

2026 - may - 29

FW : Host 1.11 RC - DSP 1.11 / rc 3

Disclaimer: This is an unofficial add-on to the M user manual, describing some of extra features added from firmware 1.05 up to firmware 1.11. This add-on also covers some complex aspects of the Multi Mode, which remains somewhat unusual.

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Algorithms of FW version 1.11:

Varysaw : A VA oscillator ranging from SAW through PULSE to TRI.

Wave (red encoder): Mixes from 0 to 31 from SAW to PULSE; 32 is pure PULSE, and 33 to 63 mixes from PULSE to TRI.

Meta parameter: Pulse width for PULSE, adjustable from 5% to 95%.

CSAW : A famous Yamaha CS-80 saw oscillator with mutation towards a parabolic oscillator.

Wave : Controls the amplitude of the “step” component of the saw wave.

Meta parameter: Mixes between this saw wave and the parabolic saw wave.

Waveshpr : A SAW to TRI oscillator fed into a mild waveshaper.

Wave : 0 is pure SAW, 1-62 mixes between SAW and TRI, and 63 is pure TRI.

Meta parameter: Depth of the waveshaper.

TriplSaw : A stack of three SAW oscillators; two are detuned slightly higher and lower than the third.

Wave : Controls the amplitude of the “central” SAW oscillator relative to the two “side” SAWs.

Meta parameter: Detuning factor, from 0 to maximum detune.

S/T Fold : A SIN wave folding oscillator that folds the chosen crossmixed wave between SIN and TRI waves.

Wave : 0 is pure SIN, 1-62 mixes between SIN and TRI, and 63 is pure TRI.

Meta parameter: Folding depth. π

FM 2 OP : Classic simple two-operator FM synthesis. OP1 (modulator) generates a sine wave that modulates the phase of OP2 (carrier). Both operators generate sinewaves; the algorithm's output is the carrier signal.

Wave: Maps values from 0 to 63 to harmonics ratios of 1 to 8 in stepped increments.

Meta parameter: Amount of phase modulation applied to the carrier.

FM Fback : Classic simple two-operator FM synthesis with feedback. OP1 (modulator) generates a sine wave that modulates the phase of OP2 (carrier). Both operators generate sinewaves; the algorithm's output is the carrier signal, which modulates the modulator's amount in feedback loop.

Wave : Maps values from 0 to 63 to harmonics ratios of 1 to 8 in stepped increments.

Meta parameter: Amount of phase modulation applied to the carrier.

DuoNosie : A white noise source filtered by a pair of 12dB LP/HP filters set to the same (static) cutoff frequency. The cutoff frequency differs between the first and second oscillators of M; it is 2 kHz for OSC1 and 4.4 kHz for the second oscillator. Combined with variable resonance, this allows generating different noise flavors to shape your sound.

Wave : Blends from 0-63 between low-passed (0) and high-passed filters (63).

Meta parameter: Q factor of the filter.

Multi-Mode Tips & Tricks

The M's multimode has received considerable criticism over the years. I understand this, as users often expect it to function similarly to digital-only synthesizers. However, the multimode was originally intended as a highly flexible system, allowing users to configure the four dedicated outputs for four independent instruments and process them via external effects. This led to a complicated voice management structure with reservation of hardware voices to the outputs, parts priority, and an inability to steal voices between parts.

Later additions to the synthesizer engine have further complicated matters. However, it is possible to create a working multi-arrangement by following several building rules. Here I will attempt to explain them.

Rule 1 – Parts Priority for Voice Reserve! The V.Total parameter of an arrangement sets the number of hardware voices that M will mark as reserved and locked by the DSP for each part. However, this is not a guaranteed reservation, as one might assume. Part priority (1 → 2 → 3 → 4) plays a crucial role here; priority always determines voice allocation. For example, if you reserve six voices for part 1, four for part 2, and two for part 3 on an 8-voice M, the actual distribution will be six voices for part 1, but only two for part 2 and zero for part 3. On a 16-voice M, reservations are honored as requested. Therefore, rule one is to carefully manage your voices, considering hardware limitations and part priority.

Rule 2 — Mind the PLAYMODE parameter for the part's sound. It makes no sense to set more than 1 voices for Monophonic sounds, the Voice Stealing does not matter too.

