ADD-ON VINTAGE STOMP PACKAGE (AE-051)



This package includes a number of superrealistic recreations of vintage guitar stompbox effects that are highly values for their rich, warm sound. VCM technology brings these outstanding effects back to life with greater controllability and flexibility than ever!

MAX100

Born in the late '70's, this phaser is still available in reissue form. There are many who believe the original '70's models sounded better than the current models, and so the K's Lab team have painstakingly modeled the original circuit and components. Even the original light sensitive CdS cell that was used for modulation has been modeled so the subtle change in modulation character with modulation speed of the original is recreated in perfect detail.



DE: Selects four different combinations of modulation depth and feedback

ED: Sets the modulation speed. The character of the modulation changes with modulation speed in a truly musical manner.

DUAL PHASE

Like the MAX100, there are many guitarists who will go to any lengths to get an original version of this stomp box to enhance their sound. This is a faithful reproduction of the original with dual phaser circuits and dual LFOs that can be configured to deliver a dazzling array of effects. Special care has been taken in modeling the effect of the CdS cell in the phase-shifting circuit so that the exquisite balance at all modulation speeds that was a major part of the sound of the original has been retained.



RATOR 1 SWEEP GENERATOR 1 RATE: Sets the speed of LFO 1. 2 SHAPE: Sets the waveform of LFO 1 TE: Sets the speed of LFO 2. 4 SHAPE: Sets the waveform of LFO 2. PHASER A 5 DEPTH: Sets the modulation depth of PHASER A. ACK: Sets the amount of feedback for PHASER A. 6 F **7** ON/OFF: Turns PHASER A ON or OFF. PHASER B 8 DEPTH: Sets the modulation depth of PHASER B. 9 F K: Sets the amount of feedback for PHASER B. Turns PHASER B ON or OFF. P: Selects the LFO to be used for PHASER B modulation **2 SWEEP SYNC:** Selects the polarity of PHASER B modulation. MODE: Selects the input for PHASER B, as well as the overall connection configuration (details below).



VINTAGE PHASER

Rather than a simulation of a specific phaser, this model has been designed to deliver the best qualities of the most sought-after classic phasers in one versatile effect. Different mode settings transform this effect into dramatically different phaser types. Stereo and mono versions are provided.



: Sets the modulation speed. 1 SF Sets the phaser's operation point. H: Sets the depth of the effect. BACK: Adjusts the amount of feedback. OR: Sets the frequency of the fixed phase shift filter, thus changing the overall tonal character of the sound.

WODE Two different operation modes can be selected for each stage. STAGE Specifies the number of phase-shift stages used. Fewer stages producer a simplement while more stages create a more complex sound.







HANNEL STRIP

SURROUND POST

000000

I I HI

CHANNEL STRIP PAGKAGE (AE-011) MASTER STRIP PAGKAGE (AE-021) REVERB PAGKAGE (AE-031) SURROUND POST PACKAGE (AE-041) VINTAGE STOMP PAGKAGE (AE-051)

30066

VINTAGESTOMP

All specifications are subject to change without no All trade ed trademarks are property of their respective owner

www.yamahaproaudio.com



LPA496 (P10017302) @Printed in Japan

for DM 2000 version 2/O2R 96 version 2/DM 1000 version 2/O1V 96 version 2/ PM5D

Beyond Simulation

While many digital "simulations" of classic analog circuits and effects merely scratch the surface, Yamaha's Add-on Effects offer realism that extends to every sonic nuance and detail. These valuable signal-processing plug-ins that owe their awesome processing power to the 96-kHz audio DSP capabilities of Yamaha digital mixing consoles, and can be used with DM/0 series digital mixing consoles as well as the PM5D.* When used with Studio-Manager V2 software, the Add-On Effects provide an efficient, intuitive graphical interface that makes operation a simple pleasure.

Yamaha Add-on Effects are based on innovative Virtual Circuit Modeling technology that, rather than simply attempting to approach the desired sound using conventional digital audio methods, actually models the original analog circuitry ... right down to the last resistor and capacitor. Also available is a package of spatial effects that deliver unprecedented control for spatial surround processing using innovative Yamaha iSSP (Interactive Spatial Sound Processing) technology.

The Add-On Effects are capable of capturing subtleties that simple digital simulations cannot even approach, in effect going beyond simple simulation and delivering the truly musical performance that makes classic analog gear valuable assets even in today's digital production environment.

