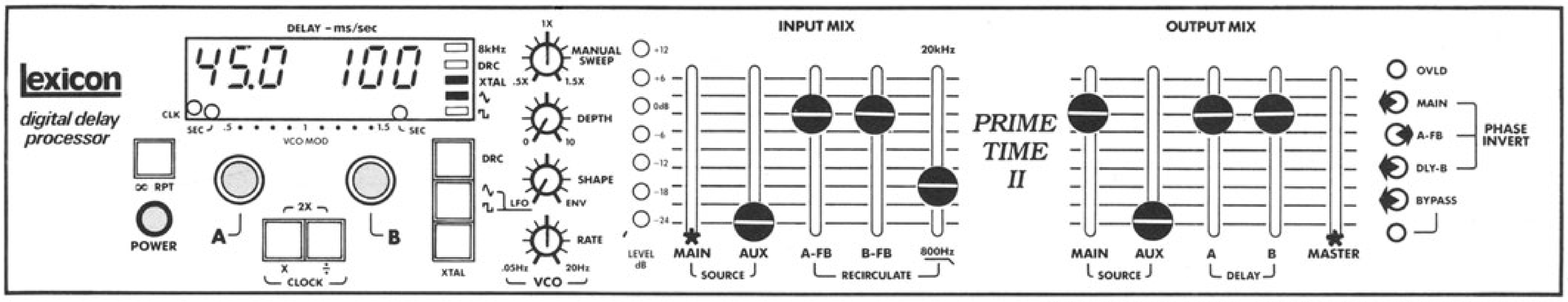
*ADJUST LEVEL APPROPRIATE TO INPUT SOURCES AND OUTPUT DEVICES USED.

FIG. 4.18. Chorusing.



*ADJUST LEVEL APPROPRIATE TO INPUT SOURCES AND OUTPUT DEVICES USED.

FIG. 4.19. Small room simulation.



*ADJUST LEVEL APPROPRIATE TO INPUT SOURCES AND OUTPUT DEVICES USED.

FIG. 4.20. Large room simulation.

4.7 "Slap" Echoes

A very popular effect for musical instruments is the use of a single repeat at around 1/10 of a second delay time. An echo at this length can be heard as a distinct repetition of the original, but so close as to be heard more as a rhythmic type of enhancement rather than as a separate sound. A repeat of this type is commonly called a *slap echo*. Often the delay time is carefully adjusted so that a simple relationship exists between the tempo of the music and the delay time. Very often the slap is positioned on the opposite side of the stereo field from the original sound, commonly on drums. Careful use of slap echoes can increase the rhythmic density and spatial interest of a musical arrangement.

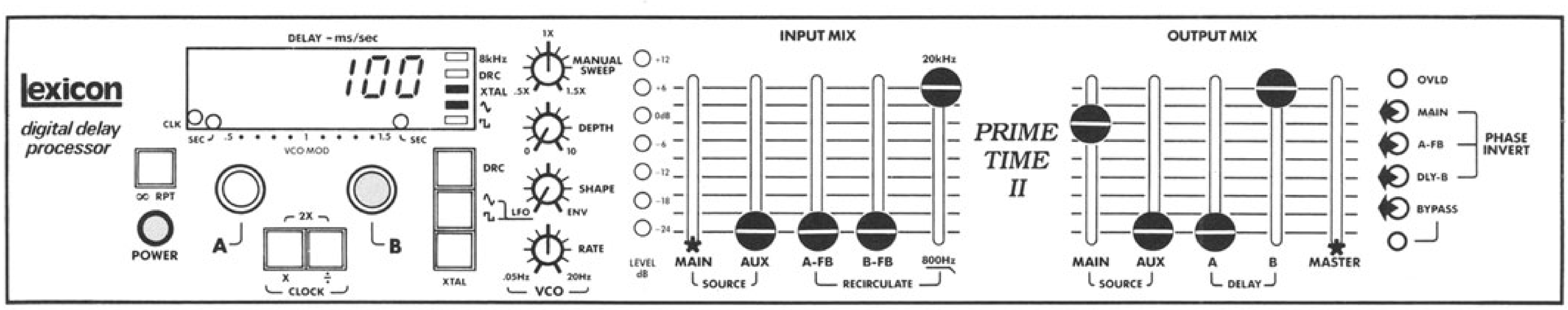
4.8 Discrete Echoes

Once delay times exceed 150 milliseconds or so, the repeats start to be heard as very distinct entities apart from the original, strongly suggesting acoustic reflections in outdoor environments such as mountains and canyons. This "echo" effect is certainly one of the most popular processing functions in popular music, and the Model 95 provides access to a very wide range of echoing sounds. Figure 4.22 shows several setups for discrete echoes, each having its own distinct quality.

4.9 Long Delay Loops and the Metronome Clock Feature

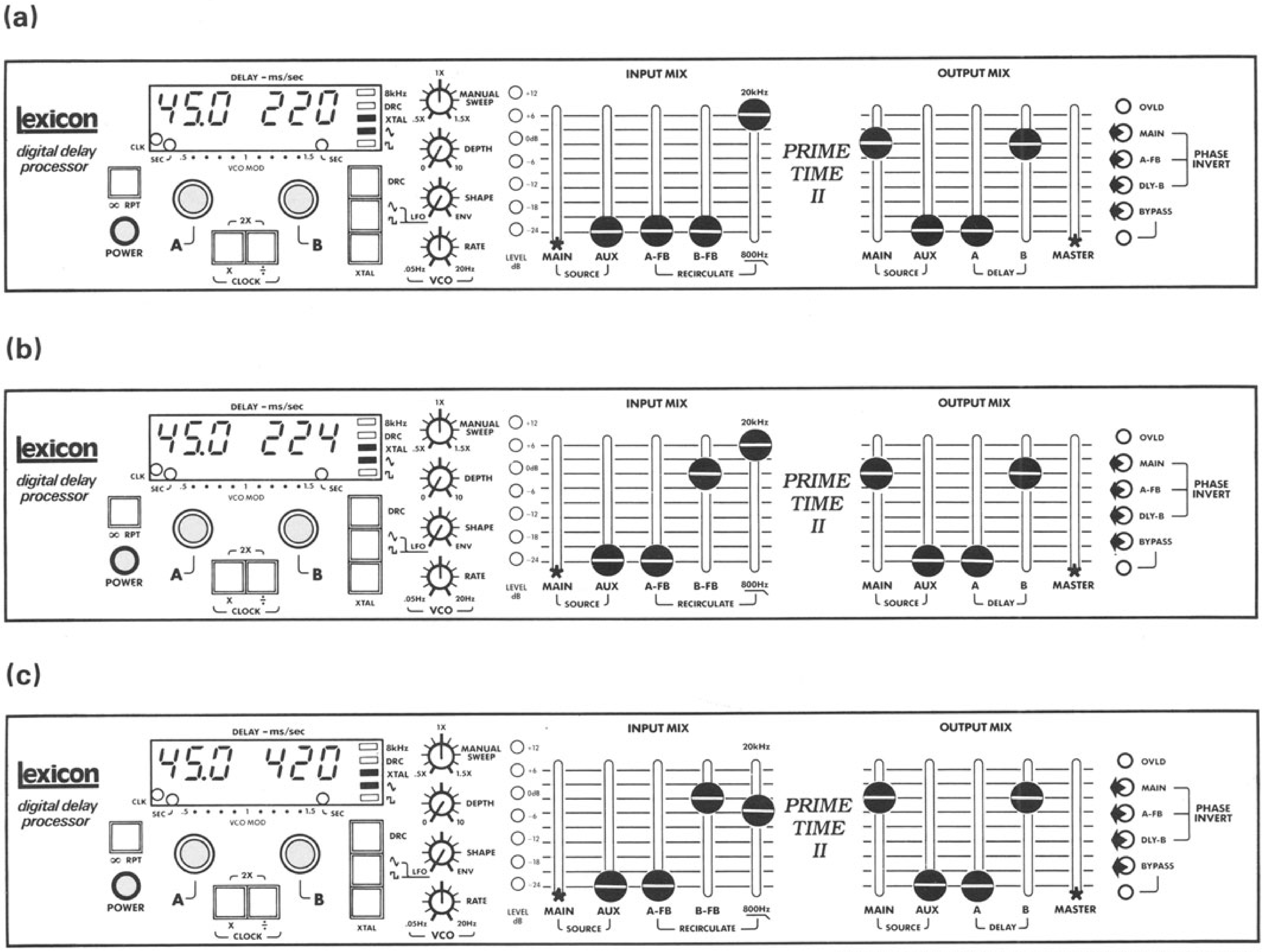
The very long delay times available with the Model 95, particularly with the different memory options, provide a range of functions that tend to very strongly influence the compositions in which they are used. When a repetitive delay of a couple of seconds or so is created with the Infinite Repeat function (perhaps also using recirculation), a delay loop results. The term *loop* is derived from the technique of using audio recording tape spliced together at the ends to form a loop. A large amount of music has been produced using bits of rhythm sections on such tape loops.

Many performers and composers have created original music by using tape loops in real time, inputing phrases, and then interacting with the repetitions. This technique is very effective (and probably more "roadworthy" as well) on an electronic delay like the Model 95. In addition, the Model 95 offers some significant features that can serve to greatly extend the use of loop techniques. Figure 4.23 shows a basic delay loop using the Model 95.



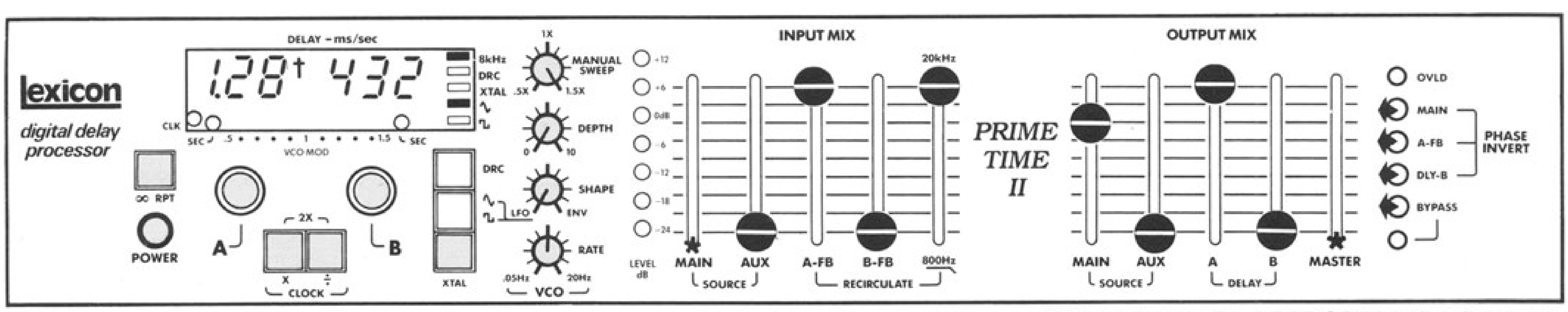
*ADJUST LEVEL APPROPRIATE TO INPUT SOURCES AND OUTPUT DEVICES USED.

FIG. 4.21. Slap echo.



*ADJUST LEVEL APPROPRIATE TO INPUT SOURCES AND OUTPUT DEVICES USED.

FIG. 4.22. Setups for discrete echoes.



†2.56 sec FOR MEMORY OPTION 1
5.12 sec FOR MEMORY OPTION 2

*ADJUST LEVEL APPROPRIATE TO INPUT SOURCES AND OUTPUT DEVICES USED.