Rule 3 – Only the First Part Can Use Modern Mode! This is an important restriction related to the limited memory resources of the M's DSP. Remaining parts can be in classic or MVA modes, but only the first part can utilize extended RAM for the "modern" (i.e., Microwave 2) wavetable layout.

Rule 4 – Consequently, No Transition-Enabled Parts Can Mix with Part One in Modern Mode! Because transition replay for "classic" (i.e., PPG 2.3 / Microwave 1) mode uses the same extended RAM as the wavetables for "modern" mode, loading a part with transitions enabled will immediately corrupt the wavetable data for modern mode!

Rule 5 – There is Only One Arpeggiator in the M. Loading or creating an arrangement with multiple parts containing arpeggiator-enabled sounds is pointless; only one (first) part's arpeggiator will function.

Rule 6 – Cross-Parts VCF Lock. Regardless of which part is selected, if a part's LOCKVCF parameter is set to On, it will follow the VCF behavior of another part. This parameter allows all locked parts to track the controls of the currently selected locked part. For example, if

you have three parts and parts 1 and 3 have LOCKVCF = On while part 2 has LOCKVCF = Off, adjusting the VCF section's controls on part 1 will cause part 3 to follow the same settings (i.e., lock to part 1), and vice versa. However, this will not affect part 2. Consequently, editing part 2's VCF section will not influence these locked parts.

Rule 7 – Be Mindful of MIDI Filters. The M has a flexible set of MIDI filters capable of blocking/passing certain messages like mod wheel, pitch bend, program/bank changes, aftertouch, channel volume, and panorama. However, there is no CC filtering! Therefore, if you have two parts in an arrangement on the same MIDI channel, both will respond to all CCs received on that channel. And, don't forget about the global MIDI filters of the M, which applied first!

Rule 8 – Voices Routed to Different AUX Outputs Still Appear on Main/Headphone Output. The M does not exclude voices routed to different AUX outputs from the main output. In other words, the main out is a sum of all Aux outputs (which is also true from the M's schematic perspective).

Tuning Tables Load (Since FW 1.11)

Since FW 1.11, M is capable of supporting micro-tuning, that is, any possible frequency for any MIDI note. This functionality allows loading and saving scales from simple, hand-editable .txt tuning files placed in the TTABLES folder on the SD card. The tuning table file uses a simple record convention, similar to the note name map format in Reaper. Each note's tuning is represented as a string containing its MIDI note number and frequency, separated by a single space. The frequency must include a decimal point, even if it's an integer (e.g., 440Hz must be represented as 440.0). Each note occupies a single line, and the first line of the tuning .txt file is ignored. The tuning table file can contain any number of notes, from one up to 127. Only the notes listed in the file will be applied to M's actual tuning table.

Generally, workflow for the tuning files is similar to workflow for the User wavetables and Transitions. To load, save, and apply tuning tables, a new set of settings was introduced in FW 1.11. These settings are located on the sixth page of the **System Settings** menu:

Tune note to RAM : Specifies a single note to tune in RAM.

1/1 Herz : The integer part of the frequency for this note.

1/1000 Herz : The fractional part of the frequency for this note.

TunT file num : The number of the tuning table file, following the naming convention tuningXX.txt, where XX is a number from 00 to 15.

System Operations

Load TunT on start: Loads the specified tuning table automatically when M starts up. These settings are used within the Tuning Table Operation menu in System Operations. The following operations are available:

Load tuning table from the SD card: Reads the selected file (specified by the TunT file num setting) from the SD card into M's tuning table memory. This does not change the current tuning; it only loads the data.

Save tuning table to the SD card: Renders and saves the current tuning from M's tuning table memory to the selected file (specified by the TunT file num setting).

Set & transfer selected note: Based on the settings for Tune note to RAM, 1/1 Herz, and 1/1000 Herz, this writes the record for the selected note with the specified frequency to M's tuning table memory, enabling that tuning for a single note at the DSP level and sending an MTS realtime sysex message (Single Note Tuning).

Transfer full tuning table: Transfers the entire tuning table memory of M (128 notes/frequencies) to the DSP and enables these tunings for all notes.

Reset to standard tuning: Returns to the standard 12-equal tempered tuning, based on the Master tune setting for MIDI note 69.

