Temecula DSP MSP-126

Multi-Tap Stereo Processor — User Guide

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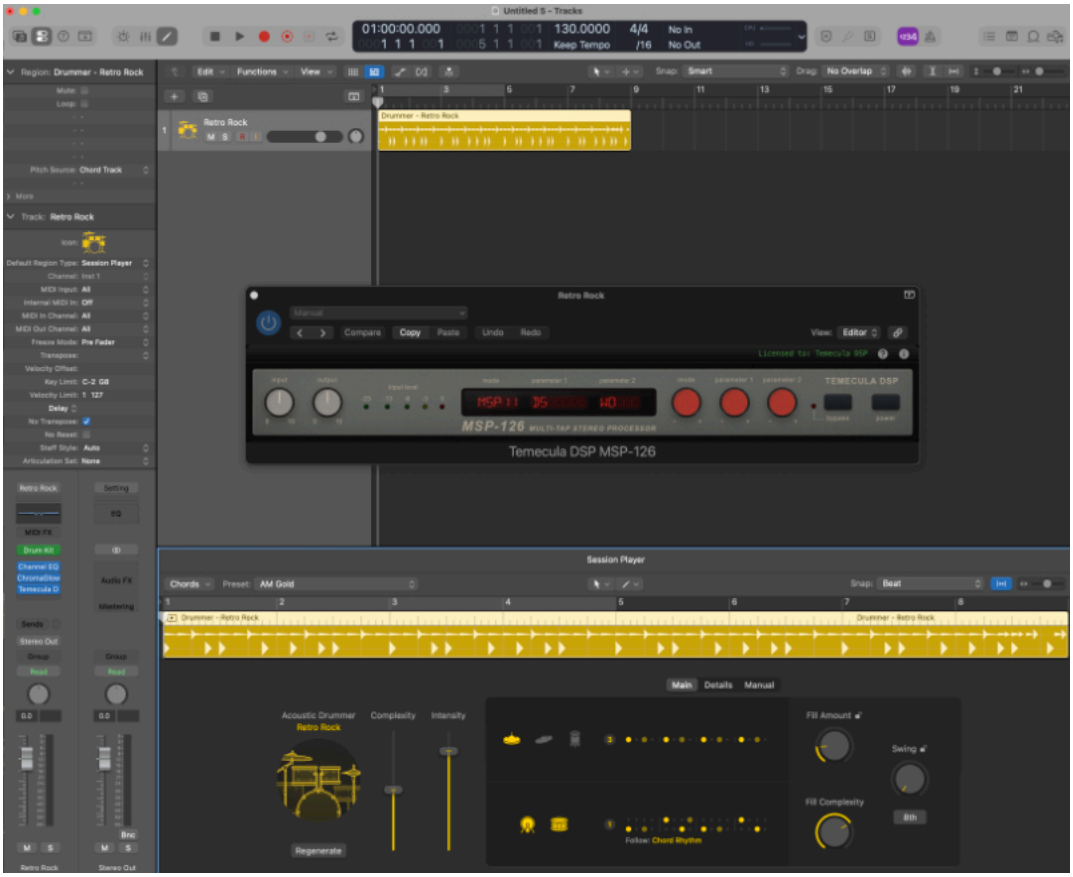
Introduction

The Temecula DSP MSP-126 is a digital recreation of the Ursa Major MSP-126 Multi-Tap Stereo Processor, a unique signal processor from the mid-1980s that used a 12-tap stereo delay algorithm to create a wide range of spatial and temporal effects from a single mono or stereo input.



Unlike conventional delay lines that offered a single tap, the MSP-126 delivered rich, full stereo through two groups of six independently controlled delay taps. This architecture enabled everything from transparent mono-to-stereo conversion to dramatic echo effects, all with careful attention to mono compatibility and natural-sounding stereo imaging.

The MSP-126 plugin faithfully emulates the original hardware's PROM-driven architecture, including all eight processing modes, the 16K sample delay memory, the sub-ranging ADC, and the multiplexed display system driven directly from the original firmware ROM data.

