

Temecula DSP MV-II

Digital Multi-Effects Processor — User Guide

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Introduction

The Temecula DSP MV-II is a precision digital recreation of the Alesis MidiVerb II — a landmark effects processor from 1986 that brought studio-quality reverb, chorus, flanging, and delay to a single rack unit at an unprecedented price point. Built around a custom VLSI chip designed specifically for the original hardware, the MidiVerb II delivered 99 carefully tuned factory programs covering everything from lush plate-style reverbs to gated drums, regenerated delays, and exotic stereo production effects.

This plugin faithfully reproduces the original DASP-16 signal processing architecture at its native sample rate of 31,250 Hz, preserving the warm bandwidth and subtle character that made the hardware a studio staple throughout the late 1980s and beyond. All 100 factory programs are included, from the smooth reverbs and punchy gates to the triggered flanges and multi-tap studio effects that gave the MidiVerb II its distinctive voice.

History of the MidiVerb II

The Alesis Revolution

In the mid-1980s, Alesis set out to democratize professional audio effects. Digital reverb units from Lexicon and AMS cost thousands of dollars and were found only in top-tier studios. The original MidiVerb (1986) broke new ground by offering digital reverb in an affordable, compact format. The MidiVerb II followed shortly after, expanding the concept with a dramatically larger program library, improved sound quality, and a more versatile control set.

The DASP-16 Chip

At the heart of the MidiVerb II sits the DASP-16 — a custom digital audio signal processor designed by Alesis specifically for this unit. The chip implements a delay-line architecture using 16 kilobytes of dynamic RAM as a circular buffer. Programs are defined as sequences of 128 instructions that read, sum, and write delay taps at precise time offsets within this buffer. By combining taps with half-amplitude summation and polarity inversion, the DASP-16 builds networks of allpass and comb filters — the fundamental building blocks of algorithmic reverb.

This single-chip design is what allowed Alesis to offer the MidiVerb II at its breakthrough price. The entire reverb engine runs on one piece of silicon, clocked at an internal rate of 31,250 Hz with 16-bit linear PCM resolution. The resulting 15 kHz bandwidth and wide dynamic range gave the unit a warm, musical character that producers found immediately usable.

A Studio Workhorse

The MidiVerb II found its way into countless home and professional studios throughout the late 1980s and 1990s. Its combination of instant-recall preset access, MIDI integration, and genuinely musical effects made it a workhorse for tracking and mixing alike. The gated and reverse reverb programs became particularly popular for drums, while the chorus and flanging algorithms found a home on guitars, keyboards, and vocals. The unit's characteristic warmth — a product of its limited bandwidth and integer arithmetic — became part of its appeal rather than a limitation.

How the MV-II Works

The Delay-Line Engine

Unlike modern algorithmic reverbs that rely on sophisticated diffusion matrices and feedback delay networks, the MidiVerb II uses a fundamentally simpler architecture: a single circular delay buffer with precisely placed read and write taps. This approach was dictated by the constraints of 1986-era silicon, but it produces a distinctive sound that more complex designs don't replicate.

The circular buffer holds 16,384 samples at the internal sample rate, providing a maximum delay time of approximately 524 milliseconds. A write head advances one position per sample tick. Each of the 100 programs defines a pattern of instructions that determine where taps are read, how they are combined, and where results are written back into the buffer.

Building Blocks

Each program defines a pattern of instructions that read delay taps, combine them at half amplitude, and write the results back into the buffer with positive or negative feedback gain. By chaining these operations at different tap positions, the engine builds networks of allpass and comb filters — the fundamental building blocks of algorithmic reverb — creating everything from tight room ambiances to long regenerated delays.

Dual Engine Architecture

The MDV-II runs two independent instances of the delay-line engine in series: Unit A and Unit B. Audio enters Unit A, is processed through A's selected program, then the result feeds into Unit B for further processing.

This allows you to chain any two programs together — a reverse reverb into a chorus, a gated reverb into a delay, or any other combination of the 100 available programs. Each unit has its own program selection, send level, pan position, and bypass control.

Series Processing

In the default configuration (both pans centered), the signal flows as a pure series chain: input → Unit A → Unit B → output. You hear only the final result after both stages of processing.