Several helpful web services can simplify preparing your tuning table. I recommend using the

excellent tool Scale Workshop from <https://scaleworkshop.plainsound.org>.

It's possible to import scale and export it in the Reaper note name map (.txt) format, allowing you to load prepared tunings directly into M.

Over 5000 thousand of interesting modern and historical scales in .scl format can be downloaded from <https://www.huygens-fokker.org/docs/scales.zip>.

Here you'll find detailed instructions on how to load tuning tables.

<https://downloads.waldorfmusic.com/cloud/index.php/s/Mb3CztCD7W99Tew>

2024 - july - 01

FW : Host 1.10 RC - DSP 1.10 / rc 06

The major changes from 1.09 are :

- Added simple MPE support
- Added third oscillators mode - algorithmic oscillator mode
- Added modulation capabilities for the Extra parameter of digital filters
- MIDI Slave Clock much more stable now
- DSP Performance issues fixed
- FM filters for modern mode fixed.

A few words about MPE support :

From now, there is an option in the settings on page 4 - MPE enabled (on / off). The whole 4th page now dedicated for MPE setting, will be updated with the new MPE functionality within the next FWs. Now MPE has been implemented with the next restrictions :

1. Only Single sound mode.
2. Only Zone Low available
3. Master channel is channel 1
4. Only polyphonic allocator, no mono allocator available
5. New Mod Source - MPE Y. Channel Pressure & Pitch Bend are per note. Rest of the MIDI Modulators - on the Master Channel.

A few words about algorithmic oscillators mode. This was an old idea, which we had discussed with Martin Stuertz - to implement Mutable Instrument Braids module as oscillators in the M. Now I did the base for this, including all needed changes on the DSP and host side.

The four algorithms now were implemented, based on Braids ideas, but not fully from Braids originated. These 4 implemented are based on Braids algorithms, but highly optimized, to be able run them for each OSC of the M independently. So effectively it is 16 Braids running in the M, if all 8 notes are on (32 for 16 voices version)

How to enable the mode - Now the button "Mode" is Tri-State. When LED is off - this is classic mode (Microwave 1 ASIC emulation). When LED is on - this is the modern mode (MW2 DSP code emulation). When the LED is blinking - it is an algorithmic mode now.

Each WAVE control now has an extra page (Wave 1 Meta, Wave 2 Meta). On this page 4 parameters are presented :

ALGO - switches the algorithm of the oscillator

META - meta parameter of the algorithm (will explained below)

MODMSRC - modulation source for META parameter

MODMAMT - modulation amount for META parameter.

In this mode Wavetable control (outer ring) is switched off. The Wave control (inner ring) change the Wave parameter of the algorithm (valid for ALL algorithms), and usually change the waveform, generated by algorithm. All classic modulation are available.

Digital VCFs are all enabled for this mode. Except for the M Shaper, all digital filters should works as expected / as defined, but the M Shaper will be based on last loaded wavetable before entering the mode.

Ring Mod & Hard sync are also enabled for this mode.

There 4 algorithms implemented :

M VA - M Virtual Analog. It generates 3 waveforms simultaneously - Saw, Pulse, Triangle. The parameter Wave regulates mix of these 3 waveforms, next way - from 0 to 32 is crossfading between saw and pulse, from 33 to 63 is crossfading between pulse and triangle. Parameter META set the pulse width (5%-50%) mapped on 0-127 range of META parameter.

CSAW - CS-80 Saw - adaptation of CS-80 Saw from Braids. Wave parameter regulates the width of the "overpeak" in the beginning of sawtooth from 0 to 5%, mapped on 0-63 of Wave parameter.

FOLD - Wavefolding osc - OSC is feeded into Wavefolder. Based on one of the Braids with some changes. Wave parameter control crossfade between sawtooth and triangle, and META parameter controls Waveshaper's depth.

3 SAW - 3xSawtooth - A stack of 3 sawtooth, mutually detuned (<,0,>). Parameter Wave regulates a mix of the 0th (i.e. not-detuned, aka middle) saw with another two. Parameter META regulates detune between saws. Used for emulate supersaw.