Features

- 8 processing modes: MSP11, CSP, ROOM, DLAY, PAN, DDL, RPTS, and SCALE
- 12 independently controlled delay taps (6 per channel)
- Up to 376ms of delay time at a 44.1kHz sampling rate
- 16 parameter variations per knob for each mode (256 combinations per mode)
- Authentic 16-segment LED display decoded from the original U64 display ROM
- Peak level metering with 5-segment LED indicator
- Mono input support with automatic stereo upmixing

- Available as VST3, AU, and AAX

System Requirements

- macOS 10.13.15 or later (AU, VST3, AAX)
- Windows 10 or later (VST3, AAX)
- A DAW that supports one of the above plugin formats

Emulated Specifications

These specifications reflect the behavior of the original hardware as emulated by the plugin:

- **Bandwidth:** 20kHz at 44.1kHz sampling rate
- **Dynamic range:** 80+ dB (15-bit PCM conversion)
- **Processing modes:** 8
- **Delay memory:** 16K samples (376ms maximum delay)
- **Delay taps:** 12 (6 left, 6 right)

Controls Reference

1. Input Level

The input level knob adjusts the gain applied to the incoming signal before it enters the delay processor. Set the level so that the green peak LEDs illuminate frequently, the yellow LED flashes on peaks, and the red LED lights only rarely. This mirrors the gain staging behavior of the original hardware's front panel control.

2. Output Level

The output level knob controls the volume of the processed (or bypassed) signal at the plugin output. Use this to match levels with your session or to compensate for gain differences between processing modes.

3. Peak Level Meter

The five-segment LED meter displays the peak signal level at the processor input. The meter uses a brief peak-hold behavior to capture transients. The processor operates with good headroom when the meter stays in the green range.

4. Display Window

The 16-character alphanumeric display shows the current mode, parameter 1 value, and parameter 2 value using the same character encoding as the original hardware's DL2416 display modules.

5. Mode Selector

The mode knob selects one of eight processing modes. Each mode configures the 12 delay taps with different time and gain patterns optimized for a specific type of processing. The display window shows the current mode name along with the parameter values.

Because changing modes alters most or all of the tap delay times simultaneously, brief audio glitches may be audible if audio is playing during a mode change. This is an authentic behavior of the original hardware caused by sudden changes in the delay line read positions.

The eight modes are: MSP11, CSP, ROOM, DLAY, PAN, DDL, RPTS, and SCALE.

6. Parameter 1

The parameter 1 knob selects one of 16 delay time variations within the current mode. Each mode assigns a different meaning to this control. For example, in ROOM mode, parameter 1 sets the longest reflection time, while in DDL mode it sets a delay multiplier.

The display shows a mnemonic indicating the current parameter 1 value in the context of the selected mode.

7. Parameter 2

The parameter 2 knob selects one of 16 gain and mixing variations within the current mode. Each mode assigns a different meaning to this control. For example, in ROOM mode, parameter 2 sets the dry/wet mix ratio, while in SCALE mode it controls the filter intensity.

The display shows a mnemonic indicating the current parameter 2 value in the context of the selected mode.

8. Bypass

The bypass button routes the input signal directly to the output, bypassing the delay processor entirely. When bypass is engaged, the bypass LED illuminates.