Pan-Spread Stereo

As you spread the pan controls apart, the two stages gradually separate in the stereo field. Unit A's output fades into one side while Unit B's output fades into the other. At full spread, you hear each processing stage independently in each speaker. This provides a smooth, continuous transition from a stacked series chain to a wide stereo effect.

Split-Stereo Mode (Unlinked)

Click the chain-link button to switch to unlinked mode. In this mode, the two engines process the left and right channels of your input independently — Engine L handles the left channel, Engine R handles the right. This is ideal for preserving the stereo image of wide sources like stereo synth

pads, drum buses, or any material where left and right carry different content. Each channel gets its own independent effect processing with no crossfeed between them.

Single Engine Mode

To use only one engine, set the unused unit to Program 00 (Defeat) or bypass it with the power button. The signal will pass through that stage unchanged, and the MDV-II behaves like a single-engine processor.

Resampling and Bandwidth

The original hardware runs at a fixed internal sample rate, which limits the audio bandwidth to approximately 10 kHz. The plugin replicates this by downsampling the input to the internal rate before processing and upsampling the output back to the host DAW's sample rate using windowed sinc interpolation. The input signal passes through an anti-aliasing filter before downsampling.

This bandwidth limitation is an essential part of the MidiVerb II's sonic identity. The gentle high-frequency rolloff gives reverb tails a natural warmth and prevents the harshness that can occur with full-bandwidth digital reverb at short delay times.

Stereo Signal Flow

The processing engines accept a mono input (stereo inputs are summed) and produce stereo output — each program reads different delay taps for the left and right channels, creating its own stereo image.

The global Mix knob blends this processed stereo output with the original stereo input at the final output stage. This means your original stereo image is always preserved in the dry signal, regardless of the mono summing that happens inside the engines.

Modulation (Flange and Chorus Programs)

Programs 50–69 add low-frequency oscillator modulation to the delay tap positions, producing flanging and chorus effects.

- **Flange programs (50–59)** use triangle-wave LFOs for the sharp, symmetric sweep characteristic of classic analog flanging.
- **Chorus programs (60–69)** use sine-wave LFOs implemented as a coupled-form oscillator, producing smoother, more subtle modulation.

The LFOs update at approximately 2,930 Hz (every 8 audio samples), which introduces a subtle stepping quality to the modulation. This stepped character is part of the original sound and is preserved in the plugin.

Getting Started

Installation

The MV-II is available in the following plugin formats:

- **VST3** — Windows and macOS
- **Audio Unit (AU)** — macOS
- **AAX** — Pro Tools on Windows and macOS

Install the plugin by running the provided installer for your platform. After installation, rescan your plugin directory in your DAW if the MV-II does not appear automatically.

Quick Start

1. Insert the MV-II on an auxiliary bus or directly on a track.
2. Set the **Mix** knob to 50% (12 o'clock) for direct insertion, or 100% (fully clockwise) when using a send/return configuration.
3. Set the **Input** knob to approximately 75% as a starting point.
4. Select a program by pressing two digits on the numeric keypad (e.g., **1** then **9** for program 19).
5. Adjust the **Output** knob to match levels with your session.
6. Browse programs using the **Up/Down** arrows or by entering program numbers directly.

Using on an Auxiliary Bus

When the MV-II is placed on an auxiliary bus fed by one or more send controls, set the **Mix** knob fully clockwise (100% wet). The dry/wet balance is then managed by the send levels from your individual tracks. This is the most flexible approach and allows multiple tracks to share a single instance of the plugin.

Using as a Direct Insert

When inserted directly on a track, set the **Mix** knob to taste — typically around 50% for reverb programs, and exactly 50% for flanging and chorus programs where the interaction between dry and wet signals creates the effect. The dry signal passes through the plugin unchanged and is blended with the processed output according to the Mix setting.

Tip: Flanging and chorus effects require a precise blend of dry and wet signal. For the fullest effect, keep the Mix at 50%. More subtle effects can be achieved by reducing the wet proportion.

Controls

Dual Engine Overview

The MDV-II runs two independent processing engines that can operate in two modes, controlled by the Link button:

Linked Mode (default)

In linked mode, the two engines (Unit A and Unit B) run in series. Audio flows through Unit A first, then Unit A's output feeds into Unit B. Each unit can run any of the 100 factory programs independently, giving you the ability to stack and layer effects — for example, a reverse reverb into a chorus, or a gated reverb into a delay. The Pan knobs control stereo spread in this mode.

