MXR digital time delay

Operation Manual

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INTRODUCTION

The MXR Digital Time Delay is a professional digital delay incorporating advanced analog and digital signal processing technology. The controls are easy to use and logically arranged, making the Digital Time Delay an all-purpose delay system.

The Digital Time Delay offers performance and versatility unmatched by other delays in its price range. Delay times from 0.31 to 320 milliseconds (one thousand milliseconds, ms, equals one second) are provided at a bandwidth of 10 kHz with no alaising (high frequency distortion artifacts).

The Digital Time Delay has sweep capabilities (as much as a 4:1 range) for professional quality flanging, chorus-effect and doubling. The full range of fixed delay effects is also available. Simulated stereo outputs are provided to enhance all the effects the Digital Time Delay is capable of.

We recommend that you first read through the operations manual to learn more about the Digital Time Delay and its operation. Note that when actual labels are referred to, they will appear in bold type (e.g. the **Regen** control).

INSTALLATION

MECHANICAL REQUIREMENTS

The Digital Time Delay conforms to standard EIA specifications of 19 inch rack mountable equipment. It is one standard rack height (1.75 inches) and 19 inches in width. The rear panel contains all audio, control and power connections.

POWER REQUIREMENTS

Power to the unit is provided through a three-wire power cord. Models used within the United States will operate on a standard AC line between 105 and 130 VAC, 60 Hz. Models manufactured for use outside the United States have been modified to comply with the required electrical specifications.

AUDIO CONNECTIONS

All signal inputs and outputs are located on the rear panel. Connections are made via standard unbalanced ¼ inch phone jacks.

The audio input has a high input impedance suitable for high or low impedance sources such as musical instruments or mixing consoles. The maximum input level is +18 dBm with the **Level** switch (on the rear panel) in the **Line** (out) position. When the **Level** switch is in its **Inst** (depressed) position the maximum input level is +6 dBm. The unit maintains unity gain from **Input** to **Output 1** regardless of the position of the **Level** switch.

The audio outputs have a low output impedance enabling them to drive long lines or any reasonable input impedance. **Output 1** is controlled by the **Mix** control on the front panel. **Output 2** consists of the delayed signal mixed equally with the input signal. The delayed signal in **Output 2** is inverted (shifted 180 degrees) with respect to the delayed signal of **Output 1**. When the unit is bypassed both outputs contain the input signal.

The **Bypass** jack allows the unit to be bypassed from a remote location. The function can be operated by a switch that connects the jack's "tip" to its "sleeve" such as MXR's 121 Footswitch.

OPERATION

The MXR Digital Time Delay combines a high degree of technical sophistication with a versatile combination of controls and features. A major goal in the development of the Digital Time Delay was to make it easy to operate. The controls are logically grouped and clearly labeled. The front panel diagram is provided for reference.

POWER Switch

The **Power** switch turns the delay on or off. The LED next to the **Power** switch will light when power is applied. Make sure that subsequent amplifier systems have their volume levels turned down when power is applied to the delay.

IN/OUT Switch

The front panel **In/Out** switch permits the user to bypass the unit. The front panel LED marked **ACTIVE**, when on, indicates that the outputs contain the respective output mixes. When off, it indicates that the outputs consist of only the input signal. Please note that if AC power is not applied to the unit no signal will be present on the outputs.

The rear panel mono phone jack labeled **Bypass** allows remote control of the **In/Out** function. The **Bypass** jack acts to invert the position of the **In/Out** switch when shorted tip to sleeve. For example, if the **Bypass** jack is open when the **In/Out** switch is depressed the unit is active and the **Active** LED is lit. If the **Bypass** jack is now shorted, the unit is bypassed. If the **In/Out** switch is out, the unit is bypassed when the **Bypass** jack is open.

LEVEL Switch

The **Level** switch on the rear panel allows the selection of **Line** or **Inst.** operating levels. A two color LED on the front panel of the Digital Time Delay indicates signal level conditions of the effect circuitry. When red, it indicates potential clipping, when green, displaying adequate signal level, and when not illuminated, low signal level or no signal present at the input. This feature allows you to correctly select the position of the rear panel **Level** switch. When properly adjusted, the **Level** LED is a constant green with the musical peaks driving it red. If the LED is red more than 25% of the time, then the signal level is too high

OPERATION

and the **Level** switch should be placed in the out position. If the LED is never red, and is either off or dimly lit green the signal level is too low, and the Level switch should be depressed. The LED is red at approximately +6 dBm in the **Inst.** position and +18 dBm in the **Line** position. The green is fully on approximately 20 dB below the point at which the red comes on.