FIG. 4.23. Basic delay loop

Most composers using loop techniques have been constrained by the difficulty of determining the relationship of tempo to delay time (or loop length). They have usually been limited to a very carefully set loop, with little capability for variation, or they have resorted to using only very slow attack times, so that an arhythmic style is produced. The Model 95 provides a facility for precisely coordinating delay time and tempo in the Clock Output feature. By using the Clock feature as a metronome, tempo can be set as a function of the delay length or, conversely, delay loop length can be set as a function of a desired tempo. This clock can be used both as a visual metronome by watching the CLK indicator lamp in the display window, or as an audible metronome by amplifying the Clock Output itself. It can be used as a master clock for most types of automatic drum units, sequencers, or synthesizers having built-in sequencer or arpegiator functions.

For a demonstration of how the Clock feature might be used in an actual application, refer to Fig. 4.24, which shows how to connect the Model 95 to other devices, and to Fig. 4.25 for the initial front-panel setup. Position the controls of the 95 for the settings shown in Fig. 4.25, with the XTAL pushbutton disengaged ("out") and both the MANUAL SWEEP and Delay Select B rotary controls in any position. Press both 2X/CLOCK pushbuttons simultaneously to enter double-delay mode, then press *one* of the 2X/CLOCK pushbuttons to enter the "Clock Program" mode ("CL OFF" will be displayed momentarily, followed by a display of previous Clock Program values—at power-up, these values are "01 01"). While the Clock Program values are displayed, press the [÷] pushbutton until the right side of the display reads "16". This divides the delay loop period into 16 "beats." Now press the [X] pushbutton until the left side of the display reads "48". This creates a rythmic pattern of 48 pulses per beat. While the new Clock Program values are still displayed, press both 2X/CLOCK pushbuttons again to enter the "Clock Output" mode. The automatic drum unit is now controlled by the number of pulses per beat output from the CLOCK OUTPUT jack of the 95. Adjust the MANUAL SWEEP control to create the desired tempo, and reprogram the number of beats (using the [÷] pushbutton in the same procedure as before) to match the output of the drum unit with the adjusted delay-loop interval.

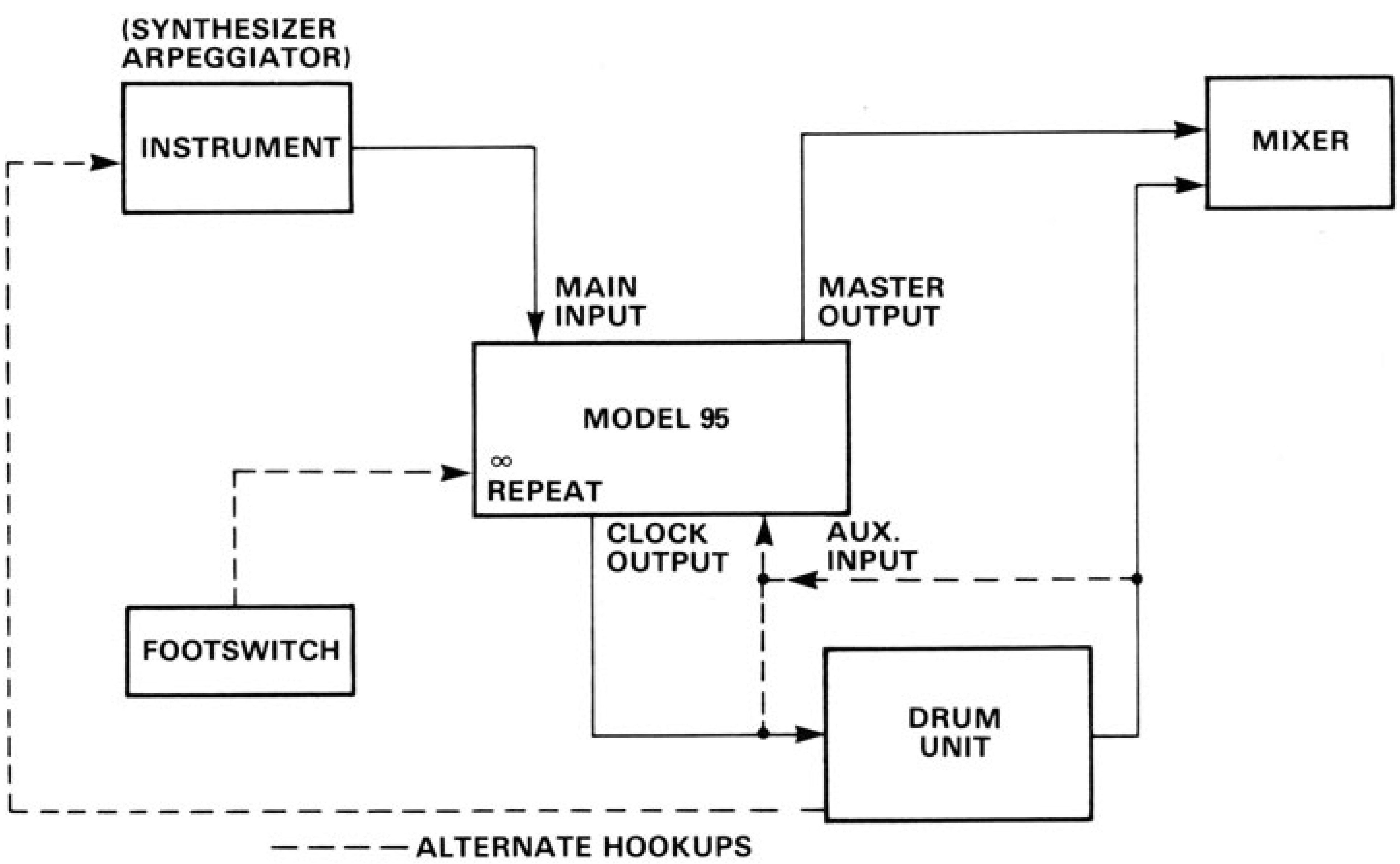
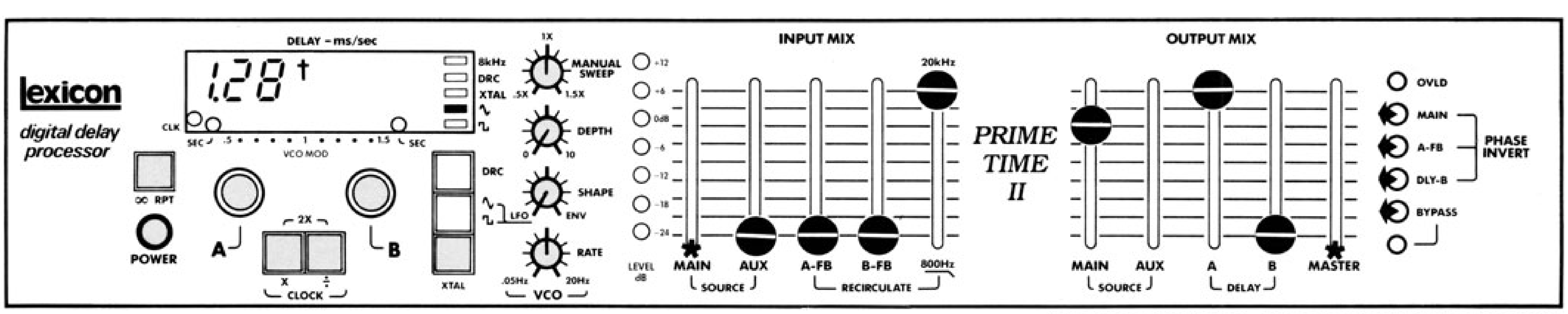


FIG. 4.24. Connection diagram for Clock function demonstration.



†2.56 sec FOR MEMORY OPTION 1
5.12 sec FOR MEMORY OPTION 2

*ADJUST LEVEL APPROPRIATE TO INPUT SOURCES AND OUTPUT DEVICES USED.

FIG. 4.25. Front-panel setup for Clock function demonstration.

Note: If you do not have access to an automatic drum unit, the CLOCK OUTPUT and AUX. INPUT of the 95 can be connected, and the AUX slider in the Output Mix section raised, so that the pulses per beat can be heard as a metronome. In this case, the number of pulses per beat does not have to equal 48, but can be any value that provides an effective beat for the tempo set by the adjusted delay loop period.

Now, start playing music into the 95 and bring up the A-FB slider all the way to the top. Experiment with different musical segments and phrases by lowering the A-FB slider all the way down to clear out the previous segment, then raising the slider again and playing a new segment. (To fade out a segment over time, only partially lower the A-FB slider.) When satisfied with the contents of the segment, engage the Infinite Repeat function (by pressing the ∞ RPT pushbutton on the front panel or pressing an optional footswitch control connected to the ∞ REPEAT jack on the rear panel) to capture the segment in memory. The captured segment will be repeated indefinitely until the Infinite Repeat function is disengaged. Using either the automatic drum unit as rythmic backing, the CLK LED as a visual metronome. or the amplified CLOCK OUTPUT as an audible metronome, play new material over the repeated segment for a variety of counterpoint or overdubbing effects. To capture layers of new material, raise the A-FB slider to the top. Disengage the Infinite Repeat function and play over the recirculated delay loop, then engage the Infinite Repeat function immediately after playing the new segment (equal in length to the delay loop period). Both the previous delay loop and the overdubbed material are captured in memory. This process can be repeated over and over again to build up layers and layers of rich, densely textured musical compositions.

To produce a syncopated delay loop, vary the setting of Delay tap B, and raise the DELAY B slider in the Output Mix section. Other variations could include the use of an output from the automatic drum unit as a control function for the arpeggiator feature of a synthesizer (any clocked audio device can be controlled in this fashion), or a simplified setup in which the output of the automatic drum unit is fed to the AUX. INPUT of the 95. The possibilities are endless. The attempt here has only been to give an example of how the Clock function can be used in an actual application.

4.10 Application of Time Delay in Sound Reinforcement

A common application of time delay is of special interest to sound reinforcement personnel and sound system designers. When speaker systems are distributed throughout a listening space to improve coverage, listeners are faced with sound coming to them from a variety of paths, with differing delay times (see Fig. 4.26). In large spaces, this can greatly affect intelligibility, and even in smaller rooms the effect can be very disconcerting because the sound is heard as originating from a remote speaker, rather than from the stage.