Unlinked Mode (Split Stereo)

In unlinked mode, the two engines process the left and right channels of your input independently. Engine L processes only the left channel, and Engine R processes only the right channel. This preserves the stereo image of your source material through the effect processing — ideal for stereo synths, drum buses, or any source where spatial separation matters. The Pan knobs are disabled in this mode since each engine is assigned to a fixed channel.

The A/B (or L/R) buttons, Send knob, and global Mix knob let you control each unit independently in both modes.

Knobs

All knobs can be adjusted by clicking and dragging vertically. Double-click any knob to reset it to its default value.

Mix (0–100%, default 25%)

The global dry/wet control. This knob blends between your original stereo input (dry) and the processed output from the dual engine chain (wet). At 0% you hear only the original signal; at 100% you hear only the processed effect.

Because this mix happens at the final output stage, your original stereo image is preserved in the dry signal regardless of the mix setting.

- **Auxiliary bus:** Set to 100% and control the blend from your DAW's send levels.
- **Direct insert:** Start around 25–50% for most effects. Flanging and chorus programs specifically require a 50/50 blend to produce their characteristic sound.

Input Level (0–100%, default 50%)

Adjusts the gain applied to the incoming signal before it enters the processing engines. Use this control to manage the level hitting the emulation. Driving the input too hard will cause digital clipping within the algorithm. If you hear distortion — particularly with programs that use heavy recirculation — reduce the Input level.

Tip: Watch the OVLD LED on the faceplate. If it lights up frequently, back off the Input knob to reduce clipping.

Output Level (0–200%, default 100%)

Sets the gain of the final output signal after the dry/wet mix. This control ranges from silence to double the unity gain, allowing you to compensate for level differences between programs or match the plugin's output to your session levels.

Send (0–100%, default 100%)

Controls the effect send level for the currently selected unit (A or B). This determines how much of the wet (processed) signal each engine contributes to the chain versus passing the dry signal through.

At 100% (the default), the engine's full wet output is used. At 50%, you get a 50/50 blend of the input and the engine's processed output within that stage of the chain. At 0%, the engine is effectively bypassed — the input passes through unchanged.

Tip: In most cases, leave the Send at 100% and use the global Mix knob to control your dry/wet balance. The Send knob is useful for advanced scenarios where you want to partially blend an effect within the chain before it reaches the next unit.

Pan (0–100%, default 50% center) — Linked Mode Only

Controls the stereo position of the currently selected unit (A or B) in linked mode. When both units are panned to center, the signal runs as a pure series chain — you hear the combined result of A feeding into B.

As you spread the pans apart, the two processing stages separate in the stereo field. For example, panning A left and B right lets you hear each effect independently in each speaker. The transition from series to spread is smooth and continuous.

- **Both centered:** Pure series chain. Only Unit B's output is heard (since it processes A's output).
- **Spread apart:** Unit A's output fades into one side, Unit B's output into the other. The wider the spread, the more you hear each stage independently.

In unlinked mode, the Pan knobs are dimmed and disabled since each engine is assigned to a fixed stereo channel (L or R).

A/B (L/R) Buttons

Two buttons on the left side of the main panel select which unit you are currently editing. The active unit is highlighted in orange. Switching units changes which program the seven-segment display shows and which unit the Send and Pan knobs control.

In linked mode, the buttons are labeled **A** and **B** (series chain). In unlinked mode, they change to **L** and **R** (left and right channels).

Both units are always running regardless of which one is selected for editing. These buttons only affect the display and knob routing — they do not mute or bypass either unit.

Link / Unlink Button

The small chain-link icon button to the left of the A/B buttons toggles between linked and unlinked mode.

- **Linked (orange icon):** The engines run in series — A feeds into B. The input is summed to mono before processing, and the Pan knobs control stereo spread. This mode is ideal for stacking and layering effects.
- **Unlinked (dim icon):** The engines run in parallel on separate channels — L processes the left input, R processes the right input. The Pan knobs are disabled. This mode preserves the stereo image of your source through the effect processing.

This setting is saved with your DAW session.

Swap Button

The small button to the right of the A and B buttons swaps all settings between the two units — programs, send levels, pans, and bypass states are exchanged. This reverses the processing order, since audio always flows from A into B.

