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1.0 System Requirements

Doppelmangler requires at least a 1 ghz Intel with 128 megs of ram or a comparable AMD machine. Since Doppelmangler loads .WAV files, you will want to have some of these as well. If you plan on using Doppelmangler's image to sound capabilities, it would help to have an image editing program capable of producing and editing .BMP files, such as Adobe Photoshop.

1.1 Features

Resynthesizes wave files allowing pitch to be played independently of time speed up, slow down, reorder, reverse samples with the Time envelope

Up to 256 freely tunable partials per source

2 sources with morphing

Spectral decomposition of wave samples

Image to sound capabilities using .BMP files

MangleFX spectral effects system includes things like frequency shifting, advanced vocoding, time warping and more.

FX modules: Distorion (with bitcrush and waveshape), phaser, delay

Formant Filter with morphing and vocal presets

Assignable modulation matrix

flexible "MultiEnvelopes" that can be used as 256 breakpoint envelopes or as drawable LFO's

Beat Sync mode for MultiEnvelopes and Time Envelope to sync to tempo

1.2 Definitions

1.2.1 What is resynthesis?

Resynthesis involves the decomposition of a sampled sound into individual sine waves or "partials". Through this method, any sound can be recreated with the advantage of being able to change the pitch of a sound without speeding up the sample. Likewise, the length of the sample

can be changed without changing the pitch which makes it ideal for remixing beats and vocals. In addition, other effects can be done in the spectral domain which are not otherwise possible with regular sampling.

1.2.2 What are partials?

Partials are individual sinewaves that can be combined to create a sound. For our purposes, a sine wave is the most basic kind of sound, from which all other sounds can be created. To perfectly recreate a sound, thousands of partials may be needed. Since computers cannot calculate this in real time, we choose the 256 most important partials in a sound and use those. This gives us a "good enough" copy of the sound which we can then manipulate in various fun ways. A partial has a few bits of information associated with it. It has a Pitch envelope which keeps track of how the frequency of the partial changes over time and a volume envelope which tracks the changes in amplitude. So, when you play back a synthesized sound, the synthesizer steps through these envelopes. When you speed up or slow down the sound it simply means that the synthesizer moves more quickly or more slowly through the envelopes. Changing the pitch of the sound changes the pitches of all the partials together.

Partials are created for you automatically whenever you analyze a sound or load a .bmp file. You can see a picture of what the partials would look like in the "Spectrum" view. Otherwise, you don't need to worry about them much. You cannot directly edit the partials. This was done purposely to make editing the synth easier and more intuitive. Instead of editing hundreds of partial values by hand, use the MangleFX and make broad spectral changes using only a few knobs.

1.3 Signal Flow Diagram

**Source1 MangleFX Morph(to Source2) Formant Filter Amp Envelope FX
(Distortion, Chorus, Phaser, Delay)**

1.4 Quick Start

Loading Wavs

Most people will be using Doppelmangler to play sampled sounds. To do this Doppelmangler must analyze a sound on your computer. The kind of sound Doppelmangler reads is .WAV files. Load a sound by finding it in the File Browser (on the Spect Page) and double clicking it. The sound will be analyzed and then ready to play. Doppelmangler also reads .BMP files and is capable of exporting BMP files. The maximum length .WAV file that Doppelmangler can read is 30 seconds. If your PC runs out of memory, encounters a bad .WAV file, or the sound is too long the sound simply fails to load and DM does nothing.

DM works best with short loops and "oneshot" sounds (ie, a single piano note), though you can use whatever you like. If you are loading beats see the section in Trouble Shooting "Beats are off time or don't sound as crisp" for tips.

Loading Presets

Doppelmangler loads presets through it's file browser, the same way you load .WAV's. Preset files are files with the extension .DMG. Doppelmangler comes with many presets - to start hearing these, press the button marked 'LIB' under the directory listing in the Filebrowser. This takes you to the default preset directory. There are several sub directories for each category of sound. Find a sound you want to hear and double click it to load it. Note: there is no undo, so be sure to save any work you are doing first, because it will be overwritten. You can save your own presets by going to the directory you want to save in with the FileBrowser, then hitting save. Whatever name you have chosen for your current sound is the one used to save the .DMG. Use your host's preset naming options to change the name of the preset. If a preset of the same

name exists, a number is attached to the end of the name so you can save several sounds with similar names without overwriting them.

2.0 Advanced Usage

2.1 Getting back to default settings

Starting a new sound is easy. Press "LIB" to go to the preset directory. Find InitPatch.dmg and double click to load it. This will reset all your settings to the default ones.