2023 - june - 21th

FW : Host 1.09 RC - DSP 1.09 RC

WALDORF M FW 1.09 changes log

Since FW 1.09 Waldorf M is supported in the „edisyn“ patch editor for both Single and Multi modes. The download link for the editor is <https://github.com/eclab/edisyn>

We thank Sean Luce for his excellent work on the Waldorf Synth's support at „edisyn“!

BUG FIXES / IMPROVEMENTS

fixed - multimode MIDI processing for parts on separate MIDI channels now works as expected (Pitchbend & Co issue)

fixed - bug with hi-res parameters (Transition Start / Transition End) in sysex sender/receiver

fixed - UserWavetable load for 16-bit waves — corrupted wave 1 (take care of RIFF chunk offset properly)

fixed - Smooth Scan misbehavior on fast modulation rates (due to fractional overflow)

improved — M now suppresses CC emitting, if a sound parameter was set up via sysex

ADDONS

added - sysex pair 0x60 / 0x61 - M state request/answer (editor support)

added - possibility to update single sound via sysex only in working RAM, do not save it directly to ROM first (editor support)

added - updater function to track bulk sound updates via SYSEX and reconfigure DSP (editor support)

added - updater function to track UI updates during bulk sound updates via sysex (editor support)

added - sysex 0x66 to changing Multi Arrangement (as regular PC/BC messages change the Sound within the Part since FW 1.08)

KNOWN BUGS & WORKAROUNDS

The MIDI clock drifts in Slave mode. Unable to fix without changing the graphic engine. Proposed workaround until fixed: Use M as the Master Clock source, if possible.

The Multimode could overdrive the DSP in dual mode if voice stealing & glide are enabled. Unable to fix without changing the graphic engine (need ROM space for updating the voice stealing algorithm). Proposed workaround until fixed: Use the Modern Mode for Part 1, to reduce DSP load.

The Transition replay is sometimes overdriven/clicky for user transitions. The possible reason is overflow due to the interpolator. Possible workaround — use normalization for -6dB gain in transitions to avoid such an issue.

2022 - december - 01th

FW : Host 1.08 RC - DSP 1.08 RC

BUG FIXES / IMPROVEMENTS

- fixed: user WT loading to modern mode with two different UWTs.
- fixed: screen refresh on receive sysexes does not occur anymore.
- fixed: sample transfer can lose samples under certain conditions.
- fixed: color bug in control slots.
- fixed: the Mod Wheel split between the parts.
- fixed: Note On Velocity 0 override if arpeggio is enabled.
- fixed transitions samples "corruptive" load on certain cases
- fixed ASIC emulation for classic mode with transitions
- fixed transition replay in FWD loop
- fixed transition volume in ST+FWD loop
- fixed onset click

- improved: removed MIDI LED blinking on MIDI Clock arrived.
- improved: MIDI Clock Slave mode more stable tracking (although not fully fixed with certain devices with 48/96 ppm).
- improved: modern user WT transforms in modern mode — lowest octaves now not losing half of the samples.
- improved: now by default all UWTs are sawtooth (valid only for new / flash formatted M).
- improved: global LFO settings rate and shape are now separated from LFO2.
- improved: an added orange marker for MW2 mode-related parameters.
- improved: checked alignment of slots in UI.
- improved: updated User WT loading routines, changed interpolation algorithms. Fixed bug with lost samples.
- improved: some adjustment in the random sound init for digiVCF
- improved: UI fixes after user requests (MIX, VCF pages).
- improved: release stage of EG now released until -105 dB instead of -85dB (as on MW1). Allow the following S&H of 508 of Microwave 1 more precisely.

NEW FEATURES

- added: new system parameters - VCA control scalers, 16-bit UWT samples setting.
- added Sustain Pedal reaction to the WaveEG and FreeEG
- added option for randomizer - spread from default or spread from the middle value of the parameter.
- added: Smooth scan implemented for modern mode.
- added: last three digital VCFs of MW2 for modern mode.

SAMPLEHOLD - S&H followed LP12 — Extra parameter is the clock of S&H.

SHAPER - MW2 shaper (variable shaper) - Extra is Wave of WT2 - the «Wave-To-Shape».

FMLP12 - OSC2 output as FM input for LP12, Extra is the depth of modulation.