Per-Unit Bypass (Power Button)

The power icon button in the top-right corner of the main panel toggles bypass for the currently selected unit (A or B). When a unit is bypassed, its engine is disabled and the signal passes through that stage unchanged.

- **Orange icon:** Unit is active (processing).
- **Dim icon:** Unit is bypassed (passing through).

When Unit A is bypassed, the dry input goes directly into Unit B. When Unit B is bypassed, Unit A's output passes through to the final mix. When both are bypassed, the processed signal is silent and only the dry signal (controlled by the global Mix knob) is heard.

SIGNAL and OVLD LEDs

The two LEDs on the faceplate provide a quick visual indication of the plugin's output level:

- **SIGNAL (green):** Lights when audio is present at the output (above -30 dB). When this LED is dark during playback, check that the Input level is turned up and the Mix is not at 0%.
- **OVLD (red):** Lights when the output level reaches 0 dB (clipping). If this LED stays lit, reduce the Input Level, Output Level, or both.

The meter reads the signal after the dry/wet mix and output gain — it shows you exactly what the plugin is sending to your DAW.

Seven-Segment Display

The two-digit LED display shows the current program number (00–99) for the active unit (A or B). During program entry, it shows the digits as you type them.

You can click and drag vertically on the display to scroll through programs continuously. Dragging up advances the program number; dragging down decreases it. The display wraps around from 99 to 00 and vice versa.

Program Name

The text label shows the name of the currently active program for the selected unit (for example, “Medium Bright 1.9 Sec” or “Triggered Flange”). This label updates immediately when you change programs or switch between units.

Button Panel

The button panel is arranged in two rows of six buttons each. The layout is as follows:

Top row: Up arrow, 0, 1, 2, 3, 4

Bottom row: Down arrow, 5, 6, 7, 8, 9

Category labels are printed above and below the numeric buttons to show which digit corresponds to which effect category:

- **Top row labels:** REVERB (0, 1, 2), GATE (3), REVERSE (4)
- **Bottom row labels:** FLANGE (5), CHORUS (6), DELAY (7, 8), EFX (9)

These labels correspond to the tens digit of the program number. For example, pressing 3 then 5 selects program 35 — a “Fast Gate” program in the Gate category. Program changes apply to whichever unit (A or B) is currently selected.

Numeric Buttons (0–9)

The ten numeric buttons are used to select programs by number.

Program entry is a two-step process:

1. Press a digit to enter the tens place. The display shows the partial number (e.g., pressing 4 shows “40”) and the ones digit blinks for approximately 4 seconds.
2. Press a second digit to complete the selection. The program changes immediately.
3. If no second digit is entered within the timeout, the display reverts to the previously active program. No change is made.

If you are mid-entry (ones digit blinking) and press the Up or Down arrow, the partial entry is cancelled and the arrow navigation takes effect instead.

Up and Down Arrows

The **Up** arrow (top-left button, marked with a triangle) advances to the next program. The **Down** arrow (bottom-left button) goes to the previous program. Programs wrap around: pressing Up on program 99 goes to 00, and pressing Down on 00 goes to 99.

Toolbar

The toolbar is a thin strip above the main faceplate. On the left side is the Vintage Mode toggle; on the right side are two icon buttons:

- **Vintage Mode button** (valve/tube icon): Toggles vintage output filtering on and off. The icon glows amber when vintage mode is active. See below for details.
- **Help button** (question mark icon): Opens this user guide.
- **About button** (info icon): Opens the About panel, which shows the plugin version, a link to the Temecula DSP website, and display scaling options.

Vintage Mode

The valve/tube icon in the toolbar toggles between two output filter modes. When the icon glows amber, vintage mode is active. When dimmed, the clean anti-alias filter is active instead. This setting is saved with your DAW session on a per-instance basis — each track can have its own setting.

Vintage Mode On

The wet signal passes through a three-stage analog reconstruction filter chain modeled after the original hardware output path:

1. **Passive RC low-pass filter** at approximately 7,204 Hz. This is the post-DAC smoothing filter from the original hardware and is the primary source of the characteristic dark, warm tone.
2. **Sallen-Key low-pass filter #1** at approximately 15,621 Hz with $Q = 2.27$. This is a resonant stage that adds a subtle peak around 15 kHz before rolling off.
3. **Sallen-Key low-pass filter #2** at approximately 15,000 Hz with $Q = 0.59$. This is a damped stage that provides additional high-frequency attenuation without resonance.