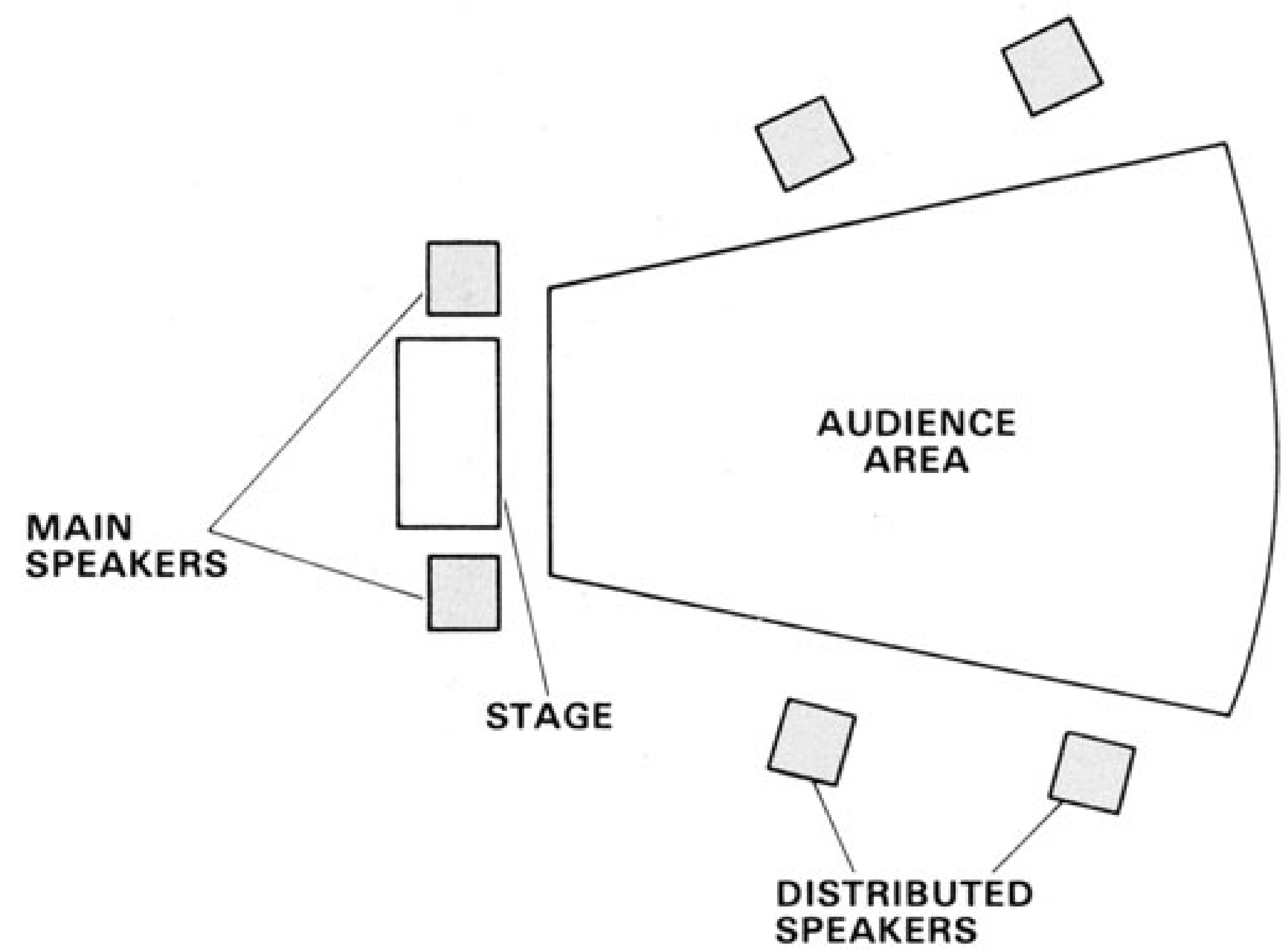


FIG. 4.26. Distributed speaker system.

If a time delay is introduced in the path to the remote speaker, the output of that speaker can be synchronized with the direct sound from the main speakers near the stage. This setup eliminates the echo effect produced by the different path lengths involved. If the delay time is increased a small amount beyond the point of synchronizing signals, the Hass effect will come into play, and the apparent origin of the sound will change from the remote speaker to the stage, even if a large amount of sound energy (not to exceed 10 dB greater than the original source) is actually arriving from the remote speaker! This allows systems to be set up with high apparent volume but without as much actual sound energy being projected about the room as would be required if all sound came from the stage speakers (see Fig. 4.27).

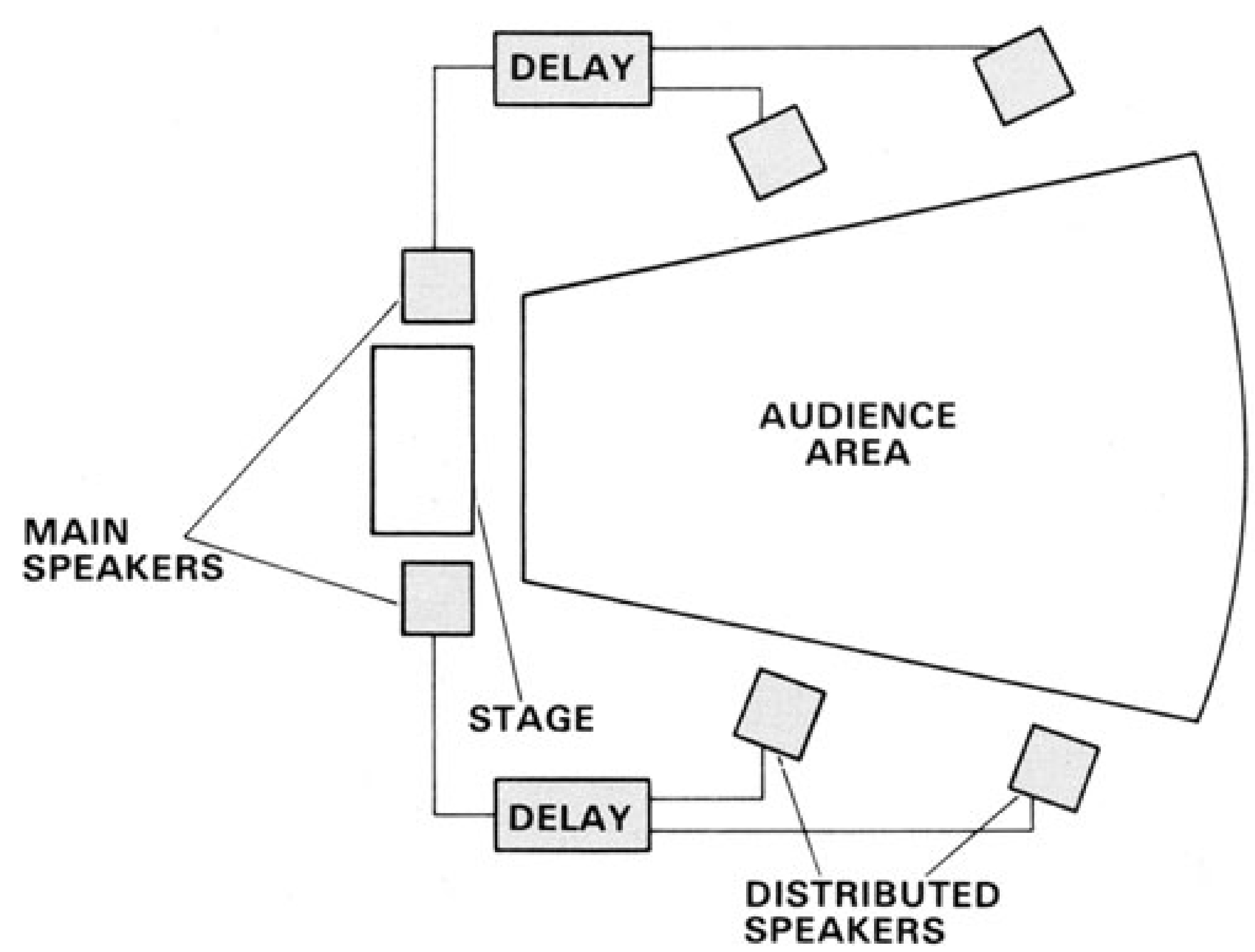


FIG. 4.27. Use of time delay in distributed speaker systems.

5
Detailed Description of Controls and Indicators

5.0 General

This section describes the Model 95's controls, connectors, and indicators. More detailed information about installation and applications is found in Sections 2 and 4, respectively. Signal levels in dBV are referenced to 0 dBV = 1 VRMS.

5.1 Front Panel

5.1.1 Delay Control Section

The Delay Control Section controls delay-tap selection and the delay range, Infinite Repeat, time-base modulation, and Clock functions (see Fig. 5.2). The Delay Control Section is divided into the Delay Select and Modulation Control Sections.

Delay Select Section

AC Power Switch [POWER] — This latching pushbutton turns the AC power on or off. Each time power is applied, the unit executes an initialization routine, consisting of a display of the model number (95), the revision number, and the memory option installed (0, 1, or 2). All discrete LEDs are turned on (except the BYPASS LED) as a test for burned-out lamps. This display holds for a few seconds while the initial contents of the delay memory (at power-up, the contents are random) are cleared. During this period, the delay time is held to 0, allowing the memory to clear without noise. The 95 is then readied for normal operation. Note that some of the functions are initialized in a fixed condition; i.e., at power-up, the 95 is in X1 (full bandwidth) mode, the XTAL clock is engaged (no modulation), the DRC function is disengaged, the LFO circuit is set to the sine-wave function, and the Infinite Repeat function is disengaged.

During the power-up sequence, the microcomputer also performs some simple tests of machine operation. If any faults are detected, "Err" followed by a three-digit diagnostic code is displayed. This is explained in Section 7.2.

Infinite Repeat Pushbutton [∞ RPT] — The Infinite Repeat pushbutton, when engaged, enables the Model 95 to capture the sound that is in the delay memory at the time it is pressed. That segment is continuously

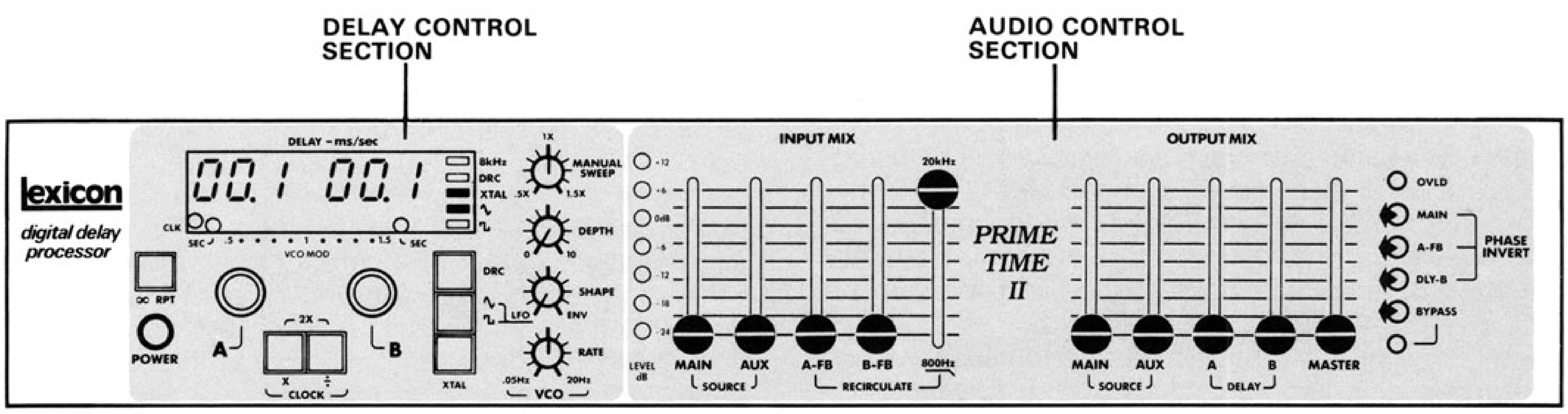


FIG. 5.1. Model 95 Prime Time II front panel.

