EFFECTRON II

OWNER'S MANUAL
ADM 256/ADM 1024

C.W.
#403.14

DeltaLab

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INPUT

· INPUT LEVEL: The input level control adjusts the signal level and consequently the thru put gain of the system.

At the MIN setting the sy stem gain is (x 0) to allow high level signals (up to 7.1 Vrms) to be processed.

At the MAX setting the system gain is +20dB (X 10) to allow low level signals (down to 0.1 Vrms) to be processed. The output of the unit will be amplified by a factor of 10 with respect to the input.

When the control is at the 12:00 o'clock (straight up) position, the system gain is approximately unity (X 1); i.e., the out put signal will be the same amplitude as the input signal.

· LIMIT: The Red LIMIT LED indicates that the signal level being processed is at the 0db reference point. 0db indicates the onset of clipping and/or slew limiting. There is an additional 6dB of headroom beyond what is indicated to allow for unanticipated overloading in actual usage.

· ACTIVE: The green ACTIVE LED indicates that the signal being processed is greater than -20dB below reference. The INPUT LEVEL should be set such that the green ACTIVE LED is full on for most of the time. If the input is too low the green ACTIVE LED will be off for most of the time.

FEEDBACK

· The FEEDBACK control varies the amplitude of the signal that is fed back and regenerated. Close to 100% feedback, just short of oscillation, is possible at either of the extreme settings. The regenerated signal passes through a 10 kHz low pass filter to mimimize noise buildup as it is being fed back.

(−) At this setting, the signal is fed back out of phase (negative feedback). The maximum setting is full counter clockwise; various amounts of negative feedback are possible starting from zero through the maximum position.

(+) At this setting, the signal is fed back in phase (positive feedback). The maximum setting is full clockwise; various amounts of positive feedback are possible starting from zero through the maximum position.

0 At the zero (click) center position, there is no feedback — hence, no regeneration.

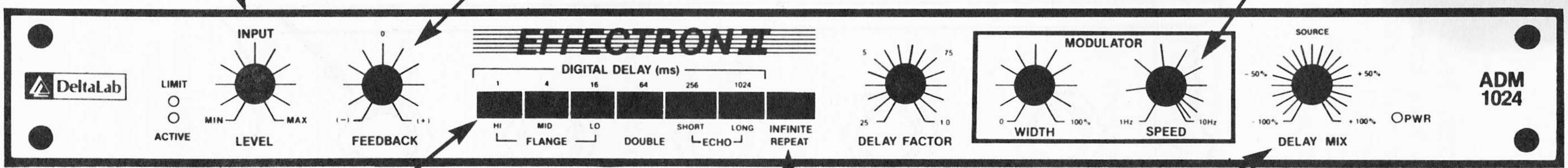
MODULATOR

The ADM 256/1024 has a precision internal VCO (voltage controlled oscillator) which is controlled by an LFO (low frequency oscillator). The LFO controls make up the MODULATOR section of the front panel. These controls interact with the DELAY FACTOR front panel control and with the (0 to +5V) CONTROL input located on the rear panel.

· WIDTH: The width control varies the sweep of the internal oscillator. In the 100% position, the delay times vary from less than (.25 X) to greater than (X 1) of the pushbutton delay setting, allowing two octaves of sweep. Should any setting less than 100% be used, the DELAY FACTOR control should be adjusted to set the desired center of the sweep. The WIDTH control should be set to the 0 position if the LFO is not being used and accurate delay times are required.

The WIDTH control also interacts with any externally applied control signal and should therefore be set at 0 for maximum control from an external source such as a foot pedal, synthesizer, envelope follower or other similar type control signal in the (0 to +5V) range.

· SPEED: The speed control varies the rate of modulation of the internal oscillator from approximately 0.1 Hz to 10 Hz.



DELAY

The delay function consists of a bank of pushbutton switches, including a full memory infinite repeat and a delay factor adjust control.

· DELAY FACTOR: The delay factor varies the basic clock frequency that determines the delay thereby acting as a manual delay adjust. The extreme settings are 0.25 and 1.0. Each line represents a change of approximately 0.05. The selected delay (via pushbutton) can be multiplied by the delay factor setting to calculate the actual delay.

· FLANGE: Three white pushbuttons are used to select short time delays used for flanging effects. The range of each button can be varied by the DELAY FACTOR or the MODULATOR controls. The ranges are as follows:

HI	0.25 ms to 1 ms
MID	1 ms to 4 ms
LO	4 ms to 16 ms

· DOUBLE: The gray pushbutton selects the delay range suitable for doubling effects. The delay range varies from 16 ms to 64 ms.

· ECHO: The black pushbutton on the ADM 256 selects delays which create echos — audible repeats. The range of delay is 64 ms to 256 ms. The ADM 1024 provides two black buttons which expand the capability of the system. The range of the ADM 1024 SHORT ECHO is 64 ms to 256 ms. The range of the LONG ECHO is 256 ms to 1024 ms.

· CAUTION: When all the delay buttons are in the OUT position, the memory is bypassed and there is virtually no delay in the output signal. If the DELAY MIX is set at -50% delay/source, the output signal will virtually be cancelled. By depressing a delay button, the signal will be restored.

INFINITE REPEAT

· INFINITE REPEAT: The red pushbutton is used to recirculate the data stored in the full memory of the unit. On the ADM 256 the full memory varies from 64 ms to 256 ms depending on the DELAY FACTOR adjust. Similarly, the ADM 1024 full memory varies from 256 ms to 1024 ms. It is possible to synchronize the start of the infinite repeat via the other delay buttons.

OUTPUT

The output level is not adjustable. The output signal level varies only with the input signal level and input level control.

· DELAY MIX: The processed signal can be mixed in or out of phase with the source via the delay mix control.

. . . At -100%, the output consists of only the delayed signal (out-of-phase).

. . . At -50%, the output consists of equal amounts of source and out-of-phase delayed signal.

. . . At SOURCE, the output consists only of the source signal.

. . . At +50%, the output consists of equal amounts of source and in-phase delayed signal.

. . . At +100%, the output consists of only the delayed signal (in-phase).

Of course, intermediate settings are also possible.

BYPASS

The bypass feature of the AMD256/1024 requires the use of a three conductor (stereo) cable. By shorting the TIP to the RING the signal flows through the cable bypassing the processing circuitry in the unit. The signal; however, still flows through the input and output circuits of the unit.

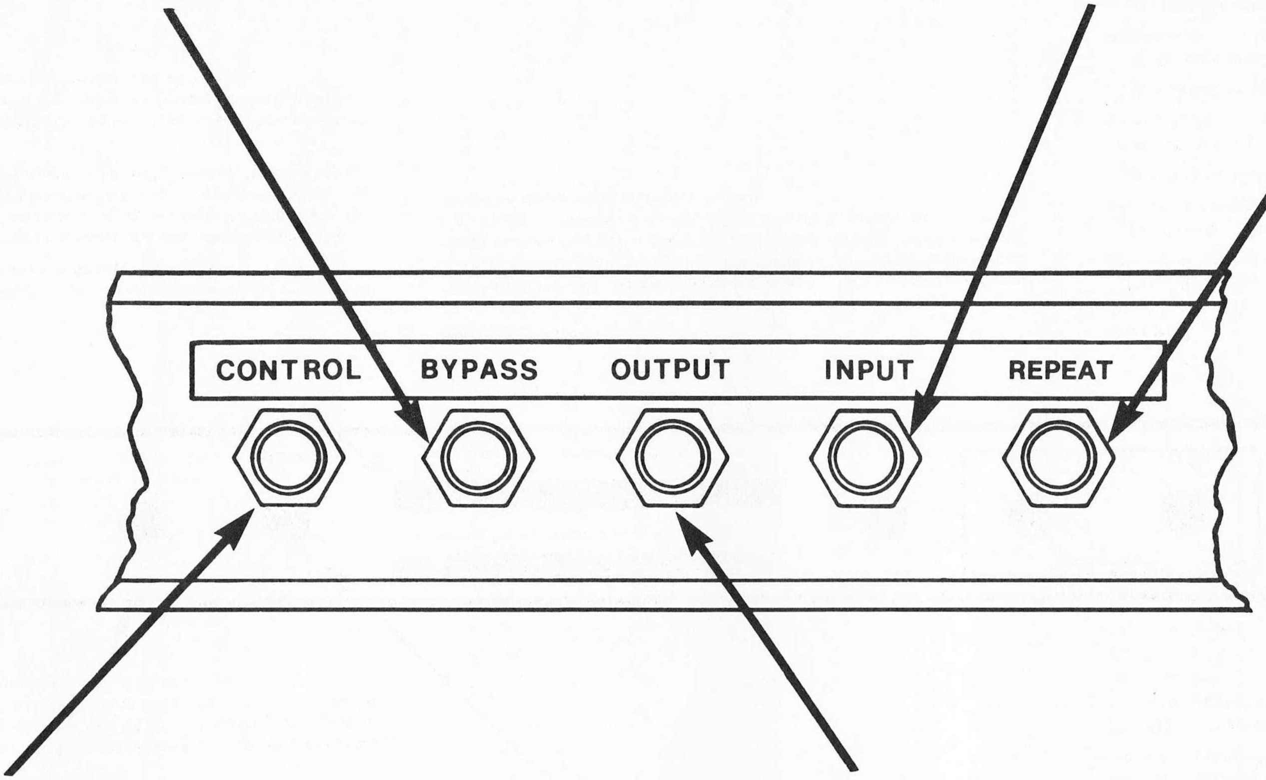
INPUT

Only one input is provided on the ADM 256/1024. This input accepts signals as low as 0.1 Vrms and as high as 2.0 Vrms for full dynamic range. The LEVEL control on the front panel is used to set the proper level.