- added: extra 4 digiVCF types for modern mode (one also for both)

the next two were proposed by gearspace user Xpanderbeanz — many thanks! :

SAMPLEHOLDHP - S&H followed HP12

FMHP12 - OSC1 as FM for HP12

FBOOST - 6db BP ac as EQ (Extra = mix between the dry signal and BP6-processed), cutoff set frequency, and resonance set the amplification of frequency. Works also in classic OSC mode!

MSHAPER — M shaper is the same as MW2 shaper (OSC2 Wave as shape curve) with one exception — it uses not an Extra parameter but real actual wave from OSC2. I.e. it is a modulable shaper, i.e. shape curve will constantly change as long as the wave of OSC2 is changing. In other words, it always takes an actual wave of OSC2 to shape the signal. (Shape curve is an actual wave from OSC2, Extra-Depth (mix between processed and dry signal)). Try it, and welcome to the darkest side of M!

- added : Gate State as modulator (Gate On = 1, Gate Off = 0).
- added: 16-bit user wavetable loader (with selectable wave size & interpolation). Just use wtslotxx.wav file (instead of wtslotxx) with signed 16-bit mono samples inside, 256 samples wavelength.
- added: Option to save Multi and update the linked sounds simultaneously. One needs press Shift + OK on the Arrangement Save screen, and all single sounds, linked to the arrangement, will be updated too.
- added: Voice Noise modulator for all OSC modes. Try it and blow your mind with blown-like sounds. Through reverb is a deep space dive.
- added: SD card simple logger (bank/patch/multi/part save/restore on load).
- added: VCA "scaler" per voice support (work like VCA manual attenuation tuning in the range 0.5 - 1.0 "scale" of max attenuation. As it is not logarithmic, applied to the CV BEFORE analog logarithmisation — the effect is relatively subtle (ca. -10 dB) but allows to «soften» VCAs response of M.

SYSEX

(details in Sysex documentation)

- added : Sysex command 0x70 0x71 (request SoundParam / answer & set SoundParam) implemented.
- added : Sysex 0x74 - full sound request with 0x72 answer. The answer is universal, i.e. can be sent in the currently active patch.
- added: Sysex 0x7A / 0x7B - multi-parameter request / answer-set accordingly.
- added: 0x75h - Multi Request Sysex with answer 0x73.
- added: System parameters request/answer (0x7C / 0x7D) Sysex.
- updated: 0x72 Sysex processing, now with bank/slot select enabled if any inside.
- updated: 0x73 Sysex (arrangement), now with bank/slot select enabled if any inside.
- other minor performance improvements.

Sincerely yours

2022 - february - 20th

FW : Host 1.07 RC - DSP 1.07 RC

Hey guys (and gals!)

The new 1.07 FW is baked, the single person tested, and ready to launch.

As this was a DSP-oriented release, most of the changes/fixes/addons were made in the DSP area.

Here are the most important bug fixes :

- fixed abrupton of LFO1 in synced mode (Decay should be = 127)
- fixed corruption of the first user wt on the save / load / mode change, in case they are different.
- fixed «first 8 notes problem»
- fixed oscillators leaking on the Master Volume = 0
- fixed bug with non-acting Hard-Sync and Glide button's LEDs when a user changes parameters using under-screen encoders

Addons are :

- Implemented bank change (as Bank Change LSB = CC32)
- Implemented various WT scan limiting settings for the Modern Mode. The new sound parameter called TRAVEL for both Waves has the next settings :

Analog On (An. On) — regular 0 ... 63 scanning

Analog Off (An. Off) — top 3 waveforms (analog) excluded. Scanning in range 0...60

Analog Only (An. Only) — scanning possible in range (60...63)

Circular — scanning is in range 0...60 with wrap. I.e. if modulation sum lays over 60 (or under 0) there will be a wrap-around over border and scanning continue from the «other side».

For example, modulation Sum 66 will lead to wave 6 if you have chosen wave 0 as start and wave 26 if you have chosen 20 as start ($20 + 66$) = full overflow + 6 wave positions.

Implemented additional fast/slow modes for the LFO1 and LFO2.

The parameters names are L1 RANGE and L2 RANGE and they are located on the new page in section LFO — LFO Tweaks.