MIX control and OUTPUTS

The two delay outputs are located on the rear panel of the unit. **Output 1** is controlled by the **Mix** control. When the **Mix** control is turned fully counter-clockwise to the **Dry** position, **Output 1** consists of the dry, unprocessed signal. As **Mix** is rotated clockwise, the amount of dry signal decreases and the amount of delayed signal increases. When **Mix** is in the 12 o'clock position the dry and delayed signals are equally mixed. When **Mix** is rotated fully clockwise to **Delay**, only the delayed signal is present at **Output 1. Output 2** contains an equal mix of the dry and delayed signals regardless of the position of the **Mix** control. The delayed signal in **Output 2** is always inverted with respect to **Output 1** for stereo enhancement.

The level of **Output 2** is 6 dB lower than the level of **Output 1** due to passive mixing. This has the advantage that when both outputs are used for stereo effects and later mixed to mono, the effect is not completely cancelled out as it would be if both outputs were the same level. **Output 2** may be modified to be a delay-only output. This is discussed in the circuit description section of this manual.

DELAY TIME controls

The delay time is set using the three push-button switches in the center of the front panel and multiplier control to the left of these switches (please refer to figure 2). Assume for new that the multiplier control is in the x1, "times one" position (fully clockwise). The maximum delay available in each delay range selected is listed above the switches. The three switches allow the user to select among six possible delay ranges. Looking at the front panel note the numbers directly above the switches (**1.25, 5,** and **80**). These are the delay times selected by depressing the left, center, and right switches respectively (all delay times are listed in milliseconds, ms, 1000ms = 1 second). Also note the two numbers between the switches, **2.5** and **20.** The **2.5** ms range is selected by depressing the left and center switches simultaneously. The **20** ms range is selected by depressing the center and right switches simultaneously. One final delay setting, **320** ms, is available by returning all switches to the "out" position. This is achieved by partially depressing any switch and releasing it when the other switches pop out.

Other delay times are possible with the use of the delay multiplier control. As mentioned above, when this control is fully clockwise, **x1**, the delay time is listed on the front panel. As the multiplier control is turned counter-clockwise the delay time is continuously shortened. At the counter-clockwise position the delay time is one-fourth the listed value. Thus a four-to-one delay range is available for each of the six switch combinations.

To summarize, the Digital Time Delay provides a continuous range of delay times from 0.31 ms (1.25 × .25) to 320 ms in six ranges.

WIDTH and SPEED controls

The delay time of the Digital Time Delay may be automatically swept over the four-to-one range available in each delay range. The constantly moving delay facilitates the flanging, chorus and doubling (or automatic double tracking — ADT) effects that are unique to time delays. The **Width** control acts as a "mix" control between the sweep oscillator and the variable **Delay Time** control, and therefore sets the amount of sweep. When **Width** is set fully counter-clockwise to the 0% position there is no sweep of time delay. As the **Width** control is rotated clockwise, the sweep oscillator gradually engages. The effect which occurs is identical to that produced by a regular back-and-forth rotation of the **Delay Time** control. At positions less than **100%**, the location within the overall delay range is determined by the variable **Delay Time** control. When rotated fully clockwise to the **100%** position, the time delay sweeps over the entire four-to-one range and the variable **Delay Time** control is disengaged. The **Speed** control adjusts the speed of the sweep from 0.1Hz in the counter-clockwise position to 10Hz in the clockwise position. The **Speed** control has no effect when the **Width** control is at **0%.**

REGEN control

The **Regen** (regeneration) control allows a variable amount of the delayed audio signal to be fed back into the input and delayed again. At short "flanging" delay settings it acts as an "intensity" control. At medium length delay it acts as a "decay time" control for the reverb effect that is created. At long delay times the **Regen** control adjusts the number of repeats that the delay line executes. In the **Min** position (fully counter-clockwise) the Digital Time Delay passes one single repeat of the input signal. As the **Regen** control is turned clockwise more repetitions are passed at successively smaller signal levels.

Regeneration is defeated when the Digital Time Delay is bypassed to prevent unexpected signals from occuring when the unit is activated.