REPEAT

The repeat jack allows remote control of the infinite repeat function.

*ADM 1024 only



CONTROL

The control jack is used to modulate the internal VCO. A signal generator, envelope follower, synthesizer or foot pedal can be used for this purpose. The voltage range is 0 to +5 volts where 0 volts will yield a delay factor of x 1 and +5 volts yields a delay factor of X 0.25.

NOTE: When the control jack is used, the DELAY FACTOR control on the front panel is totally disabled; however, the WIDTH and SPEED controls will interact with the external input. To prevent this interaction, simply set the WIDTH control to its 0 setting. This allows the external control to function independently.

OUTPUT

The output consists of both processed and unprocessed signal depending on the setting of the DELAY MIX control on the front panel. At full dynamic range the signal level will be 2.0 Vrms nominal. Also, a synthesized stereo output is present on the "ring" portion of the output jack.

INTRODUCTION

The EFFECTRON[tm] II is an updated version of the original EFFECTRON. It is a studio quality, special effects processor designed for the performing musician. The updates include suggestions received from the many satisfied owners of EFFECTRONS...improved input stage, stereo outputs and even a remote Infinite Repeat jack on the ADM 1024. You can rest assured that your EFFECTRON II will become one of the most important components in your total sound system.

CIRCUIT DESCRIPTION

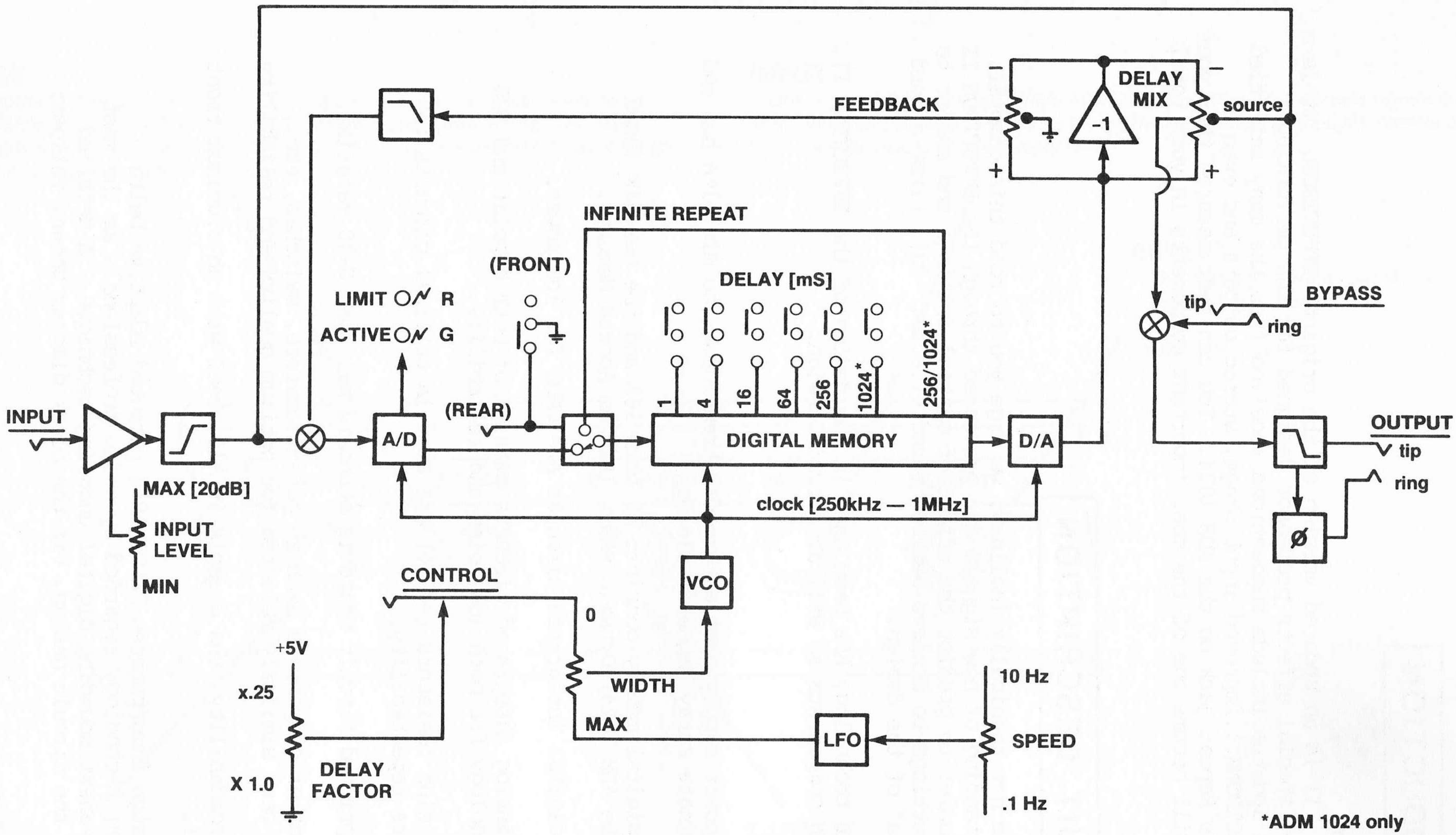
Even if you are not tecnically inclined, we urge you to read this chapter. A basic understanding of how signals are processed through the EFFECTRON II will make it easier to predict the effect of control settings and easier to plan control settings to achieve desired sound effects. But, first a word about the "guts" of the design.

Only the latest technology has been used in the design of the EFFECTRON II. This applies to components as well as circuit design.

For example:

1. The power supply uses a toroidal transformer to minimize hum and eliminate stray magnetic fields.
2. The digital memory consists of both 16K and the new 64K (used in the ADM 1024) Dynamic RAM's (Random Access Memory).
3. All digital integrated circuits are CMOS for low power.
4. All analog integrated circuits consist of both bipolar and MOS technology for both low noise and reliability.
5. Precision resistors (+/-1%) are used in critical circuits to insure repeatability.
6. The printed circuit board is standard mil spec G-10 material.
7. All other components such as potentiometers, switches, etc., have been similarly selected for optimum quality and reliability.
8. For roadability, the chassis is all steel with an aluminum front panel.

The circuit design incorporates DELTALAB's patented Adaptive Delta Modulation (ADM) technology regarded by audio professionals as the most natural and cleanest sounding digital encoding technique. A detailed explanation of the circuit design, via the block diagram shown, follows:



EFFECTRON™ II Block Diagram ADM 256 / 1024

The input audio signal is fed into the EFFECTRON II via the INPUT phone jack located on the rear panel. This signal is routed through an input amplifier stage to properly set the operating signal level via the INPUT LEVEL control located on the front panel. The INPUT LEVEL control also sets the gain of the total system (there is no compensating output level control). The minimum input voltage is 0.1 Vrms.

The correct operating level is set by observing the green ACTIVE monitor and the red LIMIT monitor located on the front panel. These LED's monitor the level as seen by the encoding circuits. For maximum dynamic range, the active LED should be full-on with a rare flashing of the LIMIT LED during peak passages in the audio input.

Once the proper input level has been set, the audio signal is directed to a pre-emphasis circuit to pre-condition the signal before the actual Analog-to-Digital conversion takes place.