Together, these three stages produce the warm, band-limited sound characteristic of the original unit. The effective audio bandwidth in vintage mode is roughly 20 Hz to 10 kHz.

Vintage Mode Off (default)

The vintage reconstruction filters are bypassed and replaced with a clean 4th-order Butterworth low-pass filter at 11,000 Hz. This filter is implemented as two cascaded second-order sections ($Q = 0.5412$ and $Q = 1.3066$) and provides anti-aliasing with a maximally flat passband — no resonant peaks, no coloration.

The result is a brighter, more open sound that preserves more high-frequency content from the reverb engine while still preventing aliasing artifacts.

Note: In both modes, the input stage DC blocker (1.6 Hz), gain stage high-pass (7.2 Hz), and output DC blocker (5 Hz) remain active. These filters are below the audible range and serve to remove DC offset — they do not affect the tonal character of the signal.

Display Scaling

The About panel includes a **Display** section with three scaling options: **100%**, **150%**, and **200%**. Click any option to resize the entire plugin interface. The selected scale is saved as a preference and persists across sessions.

Program 00 — Defeat

Program 00 (Defeat) is a special pass-through program. When selected on either unit, the signal passes through that stage unchanged. This is useful when you only want to use one engine — set the unused unit to program 00 and it will have no effect on the signal.

Programs

The MV-II contains 100 factory programs organized into seven categories. Each program is a unique configuration of the DASP-16 delay-line engine, with carefully chosen tap positions, feedback paths, and gain structures that produce a wide range of effects from a single architecture.

Program 00 — Defeat

Program 00 silences the effect output, functioning as a convenient mute for the wet signal. The dry signal (if the Mix knob is not at 100%) continues to pass through unaffected.

Programs 01–29 — Reverb

Twenty-nine reverb programs spanning a wide range of simulated acoustic environments. These programs vary in:

- **Size** — from tight, small-room ambiances to expansive concert-hall decays
- **Tonal character** — bright, warm, dark, and neutral voicings achieved through the specific arrangement of delay taps
- **Decay time** — ranging from short early-reflection patterns under one second to long, sustained tails exceeding ten seconds
- **Pre-delay** — the gap between the dry signal and the onset of the reverb, giving the ear spatial cues about room size
- **Stereo image** — different programs distribute their output taps across the stereo field in distinct patterns

Shorter, tighter programs work well on drums and percussive instruments where clarity and definition are important. The larger, longer programs suit vocals, lead instruments, and sustained pads where a sense of space enhances the performance.

Programs 30–39 — Gate

Ten gated reverb programs that cut off the reverb tail abruptly rather than allowing it to decay naturally. Gated reverb became one of the defining production sounds of the 1980s, widely used on snare drums and toms.

- **Programs 30–34** — Soft gate: the reverb tail tapers briefly before closing, giving a slightly less abrupt cutoff.
- **Programs 35–39** — Hard gate: the tail shuts down quickly and cleanly with no taper.

Each program varies in tonal character and the length of the gated burst. Experiment with different programs on different drum sounds — the interaction between the source material’s envelope and the gate length is what makes each combination unique.

Programs 40–49 — Reverse

Ten reverse reverb programs that invert the natural decay envelope, producing a tail that swells up rather than fading away. This effect was originally achieved in analog studios by physically flipping tape reels — the MidiVerb II recreated it digitally.

- **Programs 40–44** — Standard reverse reverbs with varying tonal response and swell times.
- **Programs 45 & 49** — “Bloom” effects: the signal rises into a highly diffused, dense reverb cloud before decaying smoothly. These produce ethereal, atmospheric textures well suited to ambient and slow-tempo material.
- **Programs 47 & 48** — Regenerated reverse: the reverse envelope feeds back into itself, creating trailing echoes after the initial swell. Effective on vocal phrases and lead instrument accents.

Programs 50–59 — Flange

Ten flanging programs that use triangle-wave LFO modulation to sweep delay tap positions, creating the characteristic comb-filter sweep of analog flanging. The MV-II's flanging programs also include stereo panning effects of varying speed and width.