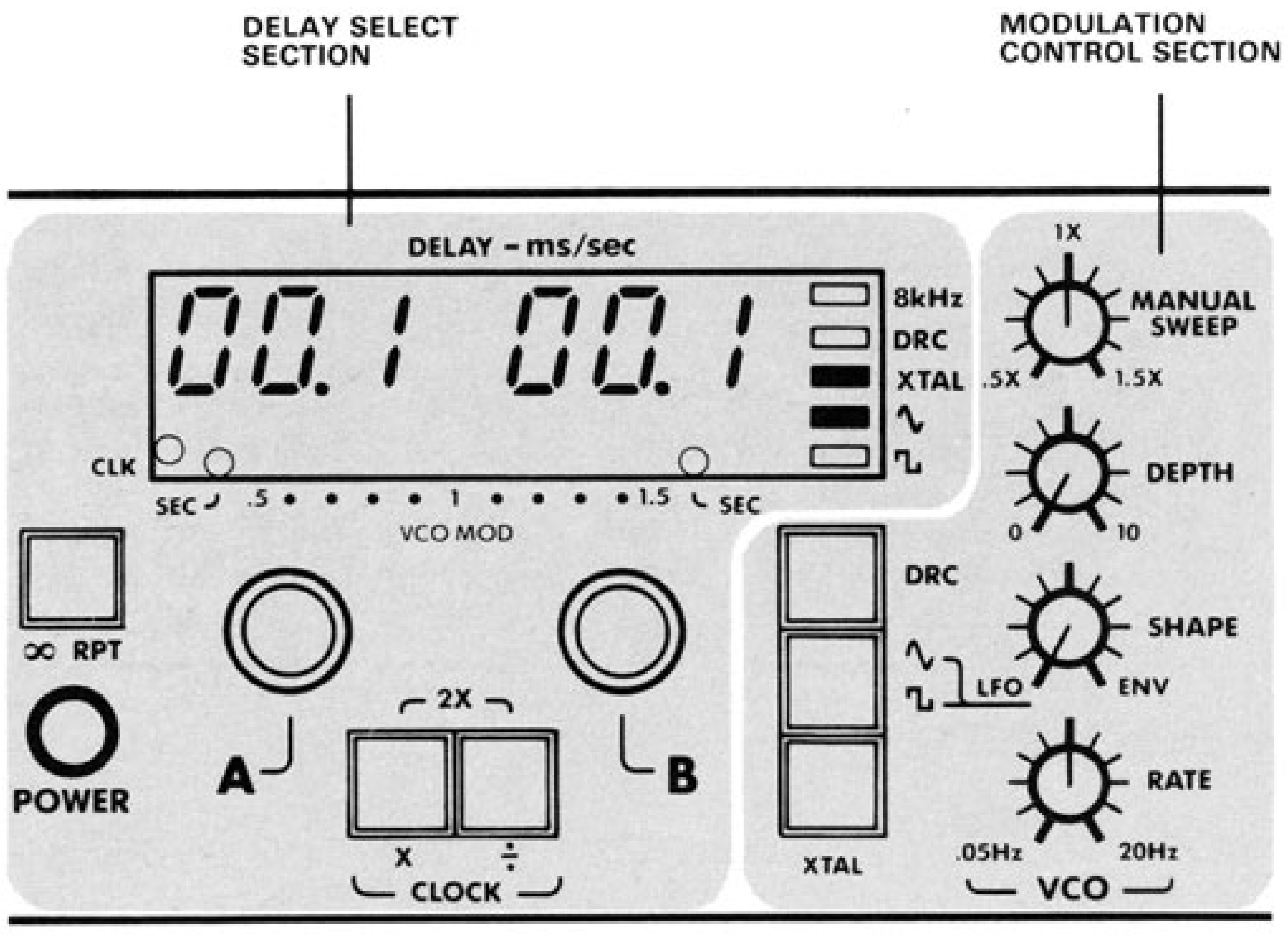


FIG. 5.2. Delay control section.

repeated until the mode is unlatched by pressing the Infinite Repeat pushbutton again. The time display and the word "Hold" are flashed alternately as a reminder that the Infinite Repeat function is engaged.

In the Infinite Repeat mode, the length of the segment captured is equal to the delay time that can be obtained from the maximum value of the full swing of one of the Delay Select controls (the value of this number is a function of the memory option installed), multiplied by both the delay range/bandwidth value (1X or 2X) and the delay value set by the modulation controls (if not in XTAL mode). Segment lengths from 0.64 to 7.68 seconds (depending on the modulation control settings and memory option) can be captured and stored.

In Infinite Repeat mode, the Delay-A and B rotary controls do not alter the length of the segment stored in memory, but they do offset the starting and ending points of two identical segment loops if varied with respect to each other, so that the memorized segment (delay loop) can be heard as two staggered versions. Many interesting, syncopated effects are possible using this property of the Infinite Repeat function.

Delay Select A and B [A,B] — The Delay Select A and B controls are used to set the location in the memory from which the two delay outputs will be read, and hence the amount of delay on each of the outputs (A and B). There are 128 delay settings available with each control. The same delay taps are available on both controls. The delay times have been selected to allow very fine control of the shorter delay times, becoming progressively coarser as delays become longer. The delay positions have also been calculated to provide even time divisions as well as uneven, or "prime" time divisions to maximize the randomness of the sound reflections, avoiding comb-filter effects or other relational artifacts that might cause unwanted coloration or otherwise detract from reverberation or other sonic ambiences.

2X/CLOCK Pushbuttons [2X/CLOCK] — This pair of pushbuttons performs dual functions. By pressing both pushbuttons simultaneously (pressing right between the two), the user selects the range of delay time available (1X or 2X) and the appropriate bandwidth (16 kHz or 8 kHz). Each time the pair is pressed, the unit toggles from its current state to the opposite state. The 8 kHz LED in the display window lights when the long-delay/reduced-bandwidth mode is engaged. Maximum delay available in either mode depends on the amount of optional memory installed, and is also affected by the MANUAL SWEEP control.

When the pushbuttons are operated individually, they are used to activate the Clock Output function, and to program its rate in relation to the memory length. The Clock Output rate is always expressed in terms of the length of the segments that will be captured in the Infinite Repeat mode, equal to the longest delay setting available with one of the Delay Select controls, multiplied by the delay range and modulation values. This is called the *loop time*. By pressing the [÷] pushbutton, the user can program the number of beats per loop (1, 2, 4, 8, or 16), allowing selection of the basic tempo for various segment lengths. The number of pulses that will be output for each beat (1, 2, 3, 4, 6, 8, 12, 16, 24, 32, 48, or 64) is selected by pressing the [X] pushbutton. This allows compatibility with a wide variety of drum units and sequencers. The CLK lamp in the display window flashes at the beat rate, and the CLOCK OUTPUT jack pulses at a rate equal to the beats per loop times the pulses per beat.

Note that when the Clock function is activated, the Infinite Repeat function operates synchronously with the "beat" rate. That is, if the Infinite Repeat pushbutton or footswitch is engaged between beats, nothing happens until the next time the CLK lamp flashes—that is, exactly on the next beat. This applies equally to capture as well as release of material, helping to ensure that loops of material can be used and layered without ugly "glitches" where the end points of the loops occur, since these points will be on a downbeat and generally masked by the beat of the music. When building up layers of sound, the end points are likely to fall at the same place in time, so there will be only one splice instead of many.

Pulses Per Beat [X]. This pushbutton and the one next to it can be used to program the Clock feature as follows. When the unit is first powered up, the Clock feature is disabled. If either one of the 2X/CLOCK pushbuttons (*not both*) is pressed, "CL OFF" is displayed for a couple of seconds, followed by a display of "01 01". This display reverts to the normal delay time display in a few seconds if no other keys are pressed, terminating the "Clock Program" mode. The Clock Program mode remains engaged, however, if the programming pushbuttons ([÷] and [X]) continue to be pressed.

The first two digits of the Clock Program mode display indicate the number of clock pulses that will be produced at the CLOCK OUTPUT jack on the rear panel during each "beat' 'of the delay loop period. This allows compatibility between the Model 95 and the majority of rhythm units and sequencing devices on the market. To set the pulses per beat, momentarily press the [X] pushbutton (on the left). Each time the [X] pushbutton is pressed, the left side of the display will increment through the series of available values (01, 02, 03, 04, 06, 08, 12, 16, 24, 32, 48, or 64 pulses per beat).

Beats Per Loop [÷]. While in the Clock Program mode (described in "Pulses Per Beat"), the number of beats to be produced for each delay loop period can be selected by pressing the [÷] pushbutton (on the right). Each time this pushbutton is pressed, the right side of the display will step through the series of available values (01, 02, 04, 08, or 16 beats per loop).

When the Clock Output mode is activated, the CLK lamp in the display window flashes at intervals equal to the delay loop period divided by the number of beats per loop, and the CLOCK OUTPUT jack on the rear panel outputs narrow pulses (+5 V, positive going) at that interval divided by the number of pulses per beat. To engage the Clock Output mode once the desired intervals have been programmed, press *both* pushbuttons together while still in the Clock Program mode. To release the Clock Output mode, enter the Clock Program mode as before and press both pushbuttons again.

The delay loop period itself is obtained in the normal delay-display mode by first rotating one or the other of the Delay Select controls fully clockwise. The delay time displayed is equal to the loop period. The variable time-base mode (Modulation Control mode) can be used to continuously vary the loop time. (Use the MANUAL SWEEP control to manually adjust the delay loop period, keeping the DEPTH control at "0", or fully counterclockwise.) The 2X function is used to double the length of the loop period. The range of the delay loop periods available is 0.64 to 1.92 seconds for a standard unit, and up to 7.68 seconds with Memory Option 2. The beat tempos range from only 8 beats/minute to 125 beats/minute at the longest loop times, and faster for all shorter loops. The frequency of the CLOCK OUTPUT ranges from 0.13 Hz, to over 600 Hz.

Dynamic Recirculation Control [DRC] — This pushbutton is used to select an automatic gain control in the recirculation path of the Input Mix section. When the DRC pushbutton is pressed, the unit will toggle into or out of DRC mode, as indicated by the DRC lamp on the right side of the display window. In DRC mode, the recirculated signals are routed through a gain-control stage which is driven by the signal level from the MAIN and AUX sliders in the Input Mix section. Recirculation is REDUCED at the highest signal levels and INCREASED, up to the level set by the A-FB and B-FB sliders, when the level drops below threshold. In setups using relatively long delays (1/4 second or more) and recirculation, the effect of this function is to provide signals which are "dry" during a passage or phrase, but which have a "tail" of echoes whenever a phrase ending is reached. This produces clean, articulated phases, with long decaying echoes. See Section 4 for setups using DRC.

Display Window — The display window groups together a number of important indications about the operation of the Model 95.

Delay-Time Display. The delay-time display consists of 6 digits, in two groups of 3, together with 2 discrete LED lamps. These functions normally indicate the exact delay time on each output at any given time. The display is in milliseconds (1/1000 of a second) unless the delay time exceeds 1 second, whereby the discrete SEC lamp for that output lights, indicating that the time is displayed in seconds.

Note: When the delay times are being swept by the automatic time-sweep functions, the display will track the sweep function and provide a readout of the instantaneous delay times at the slower sweep rates, and at the faster sweep rates, when the changes are too rapid to be visually intelligible, an averaging of the delay sweep times is displayed. At the faster sweep rates, when the average of the delay sweep times is displayed, the display will flicker. This is normal, and does not indicate a malfunction—it is simply a result of tracking the rapid changes in delay times so that the average value can be calculated.

These display indicators also function in certain modes to provide other information to the user. When programming the Clock Output, the display indicates the number of pulses per beat generated at the CLOCK OUTPUT jack and the number of beats per loop to which the CLK lamp flashes in synchronization. When the Infinite Repeat mode is activated, the display will alternate between the delay time indication and the word "Hold." On power-up, the display will temporarily display the model number (95), the revision number, and the number of the memory option installed (0, 1, or 2). If any problems are detected during power-up initialization, the display will indicate an error by displaying "Err" followed by a diagnostic code number indicating where the problem lies. See Section 7.2 for an explanation of these codes.