After pre-emphasis, the pre-conditioned signal is now sent to the A/D encoder to be converted to a digital signal. It is here that DELTALAB holds all the aces. The Analog-to-Digital encoding scheme is unique. A carefully designed Adaptive Delta Modulator converts the audio signal by analyzing both the value and slew rate of the signal. The result is the most accurate digital representation of the audio signal possible over the full dynamic range.

Next, the digital signal representing the audio is stored in the digital memory. Access to this memory is via the pushbutton switches located on the front panel. Upon selection of a given delay, as indicated by the pushbutton, the digital signal is extracted from the memory and sent to the Digital-to-Analog decoder. Note that the A/D, the memory and the D/A are synchronized to a common clock to insure the proper addressing of the memory banks and to provide the sampling rate for the A/D encoder and D/A decoder.

The D/A decoder is the perfect compliment of the Adaptive Delta Modulation encoder. At the D/A output, the signal is sent to an inverting amplifier and further directed to both the FEEDBACK and DELAY MIX controls located on the front panel.

The DELAY MIX control is a center tapped potentiometer, i.e., the unprocessed input signal is applied at the center tap and each phase of the delayed signal is applied at the extreme terminals. As such, a single control is all that is necessary to mix the processed signal (either phase) with the input (source) signal.

The final mix is then de-emphasized to restore the audio to its proper levels. The output is available at the OUTPUT phone jack on the rear panel.

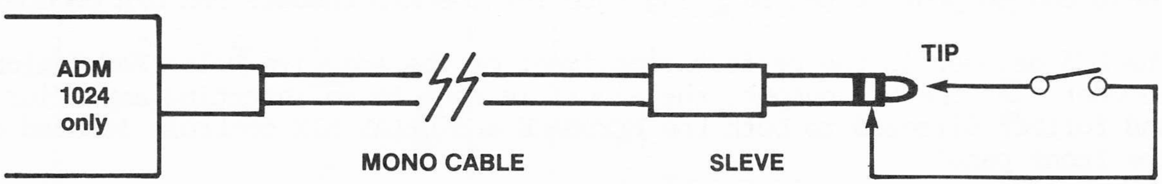
The final audio signal available at the OUTPUT jack is further processed by a patented phase shift circuit to simulate stereo output. This circuit keeps the low frequencies in phase while providing a controlled amount of phase shift for mid and high frequencies. The result is a very realistic form of simulated stereo signal. The simulated stereo output is available by using both the tip and ring of the OUTPUT jack.

Meanwhile, back at the FEEDBACK control, the processed audio (delayed signal) is fed back to a summing point and mixed with the incoming signals at the A/D converter input. The FEEDBACK control, like the DELAY MIX control is a center tapped potentiometer. The center tap is grounded so that there will be no feedback when the control is on the center position. Again, like the DELAY MIX control, both phases of the processed audio are available at each extreme terminal. Therefore, only one FEEDBACK control is used for both positive and negative feedback.

Before being mixed with the pre-conditioned input, the feedback signal is filtered via a 12kHz low pass filter to create a more natural and pleasant sounding echo repeat as well as to keep the noise level from building up when maximum feedback is used.

A digital feedback path exists from the memory output to the memory input constituting an infinite repeat cycle. By depressing the red INFINITE REPEAT push button to the "in" position, the digital memory is recirculated indefinitely. The resulting audio out is an echo that never dies out. In the ADM 256, this echo is 256 msec (1/4 of a second) at the maximum setting of the DELAY FACTOR control: This same echo is over a full second (1024 msec) on the ADM 1024. To defeat the repeat function, simply push the red INFINITE REPEAT button so that is is returned to the "out" position.

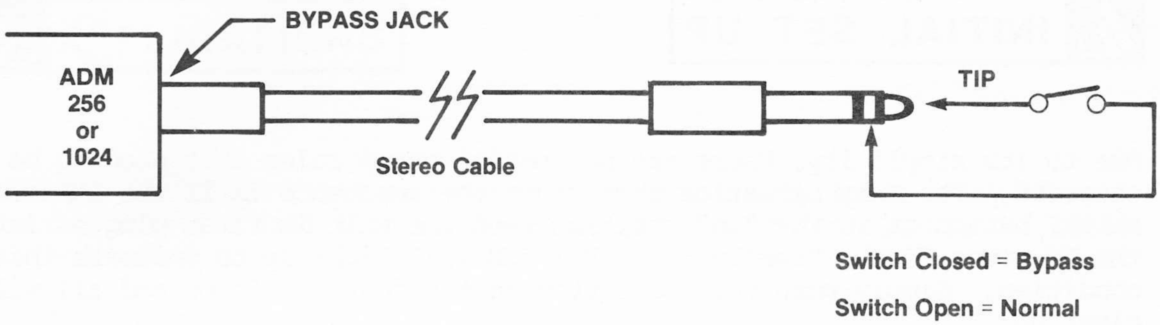
When the INFINITE REPEAT button is in the "out" position, you have the option of using the rear panel INFINITE REPEAT jack (ADM 1024 only). By using a mono phone plug, you will now be able to activate the infinite repeat function via a foot switch. As shown below, by shorting the tip to the sleeve, the digital memory will be recirculated indefinitely. To defeat this function, you simply disconnect the tip from the sleeve.



Switch Close = Infinite Repeat

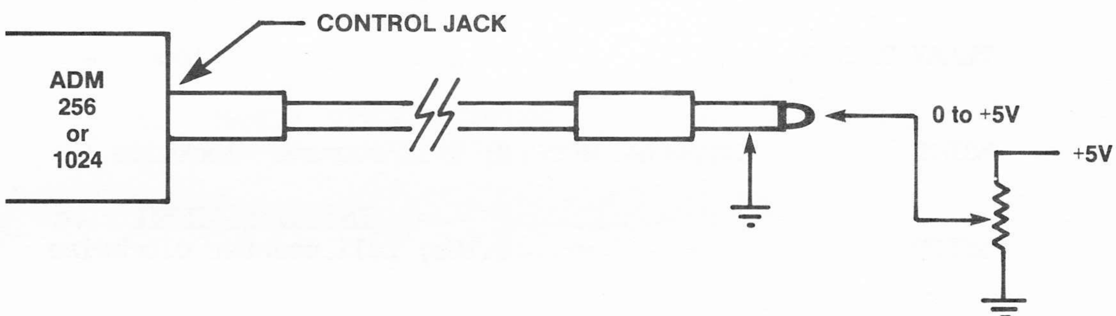
Switch Open = Normal

The last possible signal path is the BYPASS which requres the use of a stereo (three conductor cable) phone plug to be connected to the BYPASS jack located on the rear panel. By shorting the TIP to the RING, the pre-conditioned audio is directly routed through the cable and back to the de-emphasizing circuitry in the unit to bypass all digital and feedback signal paths. A simple scheme for doing this is shown on the next page.



The following functions are not in the signal path, but are used to control the basic bit rate clock to create special effects other than those that result by simple feedback. There are three front panel controls whose function is to provide a control voltage into the precision Voltage Controlled Oscillator (VCO) used as the digital bit rate and sampling clock. For example, a control voltage of 0 to 5 volts will vary the clock for 250 kHz to 1 MHz which results in a four to one (two octave) range.

Setting the WIDTH control, located on the front panel, in the 0 position, the DELAY FACTOR control (also located on the front panel) provides the required voltage as shown below in the block diagram. When an external signal is applied to the CONTROL jack located on the rear panel, the DELAY FACTOR is disabled.



The WIDTH control is used in conjunction with the SPEED control located on the front panel to apply a Low Frequency Oscillating (LFO) voltage to the VCO input. The LFO is a sinusoid whose frequency is controlled by the SPEED control. The WIDTH control determines the amplitude fed into the VCO. It can be seen that intermediate settings of the WIDTH control will cause a mix of both the DELAY FACTOR setting and the LFO output. Also, when a voltage is applied to the external CONTROL jack, the WIDTH will likewise mix internal LFO settings with the external control voltage -- all of which makes for interesting effects. Note that the WIDTH control provides a greater than 4-to-1 sweep range.

The EFFECTRON II is basically a simple to understand device, but because of its simplicity it is a very powerful effects tool.

INITIAL SET UP

Due to its simplicity, there are no special setup rules that need to be observed. The only situation that can cause annoyance is if the INFINITE REPEAT button is in the "in" position when the unit is first plugged into the AC power line. Usually the LIMIT LED will light up to indicate this condition. Simply push the RED button to the "out" position and all will clear up.