INVERT DELAY switch

This switch inverts the phase of the delayed audio signal going to both outputs. This switch is primarily useful in "flanging" effects.

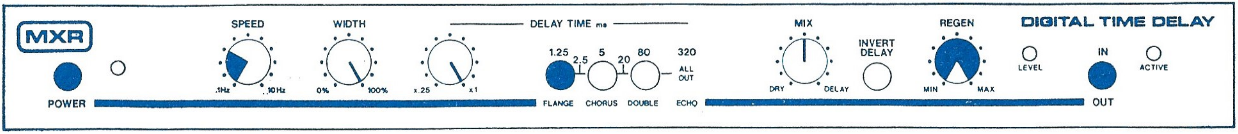
The inversion of the delay phase, in effect, swaps the two outputs because the two outputs have the opposite delay phase. This is noticeable in both stereo "flanging" and "chorus". The **Invert Delay** switch does not change the level of either output.

APPLICATIONS

Time delay is the basis of many effects. With some understanding of how much delay is required and how the controls interact you will be able to produce a myriad of sounds. The following applications are only intended as a starting point for your own discovery of new applications. Experiment - the Digital Time Delay cannot be harmed by any combination of control settings.

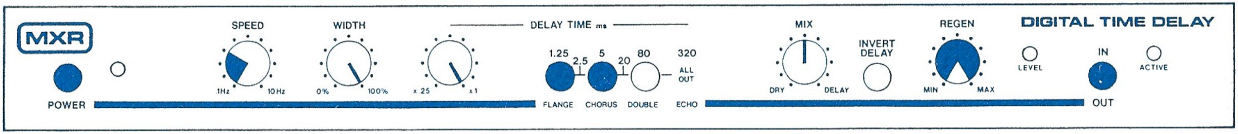
FLANGING

HIGH FLANGING



BUTTON IN

MID FLANGING



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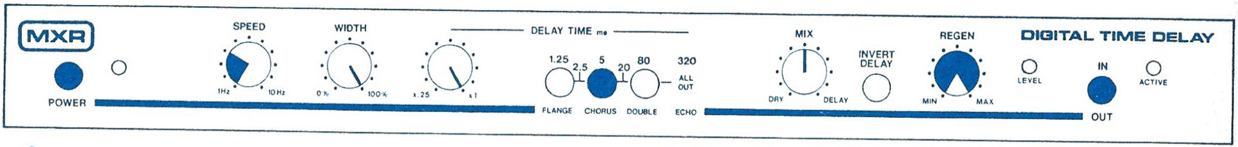
APPLICATIONS

LOW FLANGING



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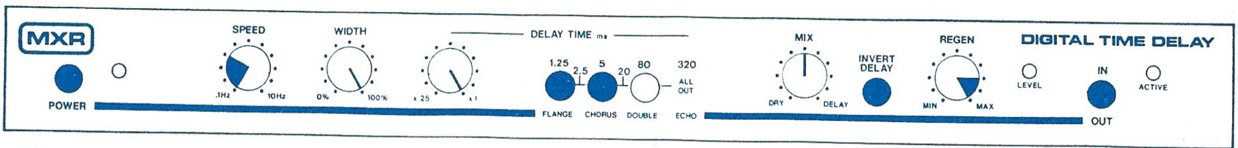
TALKING FLANGE



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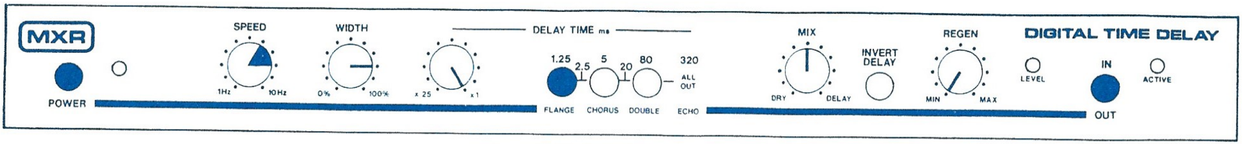
Very short time delay, up to 5 ms, when swept, produces flanging. The longer the delay time, the lower the pitch of the flange. The sweep is usually wide and slow. **Regen** is used to add intensity. The **Invert Delay** switch changes the tone of the bass frequencies. This can be used to produce a "talking" flange effect. The position of the **Delay Time** multiplier control has no effect when the **Width** control is set to 100%.