Clock Output Indicator [CLK]. When the Clock Output mode of the Model 95 has been engaged, this lamp flashes in time with the primary "beat," as programmed, for a visual indication of the basic tempo by which the delay loop is divided.

Modulation Display [VCO MOD]. When the variable time-base mode (Modulation Control mode) has been selected using the XTAL pushbutton (to disengage the crystal oscillator and engage the VCO or voltage-controlled oscillator) this group of 20 LED indicators will light, one at a time, to indicate the affect of the Modulation Control section on the delay time. The LEDs will display the position of the sweep, bar fashion, as the delay times are varied (swept) either manually or automatically. The motion of the light provides a visual portrayal of the selected sweep characteristics and rate. This sweep portrayal is known as the "flying spot" display.

Status Indicators — This group of 5 LED indicators show the status of particular control functions as follows:

Bandwidth/Delay Range Indicator [8 kHz]. When this lamp is lit, the device is operating in its long delay mode (X2). The delay time available is twice that of normal mode, and the bandwidth is reduced from 16 kHz to 8 kHz to prevent aliasing distortion in extremely long delays.

Dynamic Recirculation Indicator [DRC]. When lit, this lamp indicates that the Dynamic Recirculation Control feature, previously described, is active.

Fixed Time-Base Indicator [XTAL]. This lamp indicates that the unit is operating on its fixed time base (the crystal oscillator is engaged), and will not be affected by any of the controls in the Modulation Control section. When this lamp is *not* lit, the VCO, or variable time base is engaged, and any of the sweep functions can be used.

Sine-Wave Modulation Indicator [∿]. When lit, this lamp indicates that the LFO or low-frequency oscillator is generating sine-wave output as a modulation function.

Square-Wave Modulation Indicator [⊓⊔]. This lamp indicates that the LFO is generating square-wave output as a modulation function.

Modulation Control Section

The Modulation Control section of the front panel is used to generate "swept" or continuously variable delay. Flanging, double tracking, pitch twisting, and vibrato setups, for instance, all make use of sweep functions. In long delay loop applications, the MANUAL SWEEP control is often used to control the length of the delay loops.

Fixed/Variable Time-Base Select [XTAL]

This pushbutton serves to activate or deactivate the rest of the controls in this section, by selecting whether the unit operates on a fixed (XTAL) or variable (VCO) time base. When the fixed time base is selected, the XTAL lamp in the display window lights. In this mode, none of the other controls in this section will have any effect on the sound being produced. When the XTAL lamp is off, all functions of the Modulation Control section are enabled, and the VCO MOD indicator lights to track the modulation function in effect.

Low-Frequency Oscillator (LFO) Waveform Select [∿/⊓⊔]

This pushbutton is used to select the type of waveform output from the low-frequency oscillator. This output is used in conjunction with the output from the SHAPE control circuit as an automatic sweep control source. Both sine-wave (smooth, continuous sweep) and square-wave (alternating step sweep, up and down) functions are available. The waveform selected is indicated by an LED lamp in the display window.

Manual Sweep Control [MANUAL SWEEP]

This rotary control serves to sweep delay time manually over a range of 3 to 1, from 0.5 to 1.5 times the set delay values. It is often used as a fine adjustment of delay time, because it is continuous in operation and affects both taps equally. This control is also used normally in long delay loop applications to control the length of the delay loops. It can also be used to manually sweep flanging, resonance, chorusing, or pitch-twisting setups.

Modulation Depth Control [DEPTH]

This rotary control is used when automatic sweep is desired, such as in normal flanging or doubling. The automatic modulation, or sweep, is disengaged when the control is in the fully counterclockwise position. As the control is rotated clockwise, the modulation function from the SHAPE control is fed to the variable clock, causing audible sweep. At maximum (full clockwise) position, the modulation sweeps delay times over a range of 3 to 1 (from 0.5 to 1.5 times the set delay values). Both taps are affected equally, and it should be noted that as DEPTH is increased, the range of the MANUAL SWEEP control diminishes until DEPTH reaches maximum, when the MANUAL SWEEP control has no effect at all. It is important to note also that the audible effect of a given amount of modulation increases drastically as longer delay times are used. Generally speaking, the DEPTH control must be reduced as longer delays are selected, unless extreme effects are desired.

Modulation Mix Control [SHAPE]

This rotary control determines the mix of the two available modulation sources to be routed through the DEPTH control. If the SHAPE control is rotated fully counterclockwise, the modulation comes entirely from the low-frequency oscillator, producing either sine-wave or square-wave modulation as selected by the LFO pushbutton. As the control is rotated clockwise, a control voltage derived from the level of the Input Mix (MAIN and AUX. inputs, plus recirculation) by an envelope-follower circuit is blended with the LFO output. The envelope-follower function provides an automatic time sweep based on the level of the input signal. The highest signal levels produce the shortest delay times of the delay range determined by the setting of the DEPTH control; the lowest signal levels produce the longest delay times, and intermediate levels result in intermediate delay times. When the SHAPE control is fully clockwise (ENV position), the modulation source will be the envelope-follower function alone. The output of the SHAPE control is available to the user at the MOD OUTPUT jack on the rear panel.

LFO Frequency Control [RATE]

This rotary control varies the rate of sweep by the low-frequency oscillator from 0.05 Hz (one complete sweep cycle every 20 seconds) to 20 Hz (twenty sweep cycles per second).

Note that in order for the RATE and SHAPE controls and sine wave/square wave selection (LFO pushbutton) to have any audible effect, the DEPTH control must be in other than the "0" (counterclockwise) position and the XTAL lamp must be off (i.e., XTAL pushbutton disengaged).

5.1.2 Audio Control Section

The controls in the Audio Control section regulate the flow of audio signals around and through the delay line (see Fig. 5.3).

Input Mix Section

The Input Mix section determines the combination of signals that will be fed to the delay line and indicates the internal level of signal feeding the delay. Some signal processing, in the form of equalization and phase inversion, can also be applied to specific recirculation signals.

Headroom Indicator [LEVEL dB]

This group of 7 LED lamps indicates the level of the Input Mix signals entering the delay line in relation to the level at which the analog-to-digital converter begins clipping. The principal use of this display is to allow the user to optimize audio performance by adjusting the Input Mix levels to get the highest signal level available without clipping. Generally speaking, average levels should be set for "0 dB" (the lowest amber LED), so that 12 dB of headroom is available for peaks. If the LIMITER switch on the rear panel is engaged, Input Mix levels can be set much higher without fear of clipping because above-threshold levels will be compressed to maintain levels below the clipping point of the A/D converter. With signals that feature a lot of dynamics, either the LIMITER circuit should be engaged, or average levels should be set very conservatively to ensure adequate headroom. Note that the signal-to-noise ratio and THD (total harmonic distortion) of audio devices are compromised when lower input levels are used.

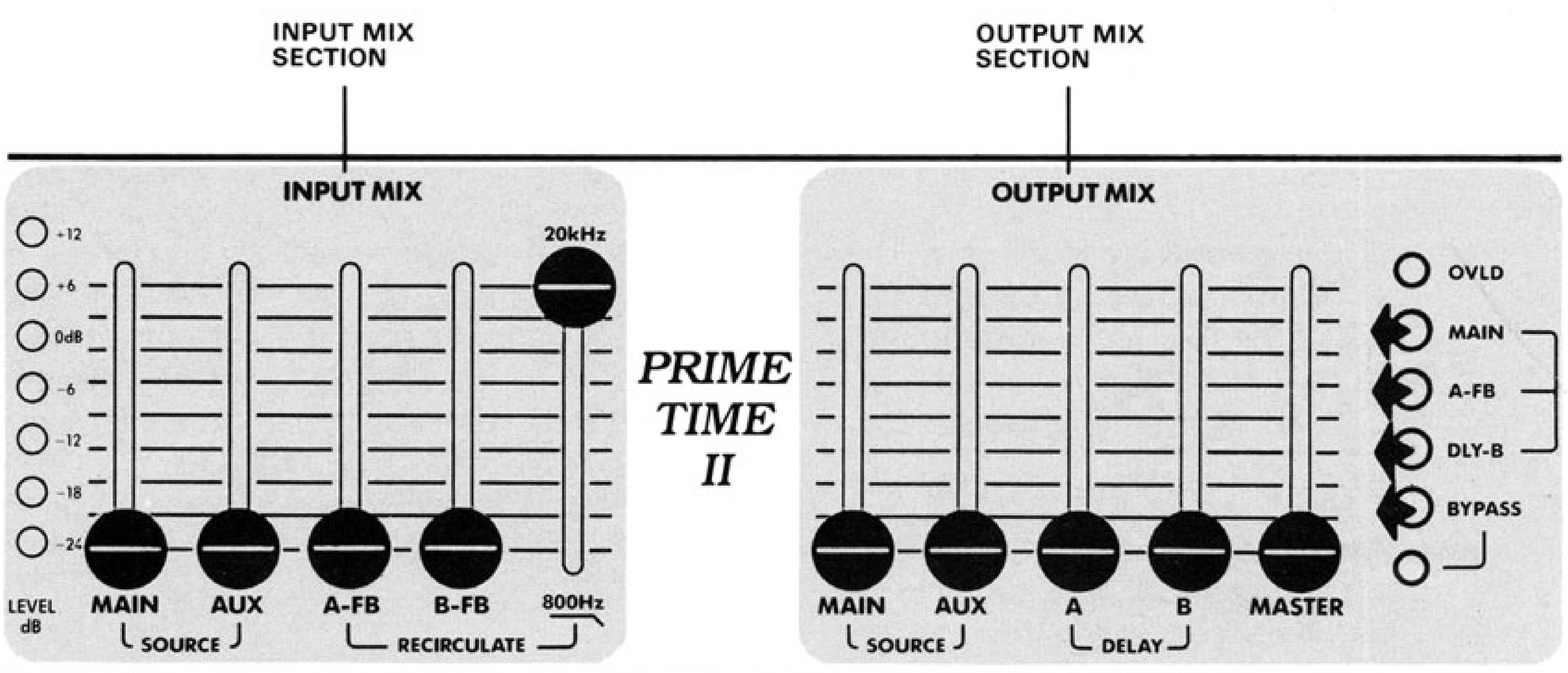


FIG. 5.3. Audio control section.

Main Input Level Control [MAIN]

This slider controls the amount of signal from the MAIN INPUT preamplifier section that feeds the delay processor. This control should be set, as previously described in the discussion of the headroom LEVEL indicators, so that average levels light the green LEDs in the headroom indicator display, and peak levels light the amber lamps. If the signal entering the unit is too low to light the appropriate LEDs, the GAIN switch on the rear panel should be engaged.

Auxiliary Input Level Control [AUX]

This slider controls the level of signal entering the delay processor from the AUX. INPUT preamplifier. Note that this input has no gain circuit, and so should be used for line-level signals only.

Delay-A Recirculation Control [A-FB]

This slider controls the level of signal from Delay-A that is mixed into the input of the A/D converter. The gain of this channel, as well as that of the B-FB channel, has been selected to be a little below unity gain when the slider is all the way up to the top of its travel. This means that the delay line will not "run away" when either one of the recirculation controls is fully raised. *However*, if both sliders are up at the same time, the net gain of the feedback can easily exceed unity, and the recirculated signal will be boosted each time through the unit, causing ugly, speaker- and ear-damaging howling. Always be cautious when using both recirculation sliders together.

Delay-B Recirculation Control [B-FB]

This slider controls the level of signal from Delay-B that is mixed into the input of the A/D converter, and functions in the same way as the Delay-A recirculation control (A-FB).