We do; however, recommend the following be used as a starting point until you familiarize yourself with each control:

INPUT LEVEL	-	MIN: full counter clockwise
FEEDBACK	-	0; center position
DELAY (BUTTONS)	-	Start with GRAY button "in"
INFINITE REPEAT	-	Button "out"
DELAY FACTOR	-	X1 position; full clockwise
WIDTH	-	0; full counter clockwise
SPEED	-	0.1Hz; full counter clockwise
DELAY MIX	-	SOURCE; Center position

The INPUT LEVEL can now be adjusted for proper level via the LED monitors. All other functions can be varied at will to create the various effects.

NOTE: IN ORDER TO INSURE RELIABILITY AND LONG LIFE BY PREVENTING FAILURES DUE TO UNNECESSARY THERMAL SHOCK, I.E., (HOT AND COLD VARIATIONS), THE EFFECTRON II DOES NOT HAVE AN OFF/ON SWITCH. THE AVERAGE POWER CONSUMPTION IS LESS THAN 5 WATTS; THIS SHOULD NOT AFFECT THE OPERATING ECONOMY OF YOUR TOTAL SOUND SYSTEM.

APPLICATIONS

While the EFFECTRON II is capable of providing a very large array of effects, they all fall into the basic categories (or combination of these) outlined below:

1. STRAIGHT DELAY

   a. DISCRETE ECHOS, SLAPBACK
   b. DOUBLING, THICKENING, BROADENING
   c. PRE-REVERB DELAY
   d. HAAS-EFFECT IMAGE SHIFTING

2. FEEDBACK OF SHORT DELAYS

   a. COMB FILTERING
   b. FLANGING
   c. FLANGING WITH FEEDBACK
   d. FEEDBACK PHASE INVERSION
   e. TUNED RESONANCE

3. FEEDBACK OF LONG DELAYS

   a. MULTIPLE ECHOS

4. TIME BASE MODULATION

   a. MANUAL PITCH SHIFTING
   b. VIBRATO, AUTOMATIC PITCH SWEEPING

5. INFINITE REPEAT

1. STRAIGHT DELAY

Sound travels in air at a finite speed — approximately 340 meters per second (1100 feet per second), or about one foot per millisecond -- and many of the uses of a digital delay processor involve the controlled electronic recreation of effects which occur in acoustics due to this finite speed.

For instance, for practical reasons, vocals and instrumentals are usually recorded in a nearly anechoic fashion by close-miking in an acoustically absorptive studio, but this sometimes yields an anemic, uninteresting sound. Whenever we hear vocal instrumental sounds in a living room or concert environment, the dry sound is accompanied by reflections off nearby walls, floor and furniture. These early reflections accompanying the direct sound — slightly delayed because of their longer airpath — add apparent volume and fullness and thus enrich the character of the sound.

The subjective effect of reflections (delayed replicas of an original sound) depends on the length of the delay as follows:

— Single or multiple delays up to about 40 milliseconds after the direct sound alters the apparent character of quality, but are not perceived separately. Typically, they add the sort of "fullness" and body which a voice has in a living room, but lacks when heard outdoors. They are called "early" reflections.

— A single delay within about 40 milliseconds and having the same volume level as the direct sound, produces an effect something like that heard when a solo voice is replaced by two identical voices singing in unison. This is called "doubling".

— A single delay longer than 40 to 50 milliseconds starts to break away from the original sound and be perceived as an echo.

— A longer delay, i.e., over 100 mS and substantially lower in level than the original sound, is heard as a "slapback" echo like that from the rear wall of a cathedral or other large space. Of course, to be acoustically authentic, any delay must be lower in level than the direct sound. A delay which is substantially louder than the dry sound will be perceived as the original, and the original sound will appear to be a false pre-echo.

— Repeated delays at intervals greater than 50 milliseconds are perceived simply as multiple echos. If the pattern of multiple delays becomes more complex with dozens or hundreds of echos per second in a pattern which fades progressively away into inaudibility, then the echos are perceived as a single continuous sound — reverberation. Acoustically authentic reverberation includes some "early" reflections beyond 100 mS becoming progressively closer in spacing as they fade away. If the reflections are spaced at unfiform intervals in time (e.g., a simple string of echos 40 mS apart), the reverberation of transient sounds acquires a chattering quality known as "flutter echo", or "hard reverb".

a. DISCRETE ECHOS, SLAPBACK

Feed a signal into the EFFECTRON II's input, push the echo button and then increase the DELAY FACTOR until the delay time is long enough to be perceived as a discrete echo. While this echo can be mixed directly with the dry sound, a more interesting result is usually obtained by panning the echo elsewhere in the stereo image with a stereo mixer; i.e., place the dry sound on the left and the echo opposite it on the right. The echo usually should be a bit lower in level than the dry source.

This left-right echo bouncing effect is particularly useful with a regular drum beat or a two-note guitar figure that is used to set the beat. By varying the DELAY FACTOR, you may be able to synchronize the echo interval to match the rhythm of the music so that the source and its echo fall on alternate beats.

If the delay is 10 to 20 dB lower in level than the source, it will be perceived as an echo -- especially if it is placed in the opposite channel. If the delay is reproduced at the same level as the source, it will be identified as a repeat rather than as an echo such as from a distant wall or canyon.

b. DOUBLING, THICKENING, BROADENING

Two voices singing together, or a single voice which is double-tracked (overdubbed to accompany its previously recorded self) produce a combined sound which is richer and more interesting than simply turning up a single vocal track 3 dB in level. One reason is that the two separate voices are never recorded in exactly precise synchronism. The waveform of one is usually a few milliseconds ahead or behind the other despite the most careful of rehearsals.

This effect can be simulated with any single source simply by delaying it by 16 to 64 milliseconds and mixing the delayed signal with the original at equal levels. This is called doubling. It thickens the texture producing a more "full-bodied" sound while it increases the apparent loudness without significantly raising VU meter levels. It is particularly useful for adding strength and character to a thin-sounding vocal.

You should experiment with the legnth of the delays used for doubling. With delays in the 16-20 mS range, the sound remains relatively tight and focused. Delays of 30-40 mS produce a more obvious broadening. Doubling with delays shorter than about 16 mS is not recommended because of the risk of coloration due to comb filtering. If you extend the delay beyond about 40 mS it may be heard as a distinct echo.

This process simulates two vocal tracks with a small, but constant delay between them. Of course, when real voices are recorded or overdubbed, they have varying small differences between them. So, to make its doubling seem more realistic, use the MODULATOR to continuously vary the delay. The action must be subtle so as to avoid audible pitch wobble. Setting the width and speed controls both to between 7:00 and 9:00 o'clock, usually produces a pleasingly realistic doubling effect with no audible side effects.

c. PRE-REVERB DELAY

Authentic acoustic reverberation in a large space requires many dozens of milliseconds to buildup. But, in many studio reverb units (spring, plate or acoustic chamber) an output signal begins to appear very rapidly after the onset of the input signal. The subjective performance of reverberators is usually improved by delaying the signal fed to them. Both the ADM 256 and the ADM 1024 can provide this delay. The delays used for doubling often serve as pre-reverb delays.

In a concert environment, the listener hears the direct sound from the performer, followed by "early" reflections from the floor and walls of the stage and finally the reverberation comprised of long delays as sound is reflected among the walls of the auditorium. When using a reverb system to process dry studio sound, pre-reverb delay can also serve as the "early" stage-area reflections, simply by including them in the mix.

d. HAAS-EFFECT IMAGE SHIFTING

The ear has an integration time of about 40 mS. If a sound is heard from one direction and the same sound also arrives at a later time from another direction, the second arrival will not be heard at all if the interval between the two arrivals is less than 40 mS. The later arriving sound may even be several dB higher in level than its earlier counterpart; nevertheless, the ear will hear both sounds as a single louder sound located in the direction of the first arrival. This is the precedence effect -- or Haas effect -- and it can be used to stabilize images in a stereo sound field.

For example, if a signal is recorded in both channels at identical levels, the feed to the right channel is delayed, then the sound will be perceived as originating exclusively in the left channel. In principle, the delay to the opposite channel may be anything from 1 mS to 40 mS. But, long delays carry the risk that some listeners may begin to perceive the delayed sounds as an echo; and if the delay is under 10-15 mS coloration due to comb filtering could become a problem if the two channels are later mixed together (for AM broadcast, for instance). A delay of around 20 mS usually turns out to be optimum for Haas-effect image panning.

It is possible to keep the signal level constant in both channels and cause the image to jump back and forth from left to right by swapping the delayed and undelayed signals by panning the mixer.