ROTATING SPEAKER AND VIBRATO

ROTATING SPEAKER I



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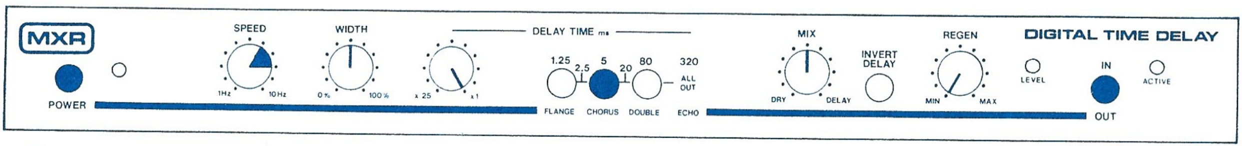
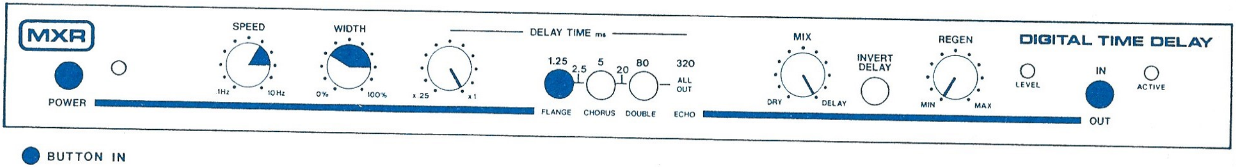
ROTATING SPEAKER II



BUTTON IN

APPLICATIONS

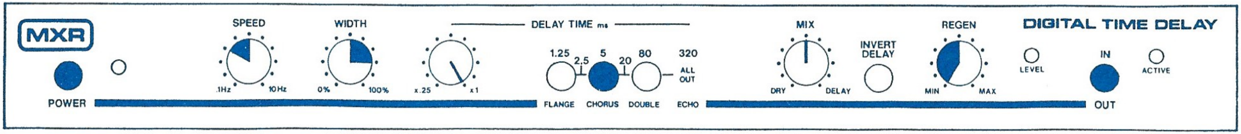
VIBRATO



Rotating speaker and vibrato effects are also in this range. Medium wide and fast sweep with no **Regen** produce these effects. Try these in stereo.

CHORUSING

CHORUS I



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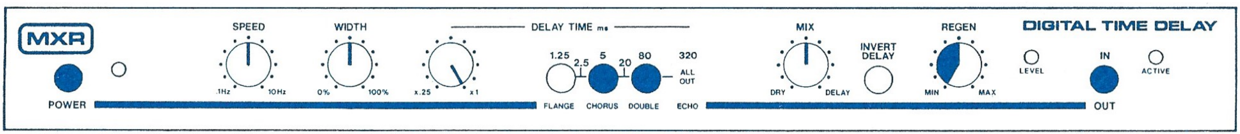
CHORUS II



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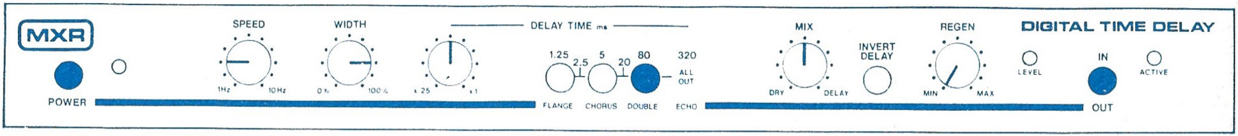
APPLICATIONS

CHORUS III



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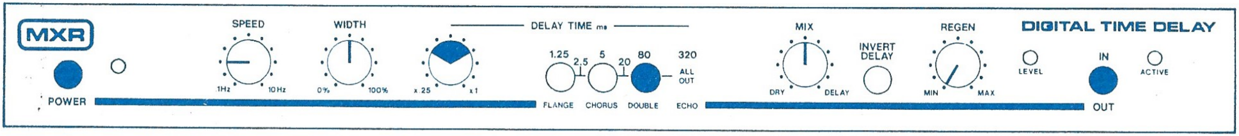
The chorus effect uses a little more delay than flanging; usually 5 ms to 40 ms. You will notice, **Speed** and **Width** interact. When increasing **Speed**, the **Width** is usually decreased to avoid to much pitch bending. The longer the delay used, the spacious the chorus. **Regen** adds intensity. This is a beautiful stereo effect.