2.2 The Time Envelope

The time envelope is used to determine the rate and order that a sound is replayed. It can be used to speed up and slow down sounds, reverse sounds, reorder sounds and loop portions of sounds. First, check out some of the presets (preset button). The vertical axis of the graph represents the time position within the analyzed sound. As you can see in the default settings it starts at zero and goes to the end of the sound. The bottom always represents the beginning of the sound and the top always represents the end of the sample, no matter how long it really is. To reverse the sound you would simply draw a line starting at the end down to the beginning. You are free to draw any kind of graph you like, to replay the sound in different ways. There are several shortcuts for easier editing:

Hold down SHIFT and draw to snap to quantized values. This makes it easier to do rhythmic things.

Hold down ALT and draw for free drawing

Hold down CTRL and drag to re-order 16th blocks. This is great for remixing beats.

Length - The length of time that the time graph represents. When sounds are imported their length is automatically set.

L - Determines if a sound is looped or not. The area of the time envelope that gets looped is the white bar underneath the time graph. You can click in this area to change the start and end of the loop. Click and release without dragging to select the whole thing. Shift also works with this.

R - Release the sound on Note Off? If this is set, then the sound stops looping and goes to the end. If not, it will keep looping even after the note is released.

S - Sync. When set, the length will correspond to Beat values for easy syncing to your song.

2.3 MangleFX

MangleFX are spectral effects. They act on the frequency and amplitude values of the partials in realtime and are non-destructive (meaning they don't actually change the partials when you go to apply another effect). Here is a rundown of the effects and their parameters:

TimeWarp - Offsets the timeline of higher frequencies so that they happen before or after the lower ones. The change gets successively more as the frequency of the partial gets higher. The end result can sound somewhat like an enveloped lowpass filter.

Params: Freq Delay - The offset amount. High values mean the frequencies happen before the lower ones, lower values mean they are delayed.

Twist - Bends the frequencies away from some center point

Params: Center, Amount, Tighten (acts as a bandpass filter around the center)

Robotize - Creates a synthesized sounding version of the sound

Params: Pitch, Timbre, Odd partial volume

Peaks - Creates spectral peaks in your sound
Params: Peak spacing, Peak Offset

Hybrid - This uses the second source, so make sure you have a second sound loaded (click the source switch over to 2 and then load a sound). What this effect does is put the amplitude envelope of the second sound on the frequency envelope of the first creating a sound which bears similarities to both sounds).
Params: Amount

Transform - Modifies the formant properties of a sound
Params: Formant shift

Harmonizer - Scales every other partial to a new pitch. Sounds like there is another copy of the sound being played at a different pitch.

Gate - Removes partials whose amplitude falls under the threshold
Params: Threshold

Ultravocoder - Creates a vocoded sound with formant control. This mode can also take advantage of root pitch information saved when a sound is analyzed. The root pitch is displayed as a jagged white line on the spectral view. Pitch detection is never perfect, so you may need to edit this line (by click on the view) to follow the lowest harmonic in the sound. The pitch following mode is activated with the first parameter. This Mangle effect sounds great on beats.

Params: Pitch follow, Formant, harmonics

Pitchfixer - attempts to keep the pitch steady which results in lots of formant variations.
Params: none

Delay - Offsets the time of different partials
Params: Crossover - Partial # where offsetting starts occurring. Time - Amount of offset. Odd: even/odd #'d partial delay amount.

Quantizer - Lowers the number of timeframes in your sound. Useful for making glitchy sounds.

FreqWarp - A nonlinear warp of the frequencies
Params: Warp Amount

Maximizer - Can be used to make certain bands of the sound fuller.
Params: Low, Mid and High bands (Lower means more maximization).

Source Vocode - Uses source 2 to vocode source 1
Params: Mix, Pitch of source 2 (for formant effects).

Resonator - Acts like you are running the sound through a short comb filter. First parameter is the mix, second is the "pitch", and the third is the bandwidth of the peaks generated.

2.4 Morphing

This morphs between sources 1 and 2 when you have two sounds loaded (do this by switching to source 2 and load a sound). Morphing linearly interpolates amplitude and pitch of each partial.

2.5 Formant Filter

A formant filter is a filter that can be used to boost or cut frequencies across the spectrum. It can be used to simulate analog filters, vocal sounds, EQ's and more. There are two views displayed here. Each is a state for the formant filter and it is possible to morph from one state to another to

create evolving sounds. Horizontally on each view runs the frequencies and the vertical domain is the level at that frequency. You can edit the levels by clicking on the view. You can also drag the points around by holding down alt and click on one. These are morph points, so when you morph, the points will morph if you have moved them. So, this way you can have a peak move across the spectrum by keeping the same points for the peak but moving them using the alt key. Try loading two vocal sounds and morphing between them to see what I mean. The points of the peaks are the same points but at different frequencies so they get smoothly interpolated when you morph.