2. <u>FEEDBACK OF SHORT DELAYS</u>

By using the feedback control, the delayed signal is fed back and mixed with the incoming audio signal and then the composite signal is encoded and read into the digital delay circuits. The strength and tonal quality of the feedback may be adjusted by the user. The maximum feedback gain is less than unity; thus, like a real sound reflecting off any surface, the recycled signal is at least a little weaker than the original, and as the sound is repetitively recycled through the system, it gradually fades away.

a. COMB FILTERING

Whenever any original sound and a delayed version of itself are mixed together, the two sounds are mutually reinforced at some frequencies and tend to mutually cancel at other frequencies. This occurs in an electronic delay line and it also is a common acoustic phenomenon both in recording (where the mike picks up both the direct sound and a reflected sound from the floor, wall or music stand) and in playback (where a loudspeaker's direct sound combines in the air with reflections off room boundaries and furnishings). This pattern of alternating reinforcement and cancellation causes an audible coloration of the sound.

Two things are noteworthy. One is that the reinforcements and cancellations occur at harmonically-related frequencies. The other is that the pattern of peaks and dips can be varied, i.e., tuned, just by varying the delay time. In almost all normal vocal and musical sounds, most of the energy is found at fundamental frequencies and their harmonic overtones. By varying the delay time of a delay-and-mix circuit, we can easily but dramatically alter the overtone structure and the tonal quality of any steady sound. The precise delay times and unusually flat frequency response of the EFFECTRON II make it ideal for comb filtering on demand.

b. FLANGING

If the unit is setup to produce comb filtering and then the delay time is smoothly varied, the pattern of peaks and nulls will shift in frequency. Modulate the delay time rapidly and the pattern of peaks and nulls will sweep rapidly up and down the frequency spectrum passing in and out of synchronization with the frequencies of musical signals and their overtones. Select relatively short delays so that the spacing of the nulls corresponds to that of musical harmonics. As the delay is swept, at one moment, the odd-numbered harmonics are enhanced by 6 dB — a moment later, the harmonic structure of that note is sliding into alignment with the harmonic overtones of a different note. This sweeping, shifting change is called "flanging".

To achieve it, one need only setup the delay and mixing as required for comb filtering and then activate the modulator to sweep the delay time up and down. Typically the modulator WIDTH is set at maximum in order to produce a broad sweep and the modulator SPEED is set at a modest value (e.g., between 7:00 and 9:00 o'clock) so that the comb filter sweeps up and down every few seconds. A high setting of the modulator SPEED would sweep the notch pattern over the musical spectrum too rapidly for the changing harmonic structure of the music to be heard and might also cause audible pitch wobble of the musical signal itself. You should experiment to find the settings of DELAY, WIDTH, SPEED and mixing ratio which yield interesting flanging effects with various vocal and instrumental sounds. In general, higher-pitched sounds work best with shorter delays.

c. FLANGING WITH FEEDBACK

Mixing the dry and delayed signals at approximately the same level causes comb filtering (a pattern of nulls and peaks in frequency response). Modulation of the delay time causes the pattern of nulls and peaks to sweep through the musical spectrum producing "flanging".

A stronger, richer flange is reproduced by using maximum feedback to create a chain of closely-spaced delays whose uniform spacing sharpens and strengthens the comb filter peak and nulls. Select short delays such as 1 or 4 mS and set the FEEDBACK near the maximum setting, either positive or negative. Finally, modulate the delay by setting modulator WIDTH at maximum and SPEED between 7:00 o'clock (for a slow flange) and 12:00 o'clock (for a fast flange). Note that is is important to use near maximum feedback so that each successive recycled delay is at nearly the same strength as its predecessor, yielding the strongest reinforcement/cancellation pattern. The flange can be varied, of course, by selecting different front panel delays. If the WIDTH control is set less than maximum, the DELAY FACTOR can be used to alter the delays and thus shift the frequencies of the comb-filter pattern. Finally, the flanged signal must be mixed with the dry signal via the delay mix control.

d. FEEDBACK PHASE INVERSION

The feedback phase control gives you the option of mixing the feedback delays either in phase (+) or out of phase (-) with the incoming dry signals. This choice alters the frequency distribution of any comb-filtering coloration which may occur as a result of combining dry and delayed signals. You can make this choice by ear. When using feedback, simply rotate the feedback control to find out whether positive or negative feedback sounds better in each particular situation.

e. TUNED RESONANCE

Select medium-length delays (between approximately 2 and 32 milliseconds) to produce comb-filtering and then use maximum feedback to create a string of closely spaced delays whose uniform spacing sharpens and strengthens the peaks and nulls of the comb-filter pattern. As a result, some of the peaks will be fed back at almost sufficient strength to cause a sustained feedback oscillation. Whenever the system is stimulated by a broadband input signal such as drums or a speaking voice, it will tend to "ring" at frequencies of the comb-filter peak. Since the peaks in a comb-filter form a harmonic series, a distinctly musical tonality will be produced. By varying the delays, the pitch of the tuned resonance can be tuned as desired. Inverting the polarity of the feedback will also alter the pitch of the resonance. For best results, the FEEDBACK control will normally have to be at a maximum setting.

A principal application of tuned resonance is the processing of human speech to produce "computer speech" by adding a fixed-pitch, metallic resonance. Such resonances can also be used to color various percussive sounds in interesting ways.

The pitch of the tuned resonance may be varied dynamically by means of the VCO — typically by setting the WIDTH close to maximum SPEED between 9:00 and 12:00 o'clock. An unpitched broadband sound such as a repeating drumbeat can be made to sound rather like a guitar.

3. FEEDBACK OF LONG DELAYS

In terms of control operation, the feedback of long delays is essentially the same as that for short delays. The feedback of long delays is primarily for the creation of strings of echos.

a. MULTIPLE ECHOS

To create a string of echos, setup a long delay (over 40 mS) to produce a discrete echo and then use feedback to recycle the signal repetitively through the delay. In general, echos are more interesting if they are separated spatially from the dry source and from each other by panning at the mixing console.

4. TIME BASE MODULATION

Much of the EFFECTRON II's flexibility and usefulness as a studio or onstage tool arises from its ability to vary the speed of the "clock" which governs how rapidly signals are shuttled through the digital memory, under either manual or automatic control.

The effects obtainable by varying the clock are easily understood by analogy with a tape recorder having variable tape speed whose recording and playback heads are separated by some distance. The time-delay between recording and playback is governed by the separation of the heads and by the tape speed. If the speed is doubled, the tape will traverse the distance in half the time. As long as the tape speed remains constant while the tape is being recorded and played, then the tape speed will affect only the delay. But if the tape is recorded at 7 1/2 ips and then the speed is doubled before the tape arrives at the playback head, the waveforms will pass the playback head twice as rapidly as they passed the recording head and the frequency of each sound will be doubled, i.e., the musical pitch will rise an octave. Thus, a change in tape speed which occurs in the interval between recording and playback alters not only the delay time, but also the pitch — delay modulation causes pitch modulation.

a. MANUAL PITCH SHIFTING

The DELAY FACTOR control adjusts the clock speed over a 4-to-1 range and consequently is capable of varying the pitch of a musical tone over a 4-to-1 range in frequency — or two full octaves in pitch. This is easily demonstrated with the aid of the infinite repeat mode. Set the DELAY FACTOR control to X1, set delay maximum and monitor the delayed output. Play "middle C", for example, and while the note is sounding, push the repeat button to store a fragment of the note in memory. Then, vary the DELAY FACTOR from X1 down to X.25 and as you do, the pitch of the recirculating sound will rise by approximately two octaves and it will fall as the DELAY FACTOR is moved back toward X1.

It is not necessary to recirculate a signal to manipulate its pitch. With the repeat button OFF, any signal passing through will change in pitch as the DELAY FACTOR is changed. As noted earlier, the pitch shift depends on the change in clock speed. The faster the DELAY FACTOR knob is turned, the greater the pitch shift will be and the longer the delay time is, the easier it will be to alter the clock speed significantly during the delay. This pitch is shifted downward as the DELAY FACTOR is increased and vise versa.

Gradual pitch changes are sometimes referred to as Doppler shifts, an analogy to the Doppler effect which occurs with moving sound sources such as train and auto horns (the pitch rises when a sound source is moving toward the listener and falls as the source recedes).

b. VIBRATO, AUTOMATIC PITCH SWEEPING

It is difficult to accomplish really smooth pitch changes by manually rotating the DELAY FACTOR control so the modulator function provides an automatic method of achieving smooth pitch modulation effects. The LFO modulates the internal clock up and down by an amount set by the WIDTH control and at a rate set by the SPEED control.