*The Add-on Effects require digital console operating system Version 2 (except PM5D)

Package/Console Compatibility

	OIV 96 Version 2	02R 96 Version	2 DM 1000 Version 2	DM 2000 Version 2	PM5D DIGITAL MIXING CONSOL
AE-011 (CHANNEL STRIP)	Yes	Yes	Yes	Yes	Yes
AE-021 (MASTER STRIP)	Yes	Yes	Yes	Yes	Yes
AE-031 (REVERB)	Yes	Yes	Yes	Yes	Yes*
AE-041 (SURROUND POST)	No	Yes	Yes	Yes	No
AE-051 (VINTAGE STOMP)	Yes	Yes	Yes	Yes	No

*The AE031 Reverb Package is supplied with the PM5D

Disclaimer

The names of programs or menus incorporated in Add-On Effects are for descriptive purpose only. Reference to product names, trademarks, artists and songs is made for the sole purpose of identifying products and sounds studied for modeling and describing the sound nuances Yamaha attempted to create through use of its proprietary technology. Such reference does not constitute representations that they physically possess equal qualities, and does not imply any cooperation or endorsement by such manufacturers or artists. The products, trademarks are the property of their respective owners.

The Team and the Technology Behind the Sound





importance. The goal? In a word, "musicality." The K's Lab team were aware that the earliest effect modeling technologies were focused more on superficial reproduction of specific characteristics and tonalities than on actually making music, and it was clear that by applying the same physical modeling technology that was used in the original VL1 and VP1 synthesizers, although in a significantly more evolved form, it would be possible to deliver truly accurate, eminently musical effects. And rather than relying on frequency response graphs and other "precision" measurements to evaluate final performance, many critical performance decisions were made using the trained ears of top-level music and sound specialists.

The Birth of VCM

It took more than two years of concentrated work, but by 2003 K's Lab had refined and re-purposed physical modeling to the point where it was ready for practical implementation ... in the form of Virtual Circuit Modeling. VCM is the cornerstone of Yamaha's Add-On Effects, and achieves it's stunning sonic and musical performance by actually modeling the individual characteristics of the multitude of parts and components that contributed to the final sound of the original analog circuits: transistors, tape, tape heads, etc. Even subtle saturation effects have been painstakingly modeled to bring the warmth and richness of the original analog gear back to life in stable, easy-to-operate digital form.

Making Space



simulation available anywhere:

- decay of the reflections based on source directivity and room surface materials.
- matrix channel, and simulates distance-related decay through delay and filter processing.

"Modeling is a means to an end, not the final goal." Mr. Toshifumi Kunimoto, the central figure of Yamaha's physical modeling technology team, has a fine track record when it comes to meeting some very challenging goals. The division known at Yamaha as "K's Lab" ("K" for "Kunimoto") was established in 1987 to develop new modeling technology that would become the next phase in synthesizer evolution after the FM and PCM tone generators that were the mainstay of the synthesizer world at the time. The result was the world's first physical modeling synthesizers – the VL1 and VP1 – released in 1993. Research and development has continued relentlessly ever since, and in 2001 the K's Lab team began aiming it's formidable technological capabilities at physical modeling for effects, and that's when Mr. Kunimoto's goal began to take on primary

A new addition to Yamaha's powerful Add-On Effect arsenal is iSSP (Interactive Spatial Sound Processing). This innovative effect takes surround sound to new levels of reality and creative control. iSSP is actually a combination of two advanced modeling technologies that add up to the most realistic spatial

• Room acoustics modeling that both predicts sound reflection patterns based on room shape, and actually models the

• Matrix sound processing that converts source position data to parameters that precisely control the output of each

ADD-ON EFFECTS **CHANNEL STRIP** PAGKAGE (AE-011)



The AE-011 Channel Strip Package includes 5 models that employ VCM (Virtual Circuitry Modeling) technology to recreate the sound and characteristics of several classic compression and EQ units from the 70's. Not only do these models faithfully capture the unique saturation of analog circuitry – in part thanks to precise modeling of the original FET gain reduction. Tube/Transformer buffer amplifier, VCA (Voltage Controlled Amplifier) and RMS level detection

circuitry – but they have also been fine-tuned by leading engineers and feature carefully selected parameters in a simple interface that makes it easier than ever to create the ideal sound.