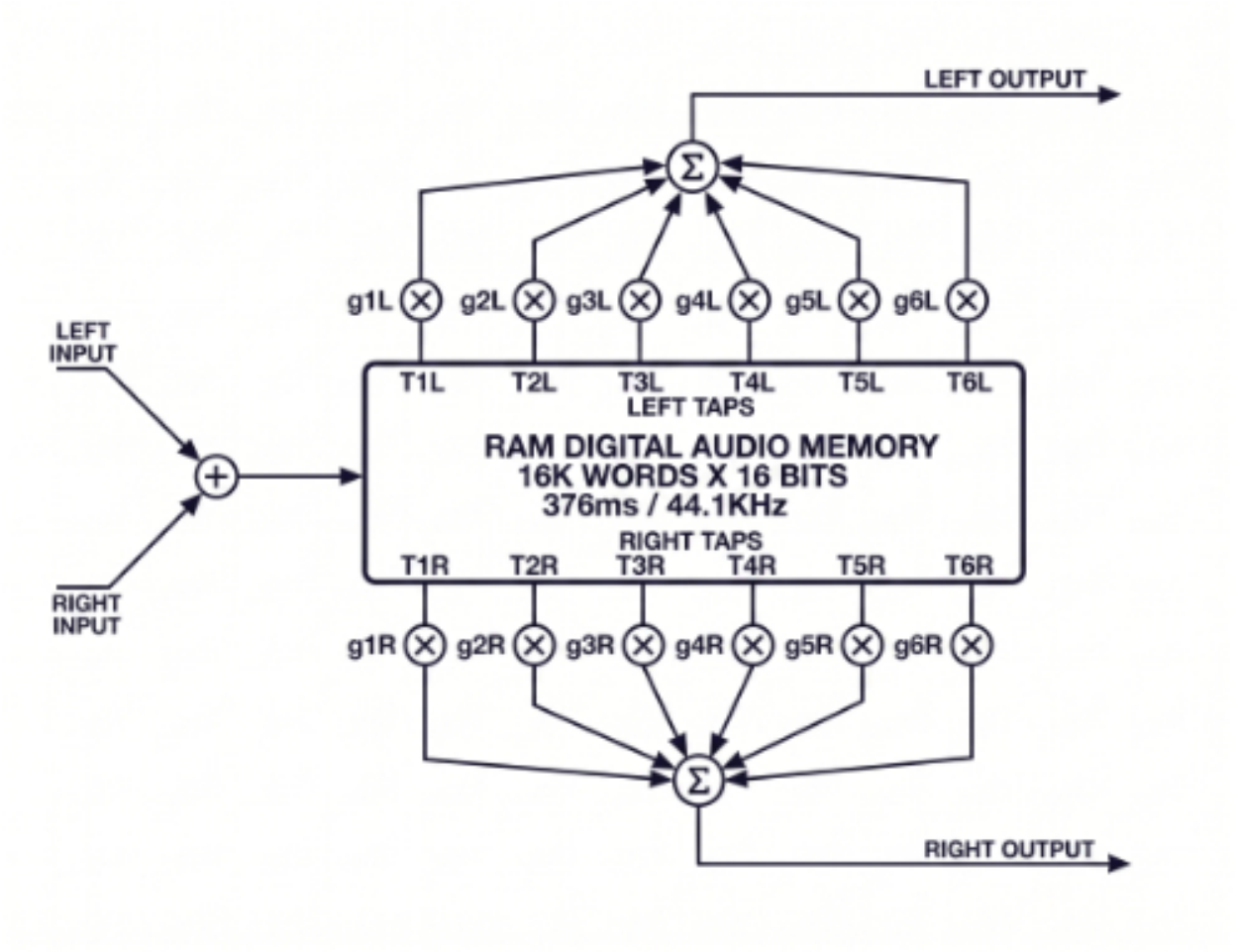
9. Power

The power button acts like the bypass button.

How It Works

The MSP-126 is fundamentally a digital audio delay system with a distinctive multi-tap architecture. It converts the input signal to digital form and writes it into a 16K-sample RAM buffer. From this

buffer, it reads twelve separate delay taps — six assigned to the left output and six to the right — each with independently programmable time delay and gain.



The time delays, gains, and polarities for all twelve taps are stored in EPROMs (Erasable Programmable Read-Only Memories). Three front panel controls — Mode, Parameter 1, and Parameter 2 — address different locations in these EPROMs to select from hundreds of pre-programmed tap configurations.

The Mode selector has the highest priority and establishes the overall processing algorithm. Parameter 1 accesses 16 variations of the tap time delay placements, while Parameter 2 accesses 16 variations of the tap amplitude and phase values. This gives each mode up to 256 unique combinations of delay and gain settings.

The Multi-Tap Advantage

Many useful audio processing functions emerge from this multi-tap architecture. When all the delay times are short, the effect manifests as frequency response changes (comb filtering, spatial widening). When the delays are longer, the individual taps become audible as discrete echoes or reflections.

This approach occupies a unique space between conventional delay lines (which offer only a single tap) and reverberators (which produce dense, sustained echo patterns). The MSP-126's

twelve taps provide enough complexity for convincing spatial processing while remaining simple enough that each tap's contribution can be precisely controlled.

The Display System

The MSP-126's display is driven by a dedicated EPROM (U64) that stores the character data for every combination of mode and parameter settings. The plugin emulates this system by reading directly from the original ROM data, applying the same bit-reversal mapping used by the hardware's DL2416 alphanumeric display modules, and rendering the result using a 16-segment display component.

Processing Modes

The MSP-126 offers eight processing modes, each configuring the twelve delay taps for a different type of audio processing. This section describes each mode and how its two parameter controls affect the sound.

MSP11 — Multi-Tap Stereo Processing

- **Parameter 1 (Depth):** D5 (minimum) to D100 (maximum)
- **Parameter 2 (Width):** W0 (minimum) to W100 (maximum)

Converts mono to stereo using carefully designed time delay patterns that maintain flat frequency response across the left output, right output, and their mono sum. At minimum width, the output is mono with a small delay offset. At higher depth settings (above approximately D60), individual delay taps may become audible with certain program material. For general-purpose processing where transparency is paramount, keep depth below D60.