APPLICATIONS

DOUBLING



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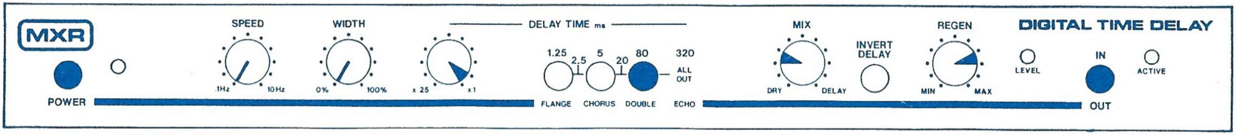
Doubling is a simulation of a voice or instrument in unison with itself. The time range is 20 ms to 80 ms. You will find that in "live use" you will need to use more delay than when recording. A little sweep is used to keep the time between the sounds constantly varying. This makes the effect more realistic because two real unison sounds will never be exactly in time. Doubling sounds great in stereo.

APPLICATIONS

SIMPLE REVERB



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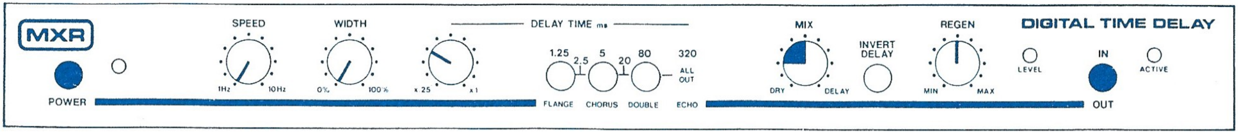
You can simulte reverb by properly adjusting the **Mix** and **Regen** controls in the 80 ms range. The **Mix** controls the amount of reverb and the **Regen** controls the decay.

APPLICATIONS

"ELVIS" ECHO



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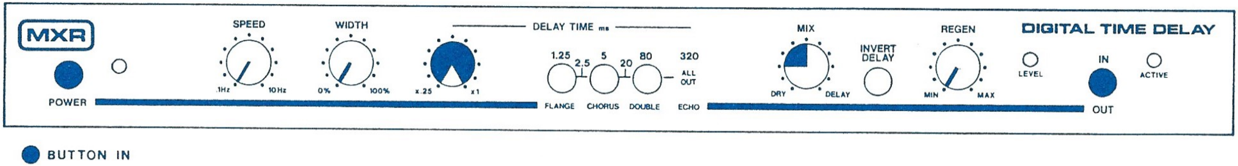
The old Sun records vocal sound can easily be duplicated.

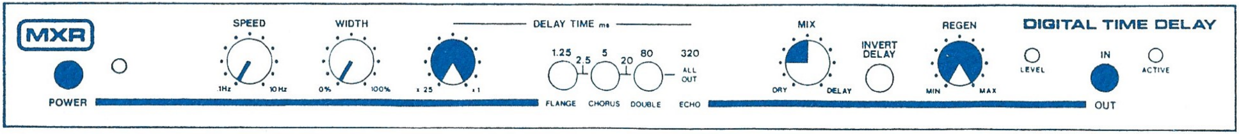
APPLICATIONS

SLAP ECHO



Single repeat echo is a popular studio effect.

REPEATING ECHO



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The **Regen** controls the number of echoes.

CIRCUIT DESCRIPTION

In the following discussion a brief description is given of each stage in the signal chain. Refer to the block diagram following this section.

Input Amplifier

The input signal is first processed by the Input Amplifier which provides buffering between the source and the internal circuitry. This amplifier provides unity gain with the rear panel **Level** switch in the **Line** (button out) position and 12 dB of gain in the **Inst** (button in) position.

The output of this amplifier goes to the Regen Sum Amp where it is combined with the delayed audio signal, if any.

Input Filter

The audio signal is then processed by a 10 kHz low pass (high cut) filter. The purpose of this filter is to prevent distortion which would occur if the incoming audio signal has frequencies (or harmonics) above half the sample rate.