Compressor 276 (mono), Compressor 276S (stereo)

These models recreate the fast response, frequency characteristics, and tube-amp saturation of the most in-demand analog compressors for studio use, delivering classic-style compression with all the punch and fatness you'd expect from a fine piece of studio-grade analog gear. Not limited to processing drums and bass, these compressors are also an excellent choice for vocals and master stereo mix compression. The 276 is a dual mono unit, while the 276S operates in stereo.



- according to the INPUT LEVEL, COMPRESSION RATIO and ATTACK/RELEASE TIME settings.
- **5** Sidechain HPF: Inserts a sidechain high-pass filter with a relatively high cutoff frequency. This limits compression on the low frequencies allowing more bottom end to come through with no change in peak level. 6 Attack Time
- 7 Release Time
- 8 Master Select: Selects the function of the meter display: GAIN REDUCTION/ -20dBFS/-26dBFS/OFF.

Compressor 260 (mono), Compressor 260S (stereo)

Featuring faithful modeling of the solid-state voltage-controlled amplifier and RMS level detection circuitry of the late 70's, these Add-on Effects bring back the sound of classic comp / limiters used primarily for live sound reinforcement applications. They offer three selectable compression knee types – hard, medium, and soft – and although variable attack and release are provided, presets recreate the fixed settings of the vintage gear. Top-level sound-reinforcement engineers have carefully tweaked the parameters of optimum response in live situations. The 260 is a dual mono unit, while the 260S operates in stereo.

KNEE GR 0

- 2 Link: Turns L & R channel linking ON (for stereo operation) or OFF.
- 3 Knee: Provides a choice of three compression knee curves: Hard (LED OFF), Medium (LED RED), or Soft (LED YELLOW)
- 4 Attack Time: Adjusts the compressor's attack time.
- 5 Release Time: Adjusts the compressor's release time.

6 Ratio

7 Output: Allows the output level to be adjusts to compensate for the degree of compression set by the threshold level control.

Equalizer 601

The 601 equalizer offers two equalizer types – Clean and Drive. The Drive type models the distortion characteristics of 70's analog EQ circuitry, delivering musical-sounding drive and saturation. The 601 is a stereo sixband parametric equalizer with LO and HI shelving filters and four MID peaking filters, and it accurately reproduces both the boost and cut frequency response and band interaction of vintage analog gear. And you get EQ capability over a wide 16 Hz ~ 40 kHz range when operating at 88.2 / 96 kHz. The 601 features a familiar knob-style interface as well as graphical editing capability on both the console and PC displays.



1 Input/Output: Adjusts the equalizer's input and output levels. When TYPE is set to "Drive" the equalizer can be driven into distortion by setting a high input level and a relatively low output level.

2 Type: Selects "Clean" (no distortion) or "Drive"

(internal distortion) operation.

O ATTRONT

- 3 Q: Sets the Q (bandwidth) of each band. The rightmost band can be set to LPF or HSH slope, and the leftmost band can be set to HPF or LSH slope.
- These parameters can also be set via the frequency response window.
- 4 Freq.: Sets the center or cutoff frequency of each band. These parameters can also be set via the frequency response window.
- 5 Gain: Sets the gain of each band. These parameters can also be set via the frequency response window.
- 6 Band Bypass: Turns equalization of the individual bands ON or OFF.
- **7** Flat: A quick click resets the gain of the corresponding band to "0". When held for longer than 0.8 seconds all parameters other than TYPE are reset to their initial values.

ADD-ON MASTER STRIP PAGKAGE (AE-021)



🔀 (³m 🕢

The AE-021 Master Strip Package Open Deck employs Virtual Circuitry Modeling technology to recreate both the analog circuitry and tape characteristics that shaped the sound of openreel tape recorders. Because of their ability to smooth out peak levels and tidy up the response, many high-end recording studios

still maintain open-reel recorders such as the Studer A80mk1, A80mk4 and A820, and the Ampex ATR100 and others from the 70's and

80's to be used to provide tape compression at the mastering stage.















Different types of tape – new BASF, old Ampex, etc. – are also selected and used according to the unique sounds they produce. The Open Deck provides models of four machine types: Swiss '70 Swiss '78, Swiss '85, and America '70. You can even combine different record and playback decks for a wider range of variation. You also have a choice of "old" and "new" tape types, tape speed, bias, and EQ settings that can vary the "focus" of the sound, distortion, and saturation characteristics.

Now you can easily take advantage of top-end sound-shaping techniques in real time using Yamaha digital consoles.