Triggered Flange (Programs 50, 53, and 57)

Three of the flange programs use a triggered mode where the LFO resets to the top of its cycle each time the input signal crosses an internal volume threshold. This produces a dramatic, attack-synchronized flanging sweep that restarts with each transient.

Triggered flanging works best with percussive, sharply attacked signals — drums, cymbal crashes, muted rhythm guitar, and similar sources. Instruments with smooth, sustained envelopes may cause the trigger to fire during the sustain portion, producing an audible thump as the delay resets. This is inherent to the triggered mechanism, not a defect.

Tip: If you hear unwanted thumping on sustained sources, ensure the Input level is set so the SIGNAL LED stays consistently lit during the sustained portion. This keeps the signal above the trigger threshold and prevents false retriggering.

Standard Flange (All Other Programs in 50–59)

The remaining flange programs use continuously running LFOs that sweep at different rates and depths. Many include automatic stereo panning that moves the flanged signal across the stereo field.

To hear the deepest flanging effect, set the Mix to exactly 50%. The interaction between the dry and delayed signals is what creates the notches and peaks in the frequency spectrum — reducing the wet level produces a more subtle effect, while removing the dry signal entirely reveals the panning motion without the characteristic comb filtering.

Programs 60–69 — Chorus

Ten chorus programs using sine-wave LFO modulation for a smoother, more subtle pitch variation than the triangle-wave flanging programs. Chorus ranges from gentle, slow-rolling modulation with minimal pitch shift to deep, multi-voice effects with pronounced movement.

The chorus programs are effective across a wide range of sources. Light settings add warmth and subtle motion to keyboards and clean guitars; deeper settings produce the lush, shimmering sound associated with 1980s production. The MV-II's bandwidth-limited processing gives its chorus a naturally warm quality without the need for additional filtering.

Programs 70–89 — Delay

Twenty simple delay programs with tap times ranging from approximately 35 ms to 460 ms. These cover a variety of common delay applications:

- **Short delays (35–80 ms)** — Doubling and thickening effects. At these times the delayed signal fuses perceptually with the original, adding body and width without being heard as a distinct echo.

- **Medium delays (80–150 ms)** — Slap-back echo territory. These produce the classic rockabilly and vocal doubling effect heard on countless recordings.
- **Long delays (150–460 ms)** — Distinct trailing echoes suited to lead vocals and solo instruments. These create rhythmic repetitions that can be matched to the tempo of a song by selecting the appropriate program.

Programs 90–99 — EFX (Special Effects)

Ten specialized production effects that showcase the creative possibilities of the DASP-16 architecture:

- **Programs 90–91** — Two-tap delay with a light stereo ambience layered on top, combining rhythmic echo with spatial depth.
- **Program 92** — Three-tap delay with automatic stereo panning, creating rhythmic echoes that move across the stereo field.
- **Program 93** — Multi-tap reverb that generates multiple voices from a single mono input, useful for thickening sparse arrangements.
- **Program 94** — Multi-tap pan: a rapid cascade of delay taps sweeping across the stereo field, producing a dense, shimmering wash.
- **Program 95** — Thickener: uses allpass filters frozen at a fixed modulation point to create the out-of-phase, hollow quality of a static flange. Particularly effective on rhythm guitar and percussive keyboards.
- **Programs 96–97** — Stereo generators that create a wide stereo image from a mono source without adding obvious reverb or modulation artifacts. Useful for spreading mono keyboards, synths, or vocal tracks across the stereo field.
- **Programs 98–99** — Regenerated echoes with varying feedback amounts and decay times. The regeneration is remarkably clean, producing long trails of repeating echoes that fade gracefully without becoming muddy.

Usage Guide

Choosing the Right Program

With 100 programs available, finding the right effect can feel overwhelming at first. The programs are grouped into logical categories, so start by identifying what type of effect you need, then browse within that range:

Category	Programs	Best For
Reverb	01–29	Ambience, space, sustain on any source
Gate	30–39	Drums, percussive sources, rhythmic effects
Reverse	40–49	Atmospheric swells, ethereal vocals, transitions
Flange	50–59	Guitars, drums, creative motion effects
Chorus	60–69	Keyboards, clean guitar, vocals, thickening
Delay	70–89	Rhythmic echoes, doubling, slap-back
EFX	90–99	Stereo widening, multi-tap textures, production tricks

Working with Reverb Programs

The 29 reverb programs cover a broad range, from tight room ambiences to long, diffused halls. A few guidelines for selecting and mixing reverbs:

- **Drums and percussion** — Start with shorter programs (01–10). These provide enough spatial context to place the drums in a room without washing out transient detail.
- **Vocals** — Medium to large reverbs (10–20) add presence and depth. Use the Mix control to keep the vocal intelligible while the reverb fills in behind it.
- **Pads and sustained instruments** — Longer reverbs (20–29) complement sustained sounds. These programs have extended decay times that blend naturally with long notes and chords.