Recirculation Cutoff Frequency Control
[800 Hz–20 kHz]

This slider adjusts the cutoff point of a 6-dB/octave low-pass filter in line with the combined recirculation signals. The range of the control is from 800 Hz to 20 kHz. This control is useful for "soft" echos, ambience effects, and for taking the "edge" out of the high end of resonance and resonant flange effects. It also helps reduce the potential for high-frequency howling when high volumes of recirculation are used.

Note here that two functions described elsewhere directly affect the signals in the Input Mix section: the DRC function in the Delay Select section modulates the level of recirculated signal, and the A-FB PHASE INVERT toggle switch in the Output Mix section causes the polarity of the recirculation of Delay-A to be inverted.

Output Mix Section

This group of controls determines the mix of signals output from the MASTER OUTPUT. Any combination of dry or delayed signals can be selected as output.

Main Input Level Control [MAIN]

This slider controls the level of the signal from the MAIN INPUT that is blended with the Output Mix signals and fed to the MASTER OUTPUT.

Auxiliary Input Level Control [AUX]

This slider controls the level of signal from the AUX. INPUT that is blended with the Output Mix signals and fed to the MASTER OUTPUT.

Delay-A Output Level Control [A]

This slider determines the level of Delay-A that is blended with the Output Mix signals and fed to the MASTER OUTPUT.

Delay-B Output Level Control [B]

This slider controls the level of Delay-B that is blended with the Output Mix signals and fed to the MASTER OUTPUT.

Master Output Level Control [MASTER]

This slider regulates the overall level of signal at the MASTER OUTPUT. By varying this control, the Model 95 can be operated effectively with any type of amplifier or console without overloading that device's input stage. The gain of this stage can be varied from minus infinity (no output at all) to about 10 dB above system level (+22 dBV maximum output level for a balanced load).

Output Mix Overload Indicator [OVLD]

This lamp lights whenever the combination of signals fed to the MASTER control exceeds the level which that stage can handle without audible distortion. Signal overload can be corrected by lowering the level of the appropriate combination of the four inputs to the MASTER control circuit (i.e., lowering the MAIN, AUX, DELAY A, and/or DELAY B sliders). The MASTER control can only raise the level of the combined signals or lower the level fed to the MASTER OUTPUT, but it cannot affect the level of the four inputs before the MASTER control circuit stage.

Main Input Signal Polarity Select Switch [MAIN]

When this toggle switch is pushed to the right-hand side, the phase (polarity) of the dry signal from the MAIN INPUT that is fed into the MASTER OUTPUT is inverted.

Delay-A Recirculation Polarity Select Switch [A-FB]

This switch inverts the phase of the signal from Delay-A that is recirculated back to the Input Mix section.

Delay-B Output Polarity Select Switch [DLY-B]

This switch inverts the phase of the Delay-B signal fed into the MASTER OUTPUT.

Note that neither the A-FB PHASE INVERT nor DLY-B PHASE INVERT switches has any effect on the signals present at the DELAY-A OUTPUT or DELAY-B OUTPUT jacks on the rear panel.

System Bypass Select Switch and Indicator [BYPASS]

The BYPASS switch on the front panel, when pushed to the right-hand position, causes all signals output from the delay line to be switched out of the INPUT MIX OUTPUT and MASTER OUTPUT. The Delay-A and Delay-B signals are still available at the DELAY-A OUTPUT and the DELAY-B OUTPUT, respectively, but only the main and auxiliary input signals are available at the INPUT MIX OUTPUT and the MASTER OUTPUT. The BYPASS LED indicator lights when Bypass mode is engaged.

5.2 Rear Panel

5.2.1 Audio Control Section

Audio Signal Inputs

Main Input Connector [MAIN INPUT]

This XLR female connector accepts balanced or unbalanced audio input signals (pin 2 = HIGH, pin 3 = LOW, pin 1 = GND). Peak signals from 0 to + 19 dBV can be handled with full dynamic range. The input impedance is 50 kΩ in parallel with 30-pf capacitance (unbalanced) or 100 kΩ in parallel with 150 pf (balanced).

Main Input Gain Control Switch [GAIN]

When depressed, this pushbutton increases the gain at the MAIN INPUT by 20 dB, allowing signals ranging in level from −1 to +20 dBV to be accepted.

Auxiliary Input Jack [AUX. INPUT]

This 1/4″ phone jack carries balanced or unbalanced signal into the auxiliary input stage of the Model 95. Signal levels between 0 to +19 dBV can be handled with full dynamic range. The impedance is 20 kΩ in parallel with 150-pf capacitance (unbalanced) or 40 kΩ in parallel with 100 pf (balanced).

Limiter In/Out Switch [LIMITER]

When engaged, this latching pushbutton switch activates a two-step limiter circuit at the input stage of the delay line. When signals feeding the delay reach a level 4 dB below the clipping point of the A/D converter (clipping point = +12 dBV), the summed input signals begin to be compressed at a 4:1 ratio. When the level 1 dB below clipping point is reached, the compression ratio increases to 18:1. This prevents inadvertent overloading of the converter input.

Audio Signal Outputs

Input Mix Output Jack [INPUT MIX OUTPUT]

This 1/4″ phone jack carries the combination of signals that is fed to the delay line from the Input Mix controls. This consists of the dry inputs (MAIN and AUX), and any delay signal that is being recirculated. This output is single-ended, line level (maximum level = +12 dBV), with an output impedance of 600 Ω . Note that if the LIMITER switch is engaged, this output will be affected by the initial 4:1 compression that cuts in when signal levels exceed the threshold of +8 dBV. Note that the

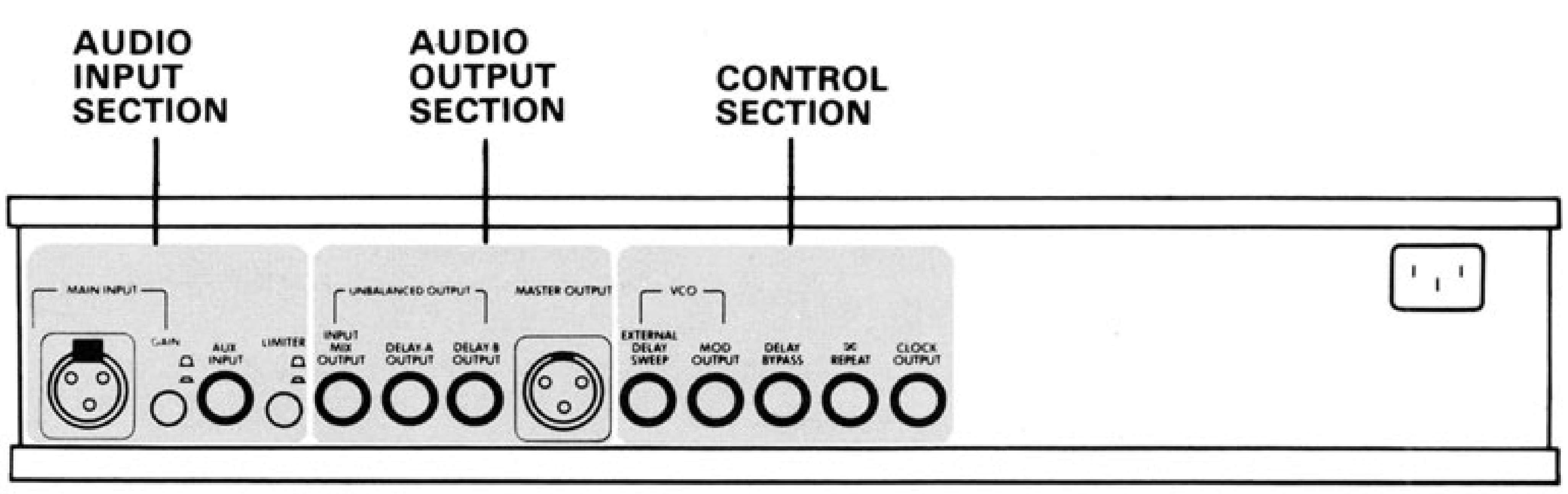


FIG. 5.4. Model 95 Prime Time II rear panel.

DRC function has no affect on the recirculated signals in the Input Mix section fed to this output.

Delay-A Output Jack [DELAY-A OUTPUT]

This phone jack carries the output of Delay-A, unmixed with any other signal. It is a line-level (maximum level = +12 dBV), unbalanced output with a 600-Ω source impedance.

Delay-B Output Jack [DELAY-B OUTPUT]

This jack carries the unmixed output of Delay-B. It has the same characteristics as the DELAY-A OUTPUT jack.

Master Output Connector [MASTER OUTPUT]

This XLR female connector carries the combination of signals formed by the Output Mix section. This output is electronically balanced, with 200-Ω actual source impedance, designed to drive inputs with impedances of 600 Ω or higher. The signal level is completely adjustable by the Output Mix section, up to approximately +22 dBV (balanced loads), or +16 dBV (unbalanced loads).

5.2.2 Control Inputs and Outputs

The connections in this group provide means to remotely control some of the system functions, and to interface the Model 95 with the control paths of other types of devices.

VCO Control Jacks [VCO]

This group consists of two phone jacks, one input and one output, that allow for external control of modulation functions, and coordination of the 95's sweep with other devices.

VCO Control Voltage Input [EXTERNAL DELAY SWEEP]

This input jack allows the user to remotely operate the MANUAL SWEEP control with a foot pedal (for example, the optional Lexicon Delay Controller A-FS-41), any potentiometer, or a voltage source in the range of 0 to +10 V. When a plug is inserted in this jack, the MANUAL SWEEP control on the front panel is deactivated.

Modulation Waveform Output [MOD OUTPUT]

This stereo output jack carries the mix of the low-frequency oscillator and envelope-follower modulation sources that is set by the SHAPE control on the front panel. The signal peaks range from 0 to +10 V. The ring portion of the jack carries a phase-inverted version of the same signal.

Delay Bypass Control Input [DELAY BYPASS]

This input jack allows the user to remotely switch the Bypass function with a footswitch (for example, the optional Lexicon A-FS-41 Dual Footswitch or A-FS-97 Single Footswitch). When the function is activated by a switch closure, the effect is the same as pressing the BYPASS switch on the front panel.

Infinite Repeat Control Input Jack [∞ REPEAT]

This jack allows a momentary switch closure to toggle the unit into or out of the Infinite Repeat mode. This function may be operated with an optional Lexicon A-FS-97 Single Footswitch or A-FS-41 Dual Footswitch Assembly (BYPASS and Infinite Repeat switches in one assembly). This function can also be operated by a standard 0 to +5 V logic signal (the low-going edge initiates toggling).

Clock Output Jack [CLOCK OUTPUT]

This jack carries a periodic pulse (peak amplitude = +5 V) when the Clock function is programmed and activated from the front panel. This narrow pulse (width = approximately 100 microseconds) is capable of driving clock inputs on most types of percussion units and sequencers in use today. It can also be amplified and used as an audible metronome output.

AC Connector

This IEC standard connector provides AC power connection using the supplied power cord or a similar type.

6 Specifications

6.0 General Performance

Total Distortion and Noise @ 1 kHz

- 0.03% typical, 0.05% maximum
- 0.3% maximum @ −30 dB
- 0.1% maximum, 20 Hz to 10 kHz

Frequency Response

- Measured with input level 12 dB below reference level.
- 1X mode: 20 Hz to 16 kHz, +0.5, −2 dB
- 2X mode: 20 Hz to 8 kHz, +0.5, −3 dB, referenced to 1 kHz.