- Selects the type of record deck.
- Selects the type of playback deck
- Selects metering of the record or repro deck
- When ON the repro level is linked to the record level.
- The record deck high equalizer. Can be used to produce more high-frequency saturation.
- Adjusts the bias of the recording deck. The range is from -1.00through +1.00.
- The repro deck equalizer.
- Sets the output level of the repro deck.





ADD-ON EFFECTS REVERB PAGKAGE (AE-031)



REV-X (HALL)

These reverb Add-on Effects employ the latest "REV-X" algorithms first introduced in Yamaha's SPX2000 Professional Multi Effect Processor. The REV-X programs feature the richest reverberation and smoothest decay available, based on years of dedicated research and development. REV-X Hall, REV-X Room, and REV-X Plate programs are provided, with new parameters

Low Freq

45 - 1

Reverb Time [2.98 s]

MIX 0

such as room size and decay envelopes that offer unprecedented definition and finer nuance control. The REV-X Hall and REV-X Room programs have a very open sound, while REV-X Plate delivers a brighter tonality that is ideal for vocals. All models deliver dense. warm reverb that does not interfere with the natural timbre of the source.

Sets the cutoff frequency of the low-pass filter (1.00 ~ 18.0 kHz, Thru). This filter can be used to reduce the higher frequencies of the reverb sound. Sets the cutoff frequency of the high-pass filter (Thru, 22.0 Hz ~ 8.0 kHz,). This filter can be used to reduce the lower frequencies of the reverb sound. Delay: Sets the initial delay from 0.0 through 125.0 milliseconds.

- 4 Decay: Adjusts the shape of the decay envelope.
- e: Adjusts the size of the "space" in which the reverb is occurring. Changing this parameter also affects the reverb time.
- e: Sets the reverb time from 0.10 through 46.92 seconds. This value is also affected by the Room Size setting.

2.7% 0 11.0.V 0 HI ANTIO 0.0 1.2

- 7 Low Freq: Sets the basic Low Ratio frequency.
- 8 Hi Ratio: Sets the ratio between the high-frequency reverb time and the overall Reverb Time.
- Sets the ratio between the low-frequency reverb time and the overall Reverb Time
- : Sets the density and left-right spread of the reverb sound.

ADD-ON SURROUND POST PACKAGE (AE-041)



The three effects in this package take full advantage of Yamaha's remarkable iSSP (Interactive Spatial Sound Processing) technology to deliver precisely-controllable spatial processing capabilities that are particularly suited to cinema or television sound post-production and mixing facilities. All effects are applicable to a range of surround formats, providing unprecedented

Room-ER

Room-ER is capable of simulating the acoustic properties of a room of about 30 meters in length, with accurate reproduction of the direct sound and early reflections as affected by distance from the source, source motion, speed of motion, and room surface characteristics. This effect is ideal for placing a mono source in a precisely controllable surround environment.



REV-X (ROOM

OUTPUT L -18 -36 -24 -12





EV-X (PLATE)



Application Ideas

As it's name implies, Room-ER is basically a room simulation that allows a monaural source to be positioned and moved within a simulated room. This can be useful, for example, to process a speaker moving away from the viewer in a movie or video scene. The sonic effects of the speaker moving away, turning to face the viewer, and moving back toward the viewer can be reproduced with remarkable precision. In addition to surround applications, this same effect can also be used to add a sense of depth to stereo music tracks, or to realistically simulate the effect of performers moving around the stage.



 POSITION VIEW: Displays and allows editing of room shape, source position, and listener position.

- 2 MARK: These buttons are used to mark and retain the positions of objects in the POSITION VIEW display. **3** SCALE: Displays and allows editing of the room size parameter.
- 4 SOUND SOURCE: Sets the characteristics of the sound source: position, motion, and directivity. 5 LISTENER: Sets the characteristics of the listener:
- position and motion.
- 6 AIR ABSORPTION: By adjusting the rate of decay caused by air absorption, this parameter simulates the effect of the sound source moving away from the listener (high-frequency decay).
- 7 WALL CHARACTER: Specifies the characteristics of the reflecting surfaces in the room: wall materials and absorption.
- 8 SURROUND MODE: Specifies 3-1, 5.1, or 6.1 surround mode. The surround mode can be switched directly from the digital console. It is also possible to adjust center-speaker divergence (the ratio of signal sent to the center and L/R speakers) via the display.
- CURRENT TIMECODE: Displays the current timecode.
- ¹⁰ AUTOMATION: Determines the motion of the sound source and listener in relation to timecode. Programming is easy: a mouse or user-defined keys can be used to specify the Start Key, Key1, Key2, and End Key timing for the sources.
- PLAYBACK MODE: Specifies the automation playback mode: timecode trigger, manual trigger, and a range of other control options are available: Once: The motion occurs once as specified by the key points and stops. Go&Rev: The motion continues through to the end point and then reverses back to the start point.
- Loop: The motion continues to loop from start point to end point.
- 2 TRANSPORT BAR: Provides the AUTOMATION controls.