As with manual pitch shifting, the amount of modulator-actuated frequency shift will depend on the choice of delay time as well as on the modulator controls. With delay of a few milliseconds and with the WIDTH and SPEED controls set at about 10:00 o'clock, a subtle pitch modulation is produced which most listeners can just detect. Lengthening the delay or increasing the WIDTH yields a larger and more obvious sweeping of the pitch up and down. Increasing the SPEED beyond 12:00 o'clock, not only sweeps the pitch up and down more rapidly, but also sweeps it over a wider frequency range — up to several octaves.

Vibrato is a rapid, low-amplitude pitch modulation which is attractive with most vocals and with some instrumental sounds. It must be used with caution lest it sound ludicrous — excessively wide vibrato will simply be perceived as an off-key pitch wobble. Typical control settings for producing an authentic-sounding and attractive vibrato are: DELAY 32 mS, WIDTH between 7:00 and 9:00 o'clock, SPEED between 1:00 and 3:00 o'clock.

5. INFINITE REPEAT

With digital feedback we can sustain a repeating echo perpetually. Simply push the red button and a small portion of the input signal will be stored in the full memory as long as the red button is in the "in" position. This function can also be operated via the rear panel INFINITE REPEAT jack. See Page 6 for diagram.

SPECIFICATIONS

<u>DELAY RANGE</u>	
ADM 256	0.25 to 256 mS
ADM 1024	0.25 to 1024 mS
<u>FREQUENCY RESPONSE</u>	
+1, -3dB @ -10dB below LIMIT	20 to 16k Hz @ all delay settings
<u>DYNAMIC RANGE</u>	
A - weighted	90 dB typ., 85 dB min
<u>DISTORTION</u>	
Ref 1kHz @ LIMIT	0.2% max
<u>INPUT RANGE</u>	-26 to + 17dBV (0.05 to 7.1Vrms)
<u>INPUT LEVEL GAIN</u>	
Min	---- (X0.0 gain)
Unity @ (center position)	0 dB (X1.0 gain)
Max	+26 dB (X20 gain)
<u>OUTPUT LEVEL (@ LIMIT)</u>	+ 6dBV (2.0Vrms)
<u>METERING</u>	
LIMIT	0 dB ref
ACTIVE	-20 dB below ref
<u>MODULATION</u>	
WIDTH	0 to max depth (5-to-1)
SPEED	0.1 Hz to 10 Hz nominal
CONTROL (External)	0 to +5V (X1 to X0.25)
<u>REPEAT</u>	Repeats signal in full memory independent of delay setting. Varies with DELAY FACTOR.
<u>FEEDBACK</u>	Recirculates delayed signal to create multiple repeats through 12 kHz Low Pass Filter.
<u>BYPASS</u>	
Front Panel (Internal)	Bypasses memory when all delay buttons are in out position.
Rear Panel (External)	Shorted - bypasses total system; open - normal operation.
<u>SIZE</u>	1 3/4x19x7in (4.45x48.3x17.8cm)
<u>SHIPPING WEIGHT</u>	10 lbs

Manufacturer reserves the right to make improvements without notice or obligation; therefore, all specifications are subject to change.

IN CASE OF DIFFICULTY

In many instances where difficulty is experienced, it is best to check out a few simple questions before "panic" sets in:

1. Is the unit plugged into the power line? The power (yellow LED) monitor on the front panel should indicate this.

2. Are all inputs, outputs and control signals connected to their respective jacks on the rear panel?

3. Is the INFINITE REPEAT button in the out position? Remember, the digital memory is disabled from the input when INFINITE REPEAT is on ("in" position). Also check the INFINITE REPEAT jack in the rear.

4. If you are using an external bypass switch, have you placed the unit in the bypass mode? This would totally prevent any kind of signal processing.

5. Are the input signal levels in the proper range? The INPUT LEVEL control and the ACTIVE/LIMIT LED's will properly set the input for optimum dynamic range. (Low noise, low distortion, etc.)

6. If this is the first use of the EFFECTRON II, have you referred to the initial setup as suggested in the Owner's Manual?

Every effort has been made to insure trouble-free performance from each EFFECTRON II. Should a problem occur, simply call collect and ask for Bruce Wayne at (617)256-9034. Should this be impractical, notify us by writing to:

DELTALAB RESEARCH, INC
ATTN: Customer Service
19 Alpha Road
Chelmsford, MA 01824

Describe the nature of the problem, the steps you have taken to diagnose it, serial number of the unit and whether or not you have retained the original shipping carton.

NOTE: If the unit must be returned to our factory, we will provide you with a Return Authorization (RA) Number which must be prominently displayed on the outside of the shipping carton. Any unit which arrives without a visible RA Number may be refused by our Receiving Department.

LIMITED WARRANTY

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DELTALAB RESEARCH, INC., ("DELTALAB") warrants to the first purchaser of a new DELTALAB EFFECTRON II that the unit is free from defects in material and workmanship. DELTALAB's sole obligation under this warranty shall be to provide, without charge, parts and labor necessary to remedy defects, if any, which appear within ninety (90) days from the date of purchase.

This warranty is the sole and exclusive express warranty given with respect to the unit and all other express warranties are hereby excluded. IMPLIED WARRANTIES, INCLUDING THOSE OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSES, ARE LIMITED TO NINETY (90) DAYS FROM THE DATE OF PURCHASE. SOME STATES DO NOT ALLOW LIMITATIONS ON HOW LONG AN IMPLIED WARRANTY LASTS, SO THE ABOVE LIMITATIONS MAY NOT APPLY TO YOU. DELTALAB IS NOT RESPONSIBLE FOR INDIRECT, INCIDENTAL OR CONSEQUENTIAL DAMAGES. SOME STATES DO NOT ALLOW THE EXCLUSION OR LIMITATION OF INCIDENTAL OR CONSEQUENTIAL DAMAGES, SO THE ABOVE LIMITATION OR EXCLUSION MAY NOT APPLY TO YOU.

This warranty does not apply if the unit has been:

(1) Repaired, worked on or altered by persons unauthorized by DELTALAB in such a manner as to injure, in DELTALAB's sole judgment, the performance, stability or reliability of the unit;

(2) Subjected to misuse, negligence or accident; or

(3) Connected, installed, adjusted or used otherwise than in accordance with the instructions furnished by DELTALAB.

This warranty is valid only when the unit is returned to DELTALAB within ninety (90) days from the date of purchase, two-way freight prepaid, together with a copy of the original invoice from an authorized DELTALAB dealer.

This warranty gives you specific legal rights, and you may also have other rights which vary from state to state.

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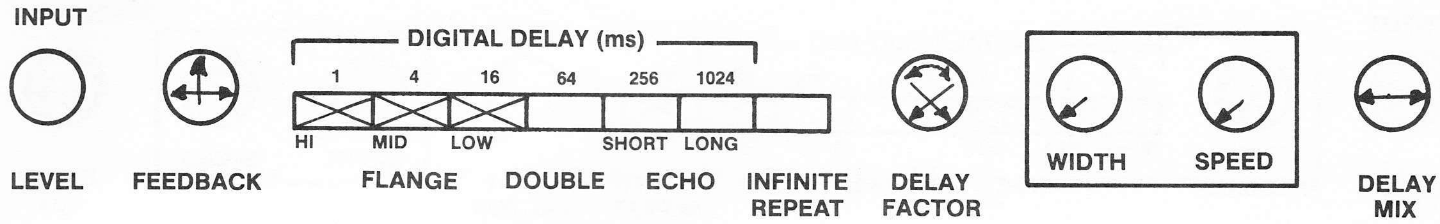
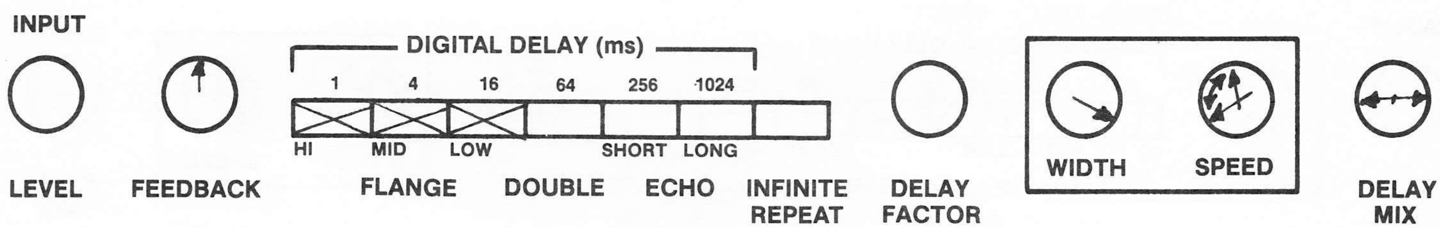
TYPICAL SET UPS