Some background information may help explain why input filtering is necessary. The time delay circuitry constitutes a sampled data system. The incoming signal is sampled at discrete instants of time. In order to maintain an accurate representation of the signal, at least two samples must be made during each period of the highest frequency component of the signal. If this criterion is not met, that is, if the frequency of the incoming signal or its harmonics are above half the sample rate, an unpleasant type of distortion known as "aliasing" is produced. Since the sample rate is limited in frequency for a number of practical design reasons, filtering of the input signal is necessary to prevent aliasing.

Compressor

After filtering, the signal is then run through the Compressor. The Compressor provides linear compression of the audio signal which compresses the dynamic range of the signal prior to the analog-to-digital, digital-to-analog process which follows. The signal will later be restored to its original dynamic range by the Expander. The purpose of this compression-expansion process (known as companding) is to limit the dynamic range required by the A-D, D-A process, and yet retain a large dynamic range (high signal-to-noise ratio) for the user.

Tied to the Compressor is a level detection circuit which activates the two-color **Level** LED on the front panel.

Analog to Digital Converter

The A-D converter samples the audio signal from the Compressor at a rate of 25,000 to 100,000 times per second depending on the position of the variable Delay Time multiplier control. The purpose of this sampling is to provide temporary storage while the A-D process takes place. After each sample is taken, it is converted to a digital word to be saved in memory.

As samples are being written into memory, they are also being read out at the same rate, the D-A portion of the circuitry converts the words from memory back into an analog signal.

Digital Circuitry

The digital circuitry may be divided into three parts, Timing, Memory and Address Generation.

The Address Generation section supplies the proper location in memory to access when a read or write takes place. When a write takes place, the address is simply the location following the previous write location. Addressing memory is circular in that the first location in memory follows the last. To do reading, one may think of the read address as some distance behind the write address. The distance between the read address and the write address is determined by the **Delay Time** range switch. When a read takes place, the distance is subtracted from the write address to give the read address.

The Timing and Control section generates timing pulses to orchestrate the sampling of audio, converting it to a digital word, storing the word in memory, reading a word from memory and converting to audio.

CIRCUIT DESCRIPTION

The Memory is where the actual storage takes place. The Digital Time Delay uses 65,536 bits of memory, all contained in one integrated circuit.

Expander and Output Filter

Following the D-A, the signal is expanded in a manner complimenting the Compressor as described above.

The Output filter is used to filter out very high frequency artifacts that were introduced by the sampling process.

Inverter

After the Output Filter is an electronic switch which allows turning off the delay signal during bypass. This is done to disable regeneration and the delayed audio signal to **Output 2** during bypass.

The Inverter circuit is controlled by the front panel **Invert Delay** switch and allows changing the delayed signals polarity with respect to the input signal. It is not necessary to provide separate delay inversion for both the mix and regen signals because when the signal is inverted (in any effect where it is noticeable) it is desirable to have the polarity of both the mix and regen signals the same.

After the Inverter the signal is distributed three places, the **Regen** control, the **Mix** control and **Output 2.** The **Regen** control simply adjusts how much signal is fed back into the delay at the Regen Sum Amp.

Output Amp

The Output Amp has two functions. It is used as a summation amplifier for the output **Mix** control and as a buffer to allow a low impedance at the **Output** jack. When the Digital Time Delay is bypassed, the **Mix** control is bypassed and dry signal from the input amp is passed to the output.

After the Output Amp is part of the **Level** switch. This switch attenuates the output signal 12 dB when the **Level** switch is in the **Inst** position. This corresponds to the boost in signal that takes place at the Input Amp in this position.

Output 2

The **Output 2** signal is generated by passively summing the buffered input signal with the delay signal. The delay signal that is used is after the switchable Inverter state and prior to the output mix amp. The main output amp is inverting which means that the signal previous to it will be inverted with respect to **Output 1.** Therefore, the delay portion of the signal at **Signal 2** has the opposite polarity of the delay signal at **Output 1.**

It may be desirable for some applications to modify **Output 2** to be a delay only output. An example would be if the Digital Time Delay is connected to the echo send and receive of a mixing board and the mixing between the delayed signal and the dry is done at the mixing board. When used as in this example, if the Digital Time Delay is bypassed there would be two signal paths for the dry signal. To properly defeat the effect, the echo return signal must be turned off. This may be done at the board or by modifying **Output 2** to be a delay only output, and using the **In/Out** switch or external **Bypass** on the rear panel.