Dynamic Range

- 90 dB typical, 86 dB minimum, 20-Hz to 20-kHz noise bandwidth in XTAL or VCO clock mode.

Delay Selection

- Two delay ranges are available: X1 (16-kHz bandwidth) and X2 (8-kHz bandwidth). Individual rotary controls, each with 128 selectable delay values, can select two independent delay taps. Both taps can be continuously varied over a 3-to-1 range (0.5 to 1.5 times the normal setting) using the MANUAL SWEEP or DEPTH controls.

Delay Capacity

Memory Option	VCO @ 0.5X		VCO @ 1X or XTAL		VCO @ 1.5X	
	1X	2X	1X	2X	1X	2X
Standard	320 ms	640 ms	640 ms	1.28 s	960 ms	1.92 s
Option 1	640 ms	1.28 s	1.28 s	2.56 s	1.92 s	3.84 s
Option 2	1.28 s	2.56 s	2.56 s	5.12 s	3.84 s	7.68 s

ms = milliseconds; s = seconds.

Delay Step Size

- Approximately exponential, with fine adjustment available at small delay settings, with larger increments as delay is increased.

Delay Display

- Each tap is displayed with 3-digit, 7-segment LED readout.

Display Accuracy

- Plus or minus 500 microseconds for settings up to 100 milliseconds, ±0.5% for settings between 100 milliseconds and maximum delay in XTAL mode.

Delay Stability

- After 5-minute warm-up period, from 25 to 50°C (77 to 122°F), ±0.01% in XTAL mode and ±1.5% in VCO mode

VCO Modulation

- Depth is adjustable from none (zero modulation of delay time) to a full 3:1 sweep of delay time. Low-frequency oscillator rate is adjustable from 0.05 Hz (20 seconds for full sweep) to 20 Hz. Modulation is displayed by a 20-segment bar-graph.

VCO Modulation Waveform

- Continuous adjustment (blend) is available between sine-wave and envelope-follower or square-wave and envelope-follower functions.

Test Conditions

Unless otherwise stated, all audio measurements are taken using a 1-kHz, 0-dBV (1 VRMS) signal source with the input and output levels referenced to the following settings: +20 dB GAIN and LIMITER switches "out," MAIN slider in the INPUT MIX section raised to the point where the "+12 dB" LEVEL LED lights (just before onset of A/D converter clipping as monitored from the DELAY-A OUTPUT jack), all other INPUT MIX sliders lowered to minimum, and the DELAY A or B slider in the OUTPUT MIX section raised to maximum, with the MASTER slider adjusted to obtain +12 dBV into 600 Ω at the MASTER OUTPUT.

Unless otherwise stated, all measurements are taken with the unit operating at room temperature.

Unless otherwise stated, all audio measurements are taken with the unit operating in XTAL mode at 16-kHz bandwidth, with LIMITER off, DEPTH control set to "0", BYPASS off, and all PHASE INVERT switches off.

6.1 Interfaces

Inputs

- Balanced differential inputs: MAIN INPUT via XLR-3 female connection, AUX. INPUT via standard 1/4″ tip-ring-sleeve phone jack, with 50-dB minimum common-mode rejection. Unbalanced inputs are also accepted.

Input Impedance

- Greater than 50 kΩ in parallel with 300 pF for MAIN INPUT, balanced or unbalanced. Greater than 20 kΩ in parallel with 150 pF for AUX. INPUT, balanced or unbalanced.

Input Level

- MAIN INPUT
  +20 dB GAIN switch out: 0 to +19 dBV
  +20 dB GAIN switch in: −20 to 0 dBV

- AUX. INPUT
  0 to +19 dBV

Input Limiting

- A dual-slope limiter activated via a push-push switch allows approximately 20 dB of additional headroom without harsh clipping of input sources, greatly simplifying operation in live performance applications. Between the preamplifier and anti-aliasing filter is a compression stage with the threshold at 4 dB below reference level and a compression ratio of 4:1. Between the filter and the sample-and-hold circuit, there is a limiter with threshold 1 dB below reference level and a composite compression ratio of 18:1.

Outputs

- The MASTER OUTPUT is a balanced source into an XLR-3 male connector. The other outputs, INPUT MIX OUTPUT, DELAY-A OUTPUT, and DELAY-B OUTPUT, are unbalanced and have standard 1/4″ tip-sleeve phone jacks.

Output Impedance

- MASTER OUTPUT: 200 Ω balanced or unbalanced actual source impedance.

- INPUT MIX, DELAY-A, DELAY-B OUTPUTS: 600 Ω actual source impedance.

Output Level

- MASTER OUTPUT: +22 dBV (12.5 VRMS) maximum, when driving balanced loads 600 Ω or greater, and +16 dBV (6.3 VRMS) maximum, when driving unbalanced loads 600 Ω or greater.

- INPUT MIX OUTPUT: +16 dBV maximum, when driving loads 2 kΩ or greater.

- DELAY-A OUTPUT: +8.5 dBV maximum, when driving loads 2 kΩ or greater.

- DELAY-B OUTPUT: +8.5 dBV maximum, when driving loads 2 kΩ or greater.

Remote Jacks

Rear-panel 1/4″ phone jacks accommodate external switches and controls as follows:

EXTERNAL DELAY SWEEP: This stereo phone jack can be connected to a potentiometer or a voltage source. Whenever a plug is inserted in the jack, the MANUAL SWEEP control is disabled and replaced by the external control. A voltage source of 0 to +10 V can feed the input, with 0 V corresponding to 1.5X, +5 V to 1X, and +10 V to 0.5X.

MOD OUTPUT: This stereo phone jack can be used to slave other Lexicon digital delay processors or other devices so that multiple units can be modulated from the same source. The signal at this jack is derived from the output of the SHAPE control and is 0 to + 10 V, and is available in-phase (tip-sleeve connection) and out-of-phase (ring-sleeve connection.).

DELAY BYPASS: At this jack, when the tip of the plug is grounded to the sleeve, the Model 95 is placed in Bypass mode. In Bypass mode, the recirculated signals are switched out of all outputs, and Delay-A and Delay-B signals are fed to the DELAY-A OUTPUT and DELAY-B OUTPUT, respectively.

INFINITE REPEAT: At this jack, when the tip of the plug is momentarily grounded to the sleeve, or connected to a low-level TTL pulse, the Model 95 captures the segment of the input that is in memory and repeats that segment indefinitely. A subsequent momentary contact or pulse input returns the unit to normal operation.

CLOCK OUTPUT: This is a TTL-level timing pulse that is synchronized to the delay interval. This output can be used to cue a musician or clock an external sequencer or rhythm unit.

6.2 Power and Dimensions

AC Requirements

- 115 or 230 VAC ±10% (selectable), 50-60 Hz, 40 W maximum. Export model available with 100-V option. An RFI mains filter is installed.

Protection

- Mains are fused (standard U.S. 3AG fuses). For Export model, mains and secondaries are fused (European style: 20-mm fuses).

Dimensions

- Standard 19-inch relay rack:
  19" (w) by 3 1/2" (h) by 13 1/2" (d)
  (483 mm by 89 mm by 343 mm).

Environment

- Operating: 0 to 35 °C (32 to 95 °F);
  Storage: −30 to 74 °C (−22 to 167 °F);
  Relative humidity: 95% maximum (without condensation).

Weight

- Net: 10.5 lbs (4.76 kg)
  Shipping: 13.5 lbs (6.12 kg).

7
Service and Warranty

7.0 General

Before attempting to verify and isolate a problem with the unit, it is important to understand the operating and installation information presented in this manual.

WARNING

All service of the Model 95 Prime Time II should be performed by qualified service personnel. There are hazardous voltage sources located under the top and bottom covers of the unit. To avoid electrical shock, remove the power cord before removing covers. Procedures consistent with good safety practices should be used at all times.

7.1 Unit Will Not Power Up

If the Model 95 will not power up, first check the AC line cord to ensure it is securely plugged into the Model 95 and the service outlet. Verify that the service outlet is live and that the voltage is correct for the Model 95 (use a voltmeter, neon test light, or a common lamp). If correct voltage is present, unplug the Model 95, and refer the problem to an authorized Lexicon service technician, or return the unit to Lexicon.

7.2 Diagnostic Messages

During the power-up sequence, the microcomputer of the Model 95 performs tests of the internal functions. If errors are found during these tests, the unit will display "Err", followed by a diagnostic code number indicating the source of the error. These codes isolate the faults as follows:

- 001 = CPU (Central Processing Unit)
- 002 = RAM (Random Access Memory)
- 003 = ROM (Read-Only Memory)

Note the digits being displayed, and refer to this diagnostic code in any communication with Lexicon service personnel or technicians.

7.3 Unit Powers Up, Gives Improper Display

If, at any time, an unintelligible display appears in the display window, or there is no display, this is indicative of a problem in the microprocessor control section. Refer the problem to an authorized Lexicon service technician, or return the unit to Lexicon.

7.4 Unit Powers Up, But Will Not Pass Audio

Check Cables

Check all audio cables to be sure they are securely plugged into the proper jacks. If the connections are good, check the cables themselves. Look for continuity and shorts between conductors while flexing the cable to check for intermittent contacts.

Check Other Sound Equipment in System

If all cables check out, verify the rest of the equipment in the signal chain to ensure it is indeed operating properly. Disconnect the Model 95 from the sound system and connect a straight cable between the audio inputs and outputs of the various pieces of audio gear in your system. If audio now passes through the system, the problem resides in the Model 95; if not, there is a problem elsewhere, and probably not in the Model 95.

Check Model 95 Control and Switch Settings

Be sure the MASTER output level control and any other appropriate slider(s) on the front panel are raised above the zero mark. Also, be sure the unit is not in the Infinite Repeat mode, because it could be "holding" no signal at all.

7.5 Excessive Audio Distortion or Noise

If audio passes through the system, but has unwanted excessive distortion or noise, check all levels through the system to be sure that a stage is not being inadvertently overloaded. The Output Mix section has its own overload indicator [OVLD], which lights if that stage is being clipped. If the levels are OK and the audio is still distorted, rotate the Delay Select controls fully counterclockwise. If the distortion goes away in this position, this indicates a problem in the memory. Also, verify if distortion is present on dry as well as delayed signals. Refer problem to an authorized Lexicon service technician, or return the unit to Lexicon.

7.6 Revision Number

When the unit is powered up, the initial display consists of the model number (95), revision number, and memory option (0, 1, or 2). The revision number is a service information number that informs a Lexicon service technician of the current hardware and software status of a particular unit to enable proper diagnostics and repair in case any difficulties or questions arise regarding that unit.

7.7 Returning Units for Repair

If it becomes necessary to return a Model 95 for service, bear in mind that Lexicon does not assume responsibility for units in shipment from customer to factory, whether or not they are in warranty. It is important, therefore, that shipments be well packed, properly insured, and consigned to a reliable agent, such as UPS or Federal Air Express. Be sure to include a note inside the carton explaining the nature of the problem, referencing conversations with Lexicon personnel you may have had. Also, detail the preferred return shipping method and indicate a date when the unit is needed. In addition, provide Lexicon with the name and telephone number of a person to contact if questions arise. Do not include accessories such as power cords, manuals, or remote switches.

7.8 Replacement Parts

Replacement parts and the service manual can be ordered from

Lexicon, Inc.
60 Turner Street Waltham, MA 02154 USA
Attn: Customer Service
Telephone: (617) 891-6790
TWX: 923 468

Parts will be shipped F.O.B. Waltham, MA. Charges will be those in effect at the time the order is received. Lexicon welcomes inquiries anytime during business hours for a parts quotation.