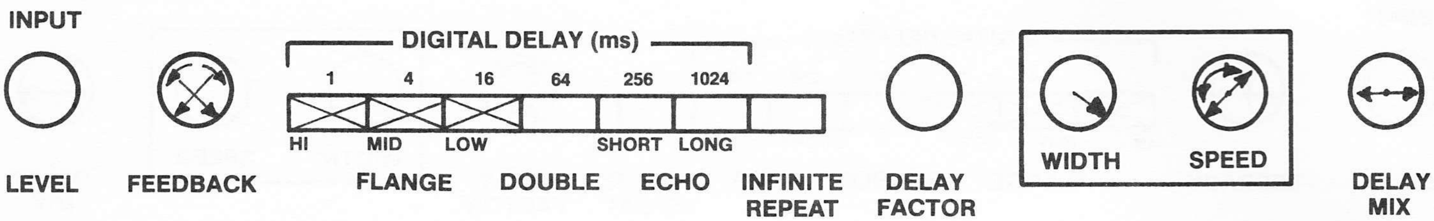
EFFECT: COMB FILTERING

Use any of the FLANGE delays to taste -- results in comb filter frequency response. FEEDBACK increases intensity. Phase reversal in DELAY MIX or FEEDBACK (if used) changes character of sound.



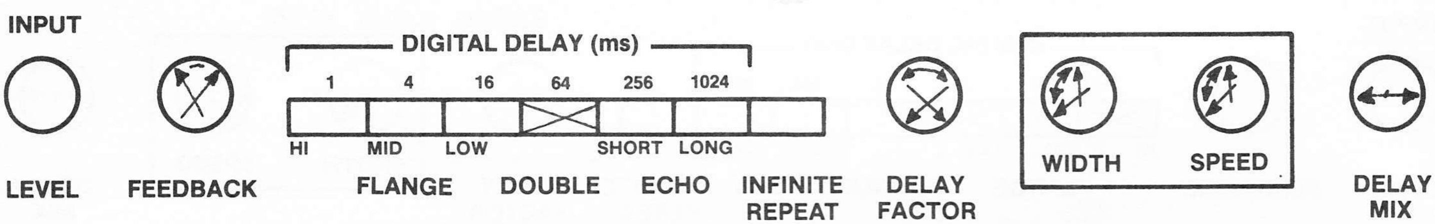
EFFECT: FLANGING

Use any of the FLANGE delays to taste. Generally the shortest delays provide the most effective flanging of sounds with dominant energy at high frequencies. Longer delays provide effective flanging for mid frequency vocal, etc.



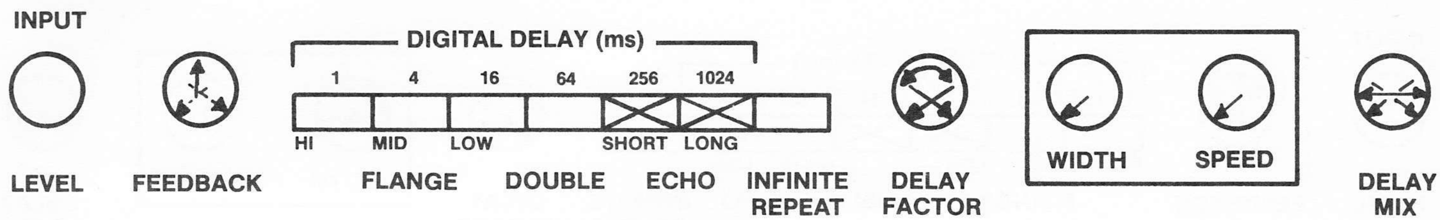
EFFECT: FLANGING WITH FEEDBACK

Use any of the FLANGE delays. Experiment with FEEDBACK, WIDTH, SPEED & DELAY MIX to taste. Note the differences in negative and positive FEEDBACK and DELAY MIX and combinations thereof.



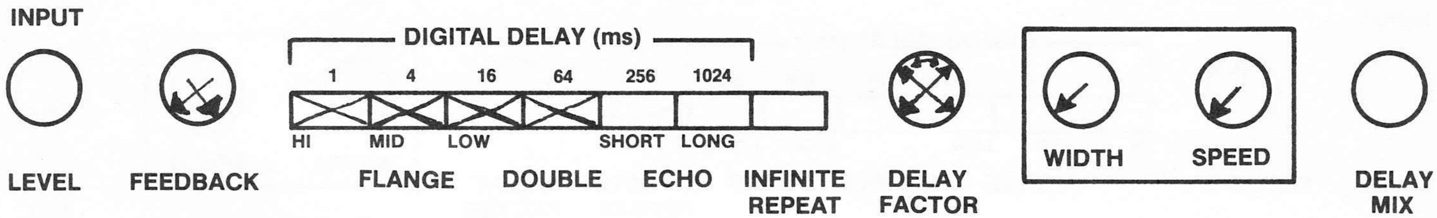
EFFECT: DOUBLING - THICKENING

Use delay button marked DOUBLE. Be sure DELAY MIX is at either +50% or -50%. Adjust DELAY FACTOR until a good doubling effect is heard. Try a small amount of FEEDBACK to get a chorus effect. Use the WIDTH and SPEED in small amounts to make the effect more realistic.



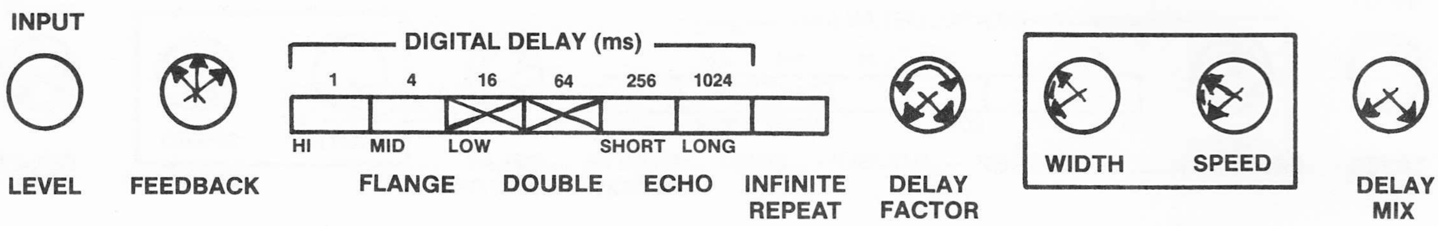
EFFECT: SLAPBACK - ECHOS

Use Delay button(s) marked for ECHO (ADM 1024 has two ECHO buttons, SHORT and LONG). With no FEEDBACK, the result is a slap echo. With FEEDBACK, the echos will repeat. The repeat time is controllable by the DELAY FACTOR. Use the DELAY MIX to emphasize the echo. By using more delay than source the result will be a pre-echo.



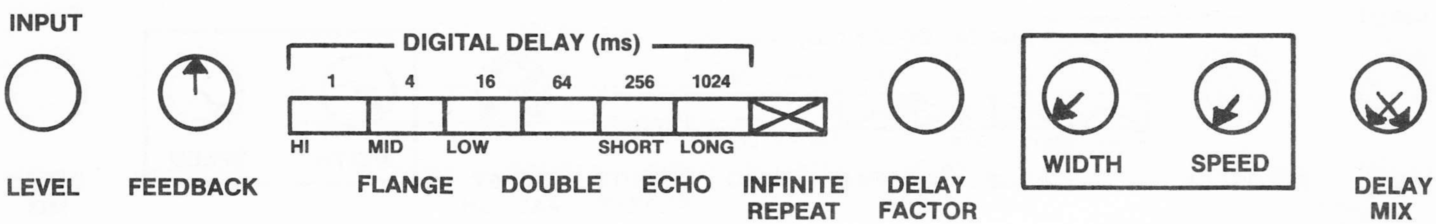
EFFECT: TUNED RESONANCE

Use any delay marked for FLANGE or DOUBLE with maximum FEEDBACK. Adjust DELAY FACTOR to control reverberant pitch. Best resonance is at 16 mS. With the DELAY FACTOR set from 0.5 to 0.75. Set DELAY MIX to either +100% or -100%.