The settings are :

Normal — standard, MW1 — derived settings (0.1 — 15.4 Hz Range)

Slow — slow range (0.012Hz — 1.54 Hz Range = $0.1 \times$ Normal Range) Fast — fast range (0.485Hz — 61.6 Hz Range = $4 \times$ Normal Range)

Implemented additional Glide mode (reverse exponential mapping). The new parameter is called GL.MODE and has next settings :

lin — standard setting (0-2s linearly in time scale)

exp — new setting (0-1.2s reverse exponential mapping scale)

Extended the range of the B.RANGE parameter from +-12 to +-48 semitones for both oscillators

Added bit reduction for the Modern Mode. The parameter name is Derez, and this is controlled by how many bits will be reduced from the normal 16 bits samples of Oscillator's Modern Mode. The Range is 0 — 14. With 14 bit you are achieve effectively 2-bit grittiness :)

Added next MW2 VCF models for the Modern Mode* :

Notch 12dB / Oct, LP 24dB/Oct, BP 24 dB/Oct,HP 24 dB/Oct, Notch 24 dB/Oct, Sin Waveshaper + LP 12 dB/Oct, Dual LP/BP 12 db/Oct, BandStop 12 dB/Oct

Added for the Modern Osc Mode all parameters to control the digital VCF as on the MW2

Envelope Amount (VCF EG to Digital VCF Cutoff), Envelope Velocity (Velocity to VCF EG scaling for the digital VCF), Keytrack, Extra parameter** (see description below), Modulator1 Source, Modulator1 Controller, Modulator 1 Amount, Release Modulator Source, Release Modulator Amount. The previous Modulator Source and Amount became Modulator 2 Source** (see below) and Amount accordingly.

* only existed 3 will be active in the Classic Mode. No of new addons will be active there!

** Extra parameter of Digi VCF controls the next :

- in the classic mode — it controls the attenuation of the VCF — the more value the less the attenuation via resonance. It allows morfing of the VCF sonically between version of FW1.05 and FW1.06
- in the modern mode — it controls the shift of the BP 12 VCF for the Dual LP/BP 12 dB/Oct VCF from the LP12 frequency. It could be only positive.

*** all parameters described below, except Extra, Mod2 Source Mod2 Amount are NOT available in the Classic (ASIC emulation) oscillator mode! Period! due to DSP MCU performance restriction. The patch parameters of FW 1.05-1.06 ModSource ModAmount of DigiVCF will be translated from existing patches to the Modulator 2 Source and Modulator 2 Amount accordingly.

- Added Range Parameter for the EG1 (VCF) and EG2 (VCA). Now the range of the timing could be changed for these two, related to the CV of Analog VCF and VCA accordingly. The new parameter called RANGE, located at the Page ENV Tweaks, and has the next settings :
normal — regular range, derived from MW1

fast - range within 0.01s — 1s on each segments (A,D,R) of the envelope

slow — range within 0.05s - 1 hour 35 minutes on each segment (A,D,R) of the envelope. Drones are welcome.

And last but not least. This is my thank you feature to all who supported me at the early launch phase of M and during the first batches. I always dreamed of

PPG Waveterm Transitions replay. So there are they here in M!

Now in classic mode one can use the second oscillator as a PCM oscillator and play one of the 64 transitions, saved into internal flash. They could be loaded from the SD Card to the M and used for your patch. So you can make 4 channel Drum Machine with analog and digital VCF on each channel. Or multisample layered patch. And the OSC1 stays free to play its own wavetable. So I am very curious, what you can introduce with this new feature.

There is next sound parameter were added (located on two new pages of the OSC section) On the page OSC2>Transition next 3 parameters are located

Transition Enable — (TR.ENA) enables replay of transition on the OSC2. The OSC2 controls Octave, Semitones, and Detune, and mixer Osc2 are active and make the same functions as on the regular Oscillator 2. The Envelope Amount is inactive as there is no wavetable scanning occurring.

Transition Number (TR.NUM) — number of slots (00-63) from where the transition sample will be loaded and transferred to the DSP.

Transition Base (TR.BASE) — base pitch for the transition (usually D#1 for converted PPG Waveterm B library) Range is within C-2 (MIDI Note #36) to C3 (MIDI Note #96).

The exact pitch in Herz for the according to midi note will be also changed if you change the Master Tuning, so mind this fact. I mean, that if you choose 442 Hz as master Tuning, the base pitch of A2 will be also shifted to 442 Herz in this case. It could lead to improper sample tuning if a sample has been recorded at 440 Hz as the base pitch.