When ordering parts, refer to the appropriate parts list in the Model 95 Prime Time II Service Manual, or order by complete description and give the following information:

1. Part ID number, if available
2. Item description (e.g., RATE control knob, etc.)
3. Quantity desired
4. Revision number
5. Serial number

7.9 Limited Warranty

Lexicon, Inc., warrants each Model 95 Prime Time II to be free from defects in material and workmanship for one year, under normal use and service. This warranty begins on the date of delivery to the purchaser or his authorized agent or carrier. During the warranty period, Lexicon will repair, or replace at no charge, components that prove to be defective, provided the equipment is returned, shipping prepaid, to Lexicon's factory or designated service facility.

This warranty is null and void under any of the following conditions:

1. Abuse, neglect, alteration, or repair by unauthorized personnel.

2. Damage caused by improper use, or operation from an incorrect power source.

3. Damage caused by accident, act of God, war, or civil insurrection.

Lexicon shall not be responsible for any loss or damage, direct or consequential, resulting from Model 95 failure or the inability of the product to perform. Lexicon shall not be responsible for any damage or loss during shipment to or from the factory or its designated service facility.

This warranty is in lieu of all warranties, expressed or implied, and of any other liabilities on Lexicon's part, and Lexicon does not assume or authorize anyone to make any warranty or assume any liability not strictly in accordance with the above.

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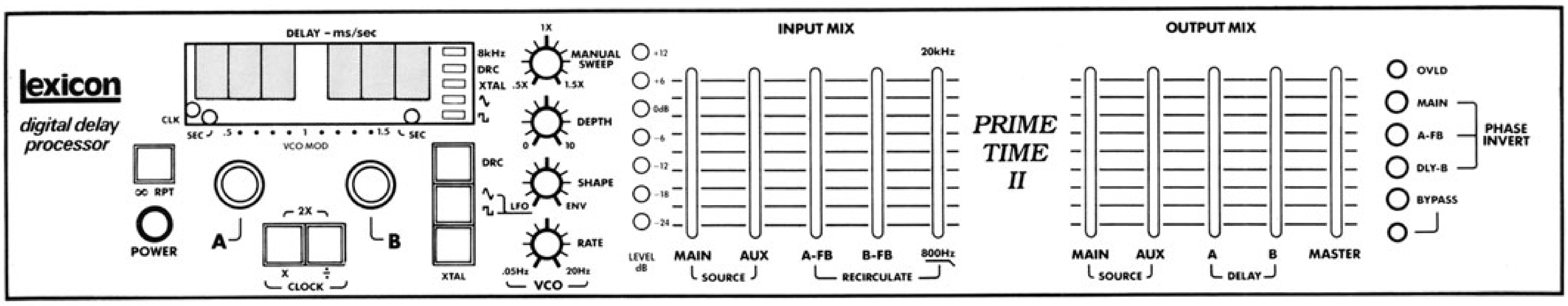

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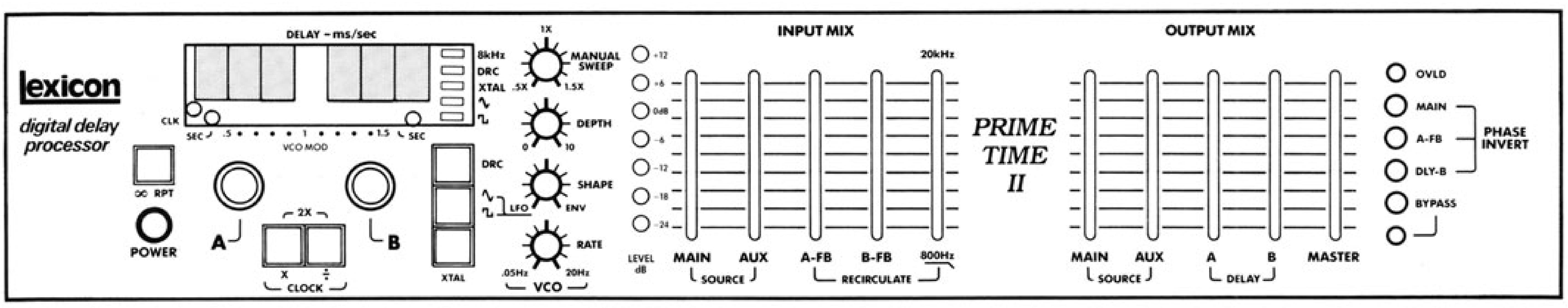

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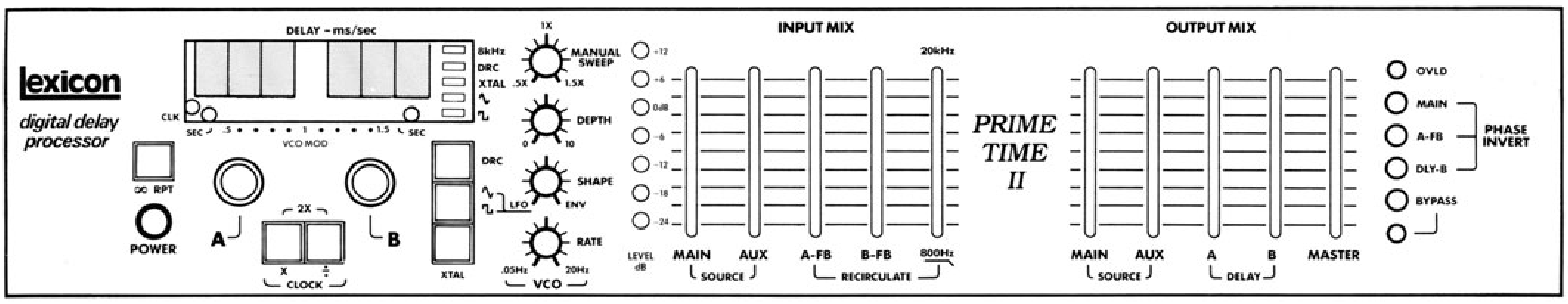

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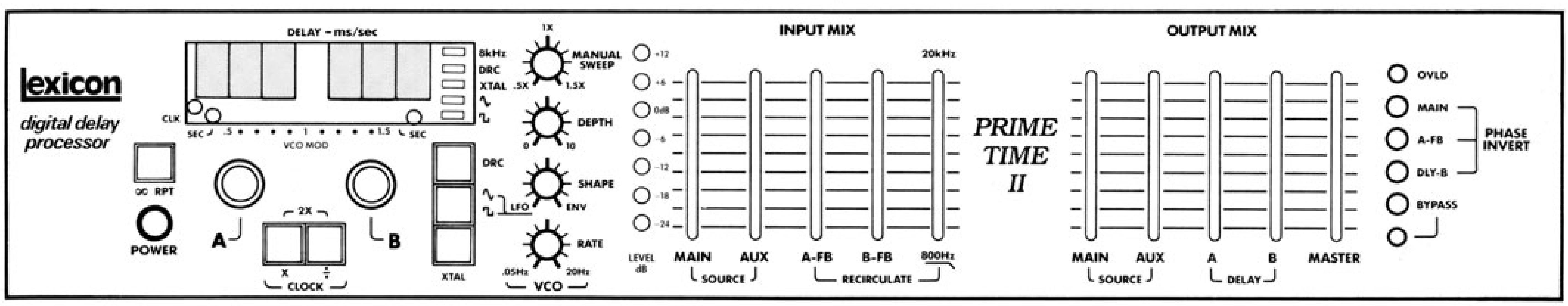

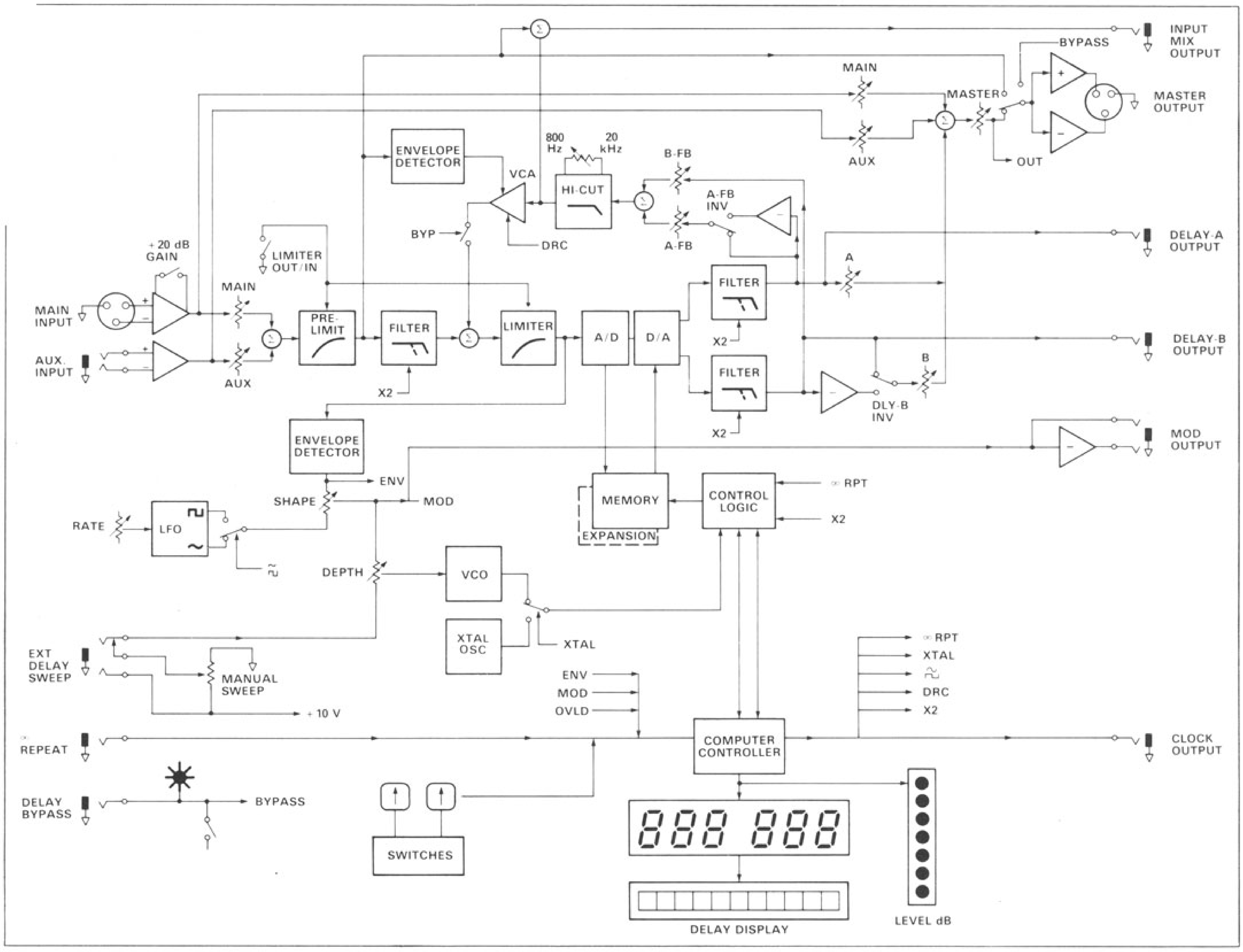

Model 95 Prime Time II Block Diagram

Accessories

Single momentary footswitch A-FS-97
Dual footswitch A-FS-41
Delay control pedal A-CP-41

Memory Options*

MEO-1 3.84-second maximum delay
MEO-2 7.68-second maximum delay

*factory installed or available as retrofit kits to be installed by qualified service personnel.