CSP — Comb Filter Stereo Processing

- **Parameter 1 (Frequency):** 200Hz to 14kHz
- **Parameter 2 (Width):** W0 (minimum) to W100 (maximum)

Creates stereo from mono using a pair of complementary comb filters. The left and right channels have opposite filter patterns: where one channel has a peak, the other has a null. Despite the dramatic spectral coloring of each channel individually, summing them to mono produces a flat frequency response — making this mode highly mono-compatible. Parameter 1 sets the frequency of the lowest comb peak (left channel) and corresponding null (right channel). Parameter 2 controls the depth of the nulls. At minimum width, the output is mono with flat response and zero delay.

ROOM — Early Reflections

- **Parameter 1 (Reflection Time):** 5ms to 360ms
- **Parameter 2 (Mix):** 0% (all dry) to 100% (all reflections)

Simulates the early reflection pattern of acoustic spaces. Parameter 1 controls the longest reflection time, which corresponds to the apparent size of the simulated space. Parameter 2 adjusts the

balance between the dry signal and the reflections, which corresponds to the apparent distance from the source. Coloration is minimal in this mode, especially at reflection times above 50ms.

DLAY — Delay Cluster

- **Parameter 1 (Pre-Delay):** 20ms to 320ms
- **Parameter 2 (Mix):** 0% (all dry) to 100% (all delay cluster)

Adds a brief cluster of stereo delays after an adjustable initial silence. The cluster itself adds body and width to the delayed sound, while the adjustable pre-delay creates separation between the direct sound and the cluster. Parameter 1 sets the time before the cluster begins. Parameter 2 controls the balance between the dry signal and the delay cluster.

PAN — Delay-Based Panning

- **Parameter 1 (Azimuth):** 90 degrees left through center to 90 degrees right
- **Parameter 2 (Width):** W0 (minimum) to W100 (maximum)

Uses short clusters of time delays to create the perception of a source positioned at a specific left-right location. Unlike conventional amplitude panning, this delay-based approach maintains equal energy in both channels at all settings and preserves flat frequency response in each channel individually. At higher width settings, the mono sum also maintains flat response. This means the spatial positioning holds up well even when the stereo signal is summed to mono.

DDL — Digital Delay Line

- **Parameter 1 (Multiplier):** 1x to 16x
- **Parameter 2 (Delay):** 1ms to 20ms (left, right, or both channels)

Provides a conventional single-tap delay. The displayed delay value is applied to the indicated output channel, while the other channel receives a zero-delay signal. The two parameters combine to set both the delay time and which channels are delayed.

RPTS — Repeats

- **Parameter 1 (Delay Time):** 10ms to 360ms (time of the last repeat)
- **Parameter 2 (Repeats):** 2 to 10 repeats with selectable gain profile

Produces evenly spaced repetitions of the input signal that alternate between left and right channels. The first repeat appears at zero delay and the last at the time set by parameter 1, with the remaining repeats distributed evenly between them. Parameter 2 selects both the number of repeats and their amplitude envelope — equal level, increasing loudness, or decreasing loudness. An increasing loudness profile creates a reversed-tape effect.

SCALE — Musical Scale Filter

- **Parameter 1 (Interval):** Unison through a chromatic scale up to the minor third above the octave (15 half-steps total)
- **Parameter 2 (Intensity):** 0 (off) to +10 (peaks at all harmonics) or -10 (peaks at odd harmonics only)

Creates comb filters tuned to musical intervals. The left channel is fixed at A=440Hz and its harmonics. The right channel's pitch rises relative to the left as parameter 1 is increased, stepping through 15 half-tones of a chromatic scale. Parameter 2 controls the filter intensity. This mode is most effective with broadband source material such as noise, percussion, or dense textures. It can also be used on speech and vocals for dramatic special effects.

Usage Guide and Tips

Mono to Stereo Conversion

One of the MSP-126's core strengths is converting mono sources to stereo. Close-miked vocals, synthesizers, drum machines, and DI'd instruments are all inherently mono. Rather than using two microphones (which introduces comb filtering when summed to mono), the MSP-126 can process a mono signal into convincing stereo while maintaining excellent mono compatibility.

For the most transparent mono-to-stereo processing, use **MSP11** mode. The processed stereo output has flat frequency response in the left channel, right channel, and their sum. The stereo image is natural and balanced, equally effective on speakers and headphones.