Next 3 parameters related to the sample replay looping and located on the next page (OSC2>TransitionLoop)

Transition Start — (TR.START) — start sample number of the loop, if any.

Transition End — (TR.END) — end sample number of the loop, if any.

Transition Loop — (TR.LOOP) — loop type selector. The values could be the next :

OFF — loops are off, transition will be replayed from sample 0 to sample 32767 once. Good for one-shots as drums or SFX

FWD (Forward loop) — transition replay will be started from the sample TR.START and will be looped between TR.START and TR.END until the note is active.

BWD (Backward Loop) — transition replay will be started from the sample TR.END and will be replayed in the reverse direction to the TR.START and then looped.

ST+FWD (Start + Forward Loop) — transition replay starts from sample 0 and will be played until TR.START, then looped between TR.START and TR.END.

ST+BWD (Start + Backward Loop) — transition replay starts from sample 0 and will be played until TR.END, then looped in the reverse direction to the TR.START

The transition file format is regular .WAV file, 16-bit mono, 22050 Hz Sample rate.

It's close to the native Waveterm's sampling frequency of 20,95 kHz, but the base pitch for 22 kHz will be D#1 instead of E1. The easiest way to take a tour of this transition world — is to [download the library](http://www.ppg.synth.net/waveterm/wt_lib.shtml) from the http://www.ppg.synth.net/waveterm/wt_lib.shtml

There are 2 new system options parameters in the System Settings,

page 4 — UserWT (now UserWT/TR). They are the same by meaning as according to parameters of loading user wavetables and allow to choose the source (wave file on SD) and destination (transition slot on the M's flash).

The source parameter (Trans. file number) is in the range of 000-999 and corresponds to any file with the name Txxx.wav, where xxx is the number of the named range.

The files should be located on SD Card in the folder TRANSITIONS. The destination parameter (Trans. load slot) is within range 00-63 and corresponds to the internal slot of M's flash

storage. The new action in the System Operations menu, named Import Transition (from SD) used to transfer the samples from the wave file to the internal M slot, according to the destination parameter.

If the chosen file will be bigger than 65Kbytes (32767 Samples) — only the first 32767 samples will be transferred. If the file is smaller — all samples not contained in the file will be filled with 0. So there are actually the major changes of the FW1.07. There are a lot of performance tweaks under the hood that were made and now the DSP performance for the classic mode is on the limit. So with this, I will lock future sound parameter add-ons/changes. Only the parameters for the global LFO could be changed/added in the new release. So, NO more new sound parameters for the classic mode for all 1. xxx firmware versions for the Waldorf M. Hope you will like the new possibilities.

The next 1.08 firmware will be UI/UX oriented to optimize the System Firmware. There I will try to cover all requests and fix all bugs related to the interaction between machine and man.

Thank you all for your priceless help and support! - Greetings from Remagen Vladimir

2021 - november - 03th

Meet the new feature - second digital VCF in both OSC modes.

Technically speaking, this is an adaptation of Dattaro-Chamberlin SVF (which is similar to MWII implementation of LP12/BP12/HP12 VCF modes).

However, it is NOT a direct clone of such MWII filters into M.

So, what we have here as DigiVCF. This is actually LP/BP/HP SVF with 12 dB / oct slope, with resonance, and **without self-oscillation**. It stays directly between Mixer output and analog VCF, i.e. this is the last digital element of M's sound path. It can be enabled/disabled, and cutoff frequency could be modulated through one modulator. On the full resonance, it's attenuate a signal at 12 dB, and with an ASIC bug enabled, it could be overdriven up to clipping. Without ASIC bug it should not be overdriven. The cutoff is within the range 10Hz - 10kHz. This was made intentionally meaning to fit into the Classic OSC Mode performance. However, it passed the M character well. This is an experimental, non-planned feature, but allows to advance the sonic possibilities of the M a bit more.

It could be found on the MIX section of UI, pages 4 and 5. The graphics were not implemented in this release. Honestly, I need feedback about this feature generally.