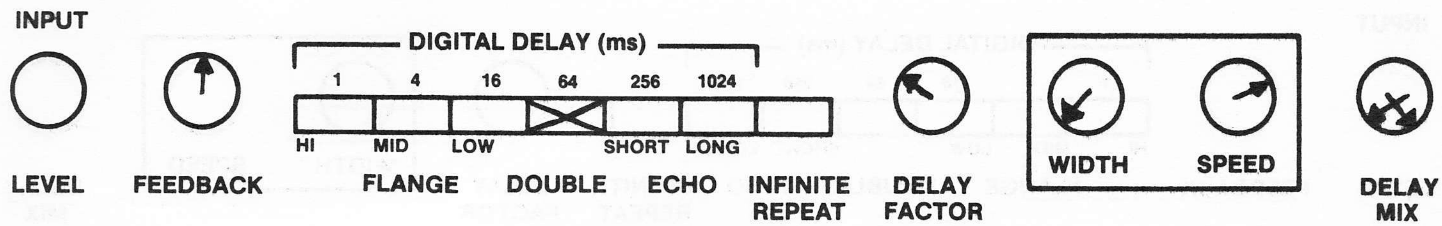
EFFECT: PRE-REVERB DELAY

Use delays greater than 10 mS to avoid comb filtering. Use moderate amounts of FEEDBACK, SPEED and WIDTH for realism.



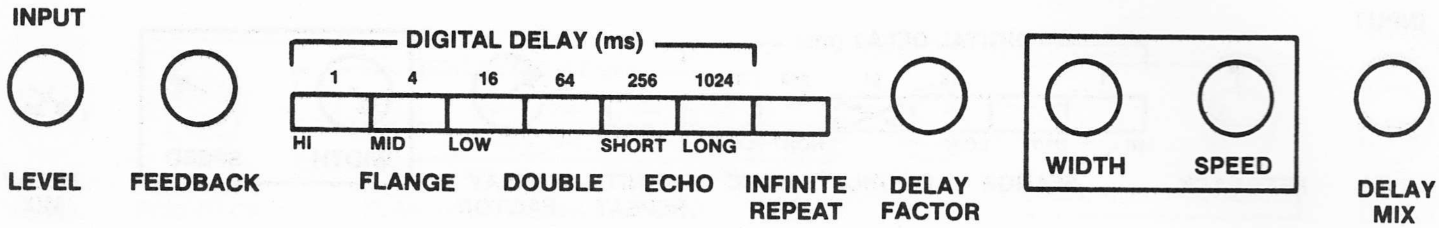
EFFECT: INFINITE REPEAT

Use the "RED" button marked INFINITE REPEAT to place the signal in full memory (256 mS max. for the ADM 256 and 1024 mS max. for the ADM 1024). The delay buttons can be used to synchronize the start of the segment to be stored. The DELAY FACTOR can be used to shorten the full memory from the indicated maximum.



EFFECT: VIBRATO

Use the DOUBLE delays and experiment with SPEED, WIDTH, DELAY FACTOR and FEEDBACK controls. Note that the ADM process also adds a little tremolo with the vibrato.



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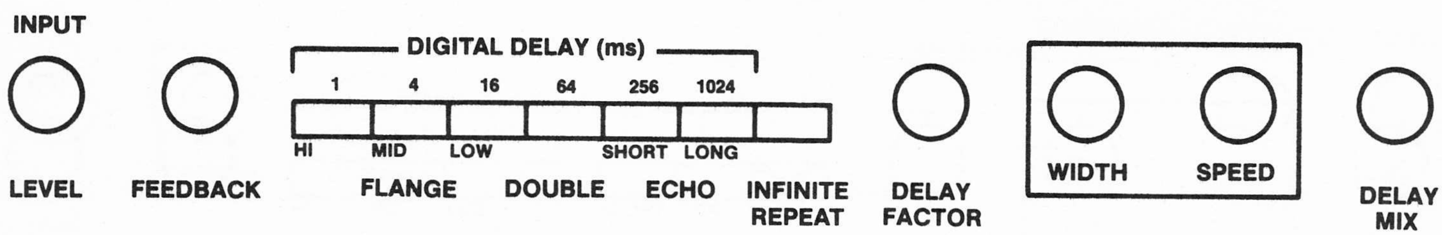

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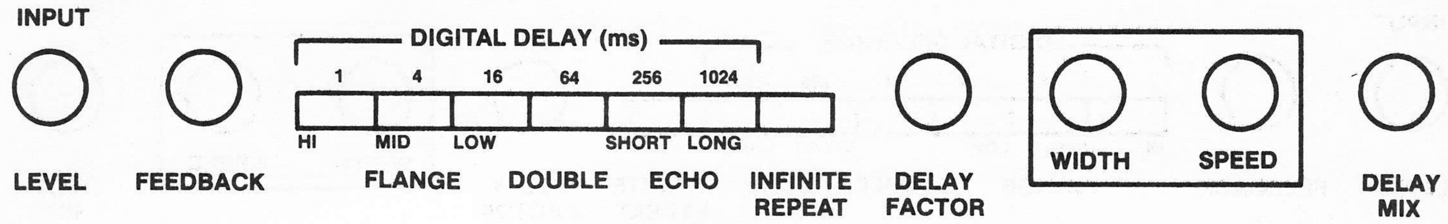

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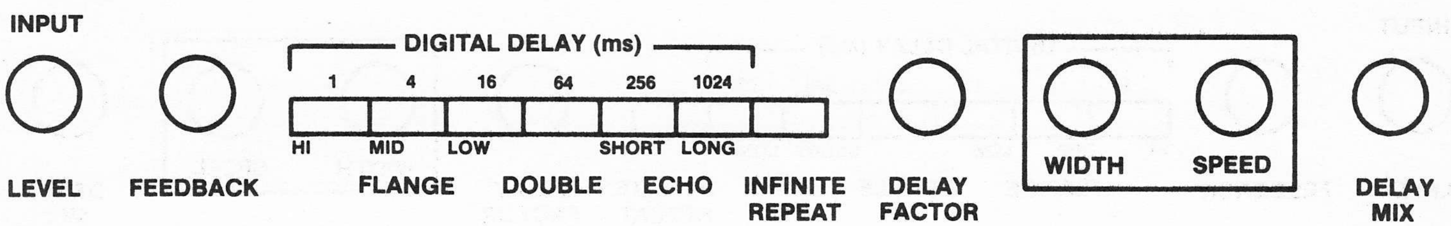

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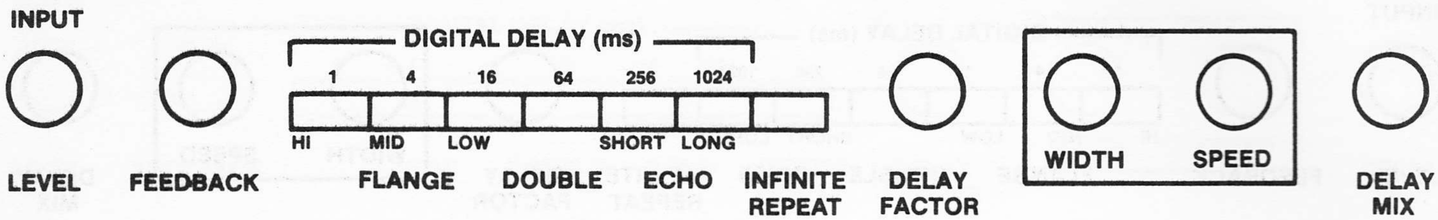

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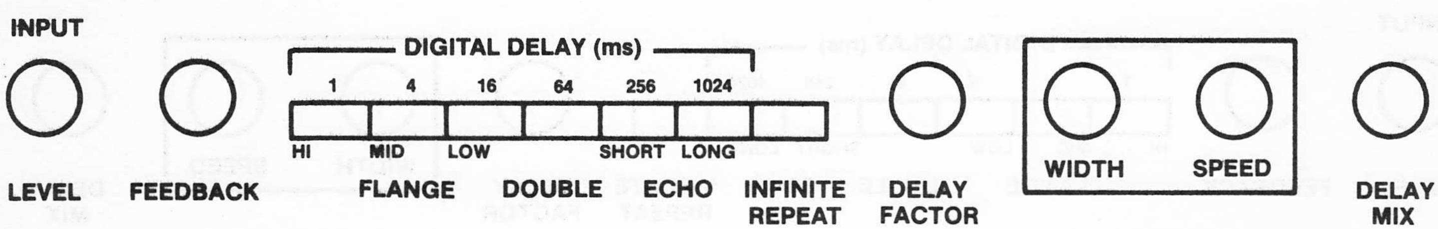

EFFECT:

DeltaLab

DeltaLab Research, Inc., 19 Alpha Rd., Chelmsford, MA 01824 Tel: (617) 256-9034 Telex: 951205