Auto Doppler

Perhaps the most common example of the Doppler effect is the change in pitch of an ambulance siren as it moves toward and then away from the listener. Auto Doppler effectively simulates this effect in a wide variety of scenarios. In addition to objects moving linearly past the listener, Auto Doppler can recreate the effect of objects moving toward and then away from the listener, for example, with precise speed and distance control. Timecode automation is also possible



- **1** POSITION VIEW: Displays the motion track of the sound source. Editing can be accomplished by using a mouse
- **2** ZOOM: Zoom control centered on the listening position.
- 3 SPEED: Adjusts the speed at which the sound source moves. Speed can be adjusted from a slow walk to a jet airplane.
- 4 FADE TIME: Adjusts the fade-in and fade-out times.
- 5 DISTANCE: Adjusts the degree of distance decay 6 AIR ABSORPTION: Simulates the effect of highfrequency absorption through air.
- **7** PITCH: Displays and turns the Doppler effect ON or OFF.
- 8 SURROUND MODE: Specifies 3-1, 5.1, or 6.1 surround mode. The surround mode can be switched directly from the digital console. It is also possible to adjust center-speaker divergence (the ratio of signal sent to the center and L/R speakers) via the display.
- 9 CURRENT TIMECODE: Displays the current timecode.
- @ AUTOMATION: Determines the motion of the sound source in relation to timecode. Programming is easy: a mouse or user-defined keys can be used to specify the start point, transit point, and end point timing for the source.
- PLAYBACK MODE: Specifies the automation playback mode: timecode trigger, manual trigger, and a range of other control options are available: Once: The motion occurs once as specified by the key points and stops. Go&Rev: The motion continues through to the end point and then reverses back to the start point

12 TRANSPORT BAR: Provides the AUTOMATION controls.

Application Ideas

Auto Doppler is the ideal effect for simulating motion in a wide variety of situations. Basic point A to point B simulation can be used for the motion of cars crossing a scene, or aircraft taking off or landing at an airport. Point A through point B to point A' simulation is also available,

and could be used, for example, in scene of a race car rounding a hairpin bend on a racecourse. Auto Doppler can simulate listener-to-source distances of up to about 1 kilometer, providing more than enough range for a wide variety of processing applications



Field Rotation

The Field Rotation effect can be used to rotate or distort the sound field around the listener. The listener can be at the center of rotation. or the listener can be rotated or moved around a sound source. The axis of rotation, amount of movement, distance from the center of rotation, and speed of motion can be specified and controlled manually via a joystick like the one provided on the DM2000 console, or automated as required.



1 FIELD SCOPE: The field scope visually displays the positions of the speakers as well as rotation, movement, and shape of the virtual sound field, and allows editing of the displayed parameters.

IN CONTROL PANEL: This control

panel includes the basic Field Rotation controls: RADIUS (initial field radius), R.RATIO (radius ratio), OFFSET (initial center point), DEGREE (amount of rotation), CAV (constant speed while maintaining channel angle), and CLV (constant speed while maintaining constant distance from elliptical path).

3 ALITOMATION CONTRO

Allows programming motion of the virtual

speaker circle, or how the various parameters change along the time axis from the start point

(Key 1) to the end point (Key 2).

Motion can be previewed non-destructively via the Automation Preview Scope prior to actually applying it to the sound.

4 SURROUND MODE: Specifies 3-1, 5.1, or 6.1 surround mode. The surround mode can be switched directly from the digital console. It is also possible to adjust center-speaker divergence (the ratio of signal sent to the center and L/R speakers) via the display

5 TRANSPORT BAR: Provides the AUTOMATION controls

Application Ideas

Any scene that involves rotation is a potential application for this effect. Place the viewer on the coffee-cup ride or carousel at an amusement park, add realistic sound motion to a boomerang in flight, UFOs, propellers .. anything that spins or follows an elliptical path. This effect will undoubtedly find many uses in 3D video games, too.





iSSP