FW : Host 1.06 RC

- fixed the voice stealing bug in Multi-Mode (broken round-robin).
- fixed mono retrigger allocator mode.
- fixed mono legato allocator mode, changed to duophonic legato (allows trill, but does not track more than one previously pressed key)
- fixed bug with loading sound if it contains two different user wavetables.
- added DigiVCF sound parameters.
- added DigiVCF UI pages.
- added DigiVCF support for existing patches.

FW : DSP 1.05 RC.

- implemented digital VCF in classic & modern OSC modes.
- improved performance of DSP engine

As the DSP engine was touched - I would like to advise making a quick test of overall functionality. I did test a lot, found it good, but perhaps can miss something.

Known bug NOT yet fixed (moved to the next update) :

- Arp does not react on note On velocity 0 (Elektron Machines).
- If two different user WT loaded, on switching the OSC Mode first table dropped from DSP RAM and should be reloaded.

p.s. The randomization of DigiVCF now is tied to the randomization of the OSC section. The randomizer will be improved at the next update to allow control of this separately.

Sincerely yours - Vladimir Salnikov

2021 - october - 26th

FW : Host 1.05 RC

- Fixed Mode behavior for parts 2,3,4 issues in multi-mode (UI too). Now definitely, only the first Part will be in Modern Mode.
- Fixed Mode state on load - now is coherent and independent for all parts in both single and multi-mode.
- Fixed bug with switch OSC Mode through parameter encoders.
- Fixed part voice stealing setting in Multi-Mode
- Added reaction to CC 7 (Vol) and CC 10(Pan). Now it changes instrument volume (VCASVolume) and Pan accordingly.
- Added option to fast load patch in System Settings (Single Sound only)
- Added CC_ALL_NOTES_OFF message processing
- Optimized MIDI CC processing

Brand new features designed after user requests :

- Added customizable random init functionality.
- Added support of 3 new modulators - Inverse, CoinFlip, Random

FW : DSP 1.04 RC.

added support of 3 new modulators

(each is static during note On and fired once at Gate On) :

1. Inverse -1.0 / + 1.0 each new note

2. CoinFlip - Coin toss with results 0.0 1.0 (equal of min or Max) each note
3. Random - random value between -1.0 and 1.0 each note

A word about random init - This is Lazy Sounddesigner in the M „Geist in der Machine“.

If you press Shift+Cancel and then Recall (instead of Regular Shift + Recall) - the patch will be initialized with random drift of settings.

The spread of randomness could be preset in the System Settings - there is a new page RND Init.

The settings are next :

Enable Mod RndInit - control whether all mod sources amounts and control will be randomized or stay as in the default. Useful if one does not want to have ALL modulators randomly init.

LFO,ENV,OSC,WAV,MIX,VCF,VCA RndInit spread - set in percent 0-100 the possible drift from default for according section of sound. The more spread - the more result will differ from regular init.

My beloved settings are the next :

Enable Mod RndInit - yes;

LFO RndInit Spread - 47;

ENV RndInit Spread - 12;

OSC RndInit Spread - 82;

WAV RndInit Spread - 24;

MIX RndInit Spread - 67;

VCF RndInit Spread - 26;

VCA RndInit Spread - 21.

These settings will allow sound to not drift so far from unusable, however far enough to produce cool valuable results (at least to my taste). And of course, as they are settings - could be stored. So one need to tune them once at the taste and then use Patch Random init as creativity implant

A word about the new 3 modulators :

1. Inverse - -1.0 / + 1.0 each new note. It is global (i.e. affect all parts if used). Starting from -1 and then each new note in any part will invert to +1 and again and again and again. Could be used to simulate PPG Stereo Basis very easy as pan mod + a certain amount which actually act then as Basis pot on the PPG.

Effectively, you can take into account this as a switch between -64 and 63 switching once for each new note and staying with the note until the note will be retriggered.

2. CoinFlip - Coin toss with result 0.0 1.0 (equal of min or Max) at each note. A source of uncertainty

Effectively, you can take into account this as a switch between 0 and 63 with a probability of 50% of each, changing for each new note and staying with the note until the note will be retriggered.

3. Random - random value between -1.0 and 1.0 for each note.

Effectively, you can take into account this as a source of something between -64 and 63, changing for each new note and staying with the note until the note will be retriggered.

So it's kinda per voice modulator, fired once at Note On, and does not change during note play.

Sincerely yours

Vladimir Salnikov