To modify **Output 2** to be a delay only output, have a technician remove the 2K ohm resistor at right rear of the pc board. The resistor is marked "DRY SUM OUTPUT 2". When this change is made, the output impedance at **Output 2** will be 300 ohms in the **Inst** position and 2k ohms in the **Line** position. The level at **Output 2** with respect to the input will be 0 dB in the **Line** position and attenuated 5 dB in the **Inst** position.

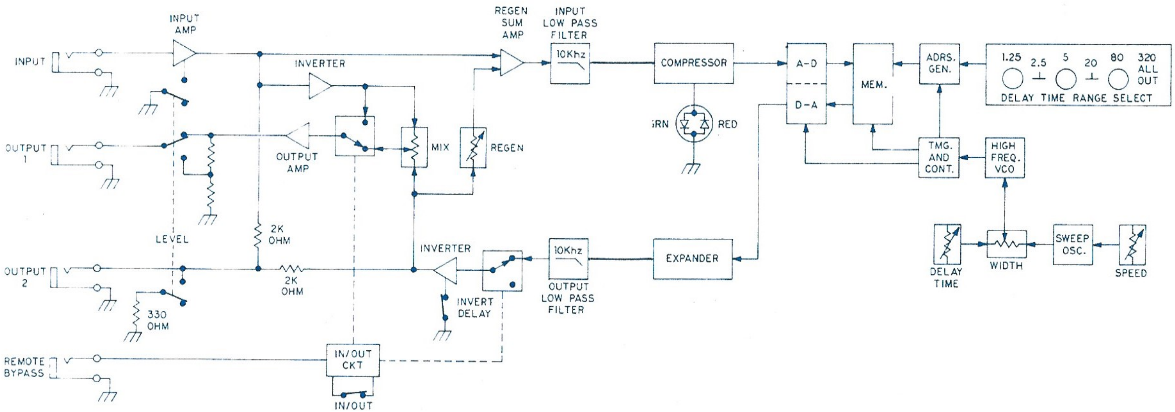
In/Out Circuit

As mentioned in previous sections, when the unit is switched out (bypassed), **Output 1** and **Output 2** are switched to the incoming dry signal, and, regeneration is defeated. The **In/Out** function may be done either by the front panel switch or electrically at the rear panel. The signal switching is done using CMOS switches. The rear panel jack has a positive voltage on it at a high impedence when open, shorting the tip to the sleeve (ground) effectively reverses the position of the **In/Out** switch.

Delay Time Controls

The **Speed, Width,** and **Delay Time** multiplier controls all have an effect on a high frequency voltage controlled oscillator (VCO). The VCO output is fed to the Timing and Control section. The **Delay Time** range switch affects memory addressing and is discussed above in the Digital Circuitry section. A low frequency oscillator is used to sweep the delay time. This oscillator produces a sine wave of a frequency set by the **Speed** control. The **Delay Time** multiplier control sets a DC voltage which is mixed with the sweep oscillator at the **Width** control. The width circuitry will not allow the four to one range of delay time to be exceeded. The **Width** control is non-linear such that at its mid point, the amount of sweep is approximately 10%.

BLOCK DIAGRAM



SPECIFICATIONS

Maximum Input Level	+20 dBm
Maximum Output Level	+18 dBm
Input Impedance	470 k ohms
Output Impedance	100/666 ohms, high/low level
Dynamic Range	greater than 90 dB
T.H.D.	less than 0.5% (at 400 Hz)
Frequency Response	20 Hz to 22 kHz ±1 dB dry 20 Hz to 10 kHz +1, −3 dB delay
Sweep Frequency	0.1 Hz to 10 Hz
Sweep Width	4:1 range, sine waveform
Delay Range	0.31 ms to 328 ms
Dimensions	1¾″ high × 6¼″ deep, EIA rack width

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FULL WARRANTY

All products in the MXR Professional Products Group are warranted to function properly for a period of one year from the date of purchase. If any unit fails to function properly within the warranty period, free repair, and the option of replacement or refund in the event MXR is unable to repair the unit, are MXR's only obligations. This warranty does not cover any consequential damages, or damage to the unit due to misuse, accident or neglect. MXR retains the right to make such determination on the basis of factory inspection. Products returned to MXR must be shipped prepaid. This warranty remains valid only if repairs are performed by MXR, and provided that the serial number on the unit has not been defaced or removed. This warranty gives you specific legal rights, and you may also have other rights which vary from state to state.

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