CSP mode offers more dramatic stereo coloring through complementary comb filters. While each channel is spectrally colored, the mono sum remains flat — an unusual and useful property.

PAN mode creates a delay-based stereo image with adjustable position and width. At higher width settings, mono compatibility is excellent.

Adding Space and Depth

The MSP-126 can add the impression of acoustic space to dry recordings without the long decay tail of a reverberator.

ROOM mode provides early reflection patterns that simulate spaces of various sizes. At shorter reflection times, the effect is subtle spatial enhancement. At longer times, the reflections become more obvious. The dry/wet control (parameter 2) lets you dial in the apparent distance from the source.

DLAY mode adds a cluster of short delays after an adjustable pre-delay. The cluster adds body and stereo width, while the pre-delay creates the impression of distance. This is effective on vocals, where a separated cluster of reflections adds presence without muddying the direct sound.

Echo and Delay Effects

Several modes produce obvious delay-based effects suitable for creative sound design.

DDL mode provides a straightforward single-tap delay with selectable channel routing. Use it for simple slap-back, vocal doubling, or as a building block in more complex effect chains.

RPTS mode creates evenly spaced stereo repetitions that bounce between left and right channels. The number of repeats, their timing, and their amplitude profile are all adjustable. Try it on percussion or vocals. The increasing-loudness profile creates a reversed-tape illusion.

DLAY and **ROOM** can also produce audible echo effects when their delay times are set to longer values.

Spatial Positioning

The **PAN** mode offers a sophisticated alternative to conventional amplitude panning. By using multi-tap delay patterns rather than simple level differences, it creates a more robust stereo image that holds up better when the listener is not centered between speakers, and that maintains better mono compatibility than a simple delay-based pan.

The **MSP11** mode takes the opposite approach: it deliberately creates a diffuse, hard-to-localize stereo image. Use it when you want a source spread across the stereo field without a strong center or side bias.

Spectral Effects

CSP mode produces comb filtering similar to a static flanger, but with two complementary outputs. The left and right channels have mirror-image frequency response curves. The display shows the frequency of the first peak in the left channel (and corresponding null in the right). Higher-order peaks and nulls occur at integer multiples of the displayed frequency. Parameter 2 controls the null depth, from subtle to extreme.

SCALE mode creates musically tuned comb filters with a fixed pitch in the left channel (A=440Hz) and an adjustable pitch in the right channel. The two channels together produce a two-note interval. This mode is most effective with broadband material — noise, percussion, dense textures. On speech, it creates dramatic robotic or otherworldly effects.

Working with Reverb

The MSP-126 complements dedicated reverb plugins effectively. Insert it before a reverb to add early reflections and pre-delay that many reverbs lack. Set the MSP-126 to **ROOM** mode and adjust the reflection time and mix ratio independently of the reverb's own settings. This gives you separate control over the direct sound, early reflections, and late reverberation — the three components of natural room acoustics.

Alternatively, use **DLAY** mode before a reverb to increase the echo density of the reverb input, creating a smoother and more natural-sounding decay.

Tips

- **Mode changes may glitch.** Switching modes while audio is playing can cause brief clicks due to sudden changes in the delay tap positions. For clean transitions, switch during pauses in the material or automate mode changes at silent points.
- **Parameter 2 is the gentlest.** Of the three controls, parameter 2 (which adjusts gain values rather than delay times) produces the least audible artifacts when changed during playback.
- **Start with MSP11.** If you're new to the MSP-126, MSP11 is the most broadly useful mode. It produces transparent stereo enhancement that works on virtually any source material.

- **ROOM at short times is subtle.** At reflection times below 50ms, ROOM mode adds spatial width without obvious echoes. This makes it useful as a subtle enhancement on individual tracks during mixing.
- **RPTS for rhythmic effects.** The repeats mode is particularly effective on percussive material and can be used to create rhythmic patterns from a single hit.

Credits

Original Hardware

The Ursa Major MSP-126 Multi-Tap Stereo Processor was designed and manufactured by Ursa Major, Inc. of Watertown, Massachusetts, circa 1984.

- **Christopher Moore** — Designer, Ursa Major founder

The MSP-126 Owner's Manual was published by Ursa Major, Inc. in December 1984.

Plugin

The Temecula DSP MSP-126 plugin was developed by Temecula DSP.

Legal

MSP-126 is a model number originally used by Ursa Major, Inc. Temecula DSP is not affiliated with the estate of Christopher Moore, Ursa Major, Inc., or any successor entities.