Tip: When comparing reverb programs, keep the Mix at 50% so you hear the full character of each effect. Once you've chosen a program, adjust the Mix to suit the track.

Getting the Most from Gated Reverb

Gated reverb is highly dependent on the source material's dynamics. The effect is most dramatic on sounds with a strong transient attack followed by a quick decay — snare drums, hand claps, and toms are classic applications. For the tightest gate response, use programs 35–39. For a slightly more natural feel, try programs 30–34 which allow a brief tail before closing.

Flanging and Mix Settings

Flanging relies on the phase interaction between the dry and delayed signals. The depth of the effect is directly related to the balance between these two components:

- **50% Mix** — Maximum flange depth. This is the textbook setting for the most dramatic comb-filter sweep.
- **Below 50%** — Reduced flange depth with more dry signal. Useful for adding a subtle shimmer without the full sweep.
- **100% Mix (wet only)** — The comb filtering disappears entirely, but the stereo panning effect becomes prominent. This can be used creatively as a standalone auto-panner.

The stereo panning built into many of the flange programs is best heard on headphones or a well-positioned stereo monitoring setup. Feed a continuous percussive source (muted guitar, simple drum loop) through the plugin with the Mix at 100% to clearly hear the panning motion.

Chorus and Subtlety

Chorus effects are at their best when they enhance without calling attention to themselves. Start with the lighter programs (60–63) and increase to the deeper programs only when you want obvious modulation. The MV-II's natural bandwidth rolloff helps chorus sit well in a mix without competing with the brightness of the dry signal.

Creative Uses for the EFX Programs

The ten EFX programs (90–99) are some of the most distinctive sounds in the MV-II. A few applications worth exploring:

- **Program 95 (Thickener)** on rhythm guitar adds the hollow, metallic edge of a cocked wah pedal without any frequency sweep. Blend to taste with the Mix control.
- **Programs 96–97 (Stereo Generator)** are invaluable for creating width from mono sources. Insert directly on a mono track and the output will have a convincing stereo spread with no obvious processing artifacts.
- **Programs 98–99 (Regenerated Echo)** produce self-sustaining echo trails that are cleaner than what most dedicated delay plugins achieve. Use them for ambient soundscapes or as a creative performance tool by feeding signal in bursts and letting the echoes develop.

Specifications

Processing Engine

Attribute	Value
Architecture	DASP-16 delay-line emulation
Internal Sample Rate	31,250 Hz
Internal Resolution	16-bit (processing)
Audio Bandwidth	20 Hz – ~10 kHz (vintage) / ~11 kHz (clean)
Delay Buffer	16,384 samples (~700 ms maximum delay)
Programs	100 factory presets (0–99)
Instructions per Sample	128

Effect Categories

Category	Programs	Count
Defeat	00	1
Reverb	01–29	29
Gate	30–39	10
Reverse	40–49	10
Flange	50–59	10
Chorus	60–69	10
Delay	70–89	20
EFX	90–99	10

Plugin

Attribute	Value
Formats	VST3, Audio Unit (AU), AAX
Platforms	macOS, Windows
GUI Size	896 x 131 pixels
Host Sample Rates	Any (internal resampling handled automatically)

Credits

Original MidiVerb II hardware (1986): Keith Barr and Alesis Corporation, Los Angeles, California.

DASP-16 custom DSP chip: Designed by Keith Barr at Alesis. Barr's pioneering work on the DASP-16 made affordable digital reverb a reality. His single-chip design — combining a delay-line processor, program sequencer, and DAC interface — brought studio-quality effects to musicians at a fraction of the cost of existing units. The MidiVerb II and its successors went on to become some of the best-selling effects processors in recording history.

Plugin: Temecula DSP

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