You can also adjust the formant bias in addition to morphing which can be used to sweep the filter across the spectrum. This is similar to the "Cutoff" knob on an analog filter. Try it with some of the LP (lowpass filter) presets.

2.6 Amp Envelope

This is a simple Attack Decay Sustain Release (ADSR) envelope. Attack is the time it takes a sound to fade in. Decay is the time it takes to drop down to the sustain level. Sustain Level is the volume that stays while a note is held, and release is the time it takes to fade back to zero after a key is released. Adjusting these can change the character of an instrument. For instance, a sound with a long attack is perceived as string-like, while a sound with a very short attack and a short decay down to a lower volume is perceived as percussive. For more information on this, please look up some general info on synthesis.

2.7 Import Options

FFT Size - To analyze a sound a Fast Fourier Transform is used to extract the frequency components. Fast fourier transforms have a trade off - the bigger the "buffer" the greater the frequency accuracy but the lower the time accuracy, and vice versa. This option lets you tailor the buffer size to your needs. For things like drum parts, you'll probably want high time accuracy. For instrument sounds you might want greater frequency accuracy. Experiment and see what works best for the source material you have. For most sounds, you can just leave this on the default.

Time Resolution - Lets you boost the time resolution by overlapping the FFT buffers. Again, how you use this depends on your source material. Greater time resolution results in more "time slices" of your sound so finer resolution will also give you a larger file size.

Pitch Detection - For a couple MangleFX, the root pitch is used (namely the Ultravocoder and PitchFixer). This changes the technique used to determine the pitch. Use the first for normal sounds, the second ("loudest") for experimental sounds and beats. The third mode, "Instrument" changes the entire way that DM imports sounds. DM will try to assign partials to their proper harmonics using the root pitch. This mode can produce more interesting results when used with morphing. It also works well on simple one-shot notes and on some vocals. If a sound is an instrument, try using this mode.

BMP Import - Determines how Bitmap images are imported.

Harmonic - Each row of the bitmap corresponds to a harmonic frequency. This is good if you want to make harmonic instruments.

FullSpectrum - The rows of the bitmap are mapped linearly to the full spectrum (0 .. 22050 hz)

FFTLike (default) - Interprets the bitmap as if it was data from a Fast Fourier Transform and attempts to extract the most important frequencies from it. This can be used in conjunction with the BMP export (see below) to

BMP Export - When set to "Yes", Doppelmangler will create .BMP everytime you analyze a .WAV. You can then edit the .BMP in your favorite paint program and reimport it. When you re-import the sound, make sure BMP Import is set to "FFTLike". Note: there is some

frequency quantization when using this. For the best results, you may want to use a higher FFT size as this also creates a more accurate .BMP. This function allows you to do visually edit the spectrum of existing sounds for wild experimental effects.

UI Skin

This allows you to change the look of Doppelmangler. There are several skins available

2.8 Global Settings

Here you can change several things about how the sounds are played.

Playmode - Determines if the sound is mono or poly (mono sounds only play one note at a time and are good for things like leads).

 Mono1 - Does not retrigger envelopes when played

 Mono2 - Retrigger envelopes with every note

 Mono3 - Retrigger envelopes only when a note is let go

Bend Amt - Number of semitones that a sound can be pitch bended with the pitch bend wheel.

Poly Limit - Allows you to limit the polyphony of a sound to ensure that your CPU doesn't get overloaded.

Master Volume - Controls the final volume before output.

3.0 MOD Section

The mod section can be used to modulate parameters. There are several important knobs that can be modulated and several sources you can use to modulate them. Modulation is useful for causing changes in your sound over time or with playing.

Sources

MultiEnv1

MultiEnv2

Amp

Env1

Keyscale

Velocity

Random

PadX

PadY

ChannelPressure

AfterTouch

ModWheel

CC2 thru CC6

Destinations

Time

FFBias

FFmorph

Morph

MangParm1 - (Mangle FX parameters)

MangParm2

MangParm2

Amp

Pitch

Phaser Frequency

Time2 - (Time envelope for source 2)

To modulate, simply select a source and a dest and move the slider to the desired amount. There are two possible pages of modulation options. Move the slider marked '1' over until it says '2' to view the second page. The modulation amount can be either positive or negative with a range of 1 in either direction. All sources with the exception of the XYPad (in one state) generate values in the range of 0...1. In the '+/-' state the xypad values range from 0...1. All knobs have values in the range of 0...1.

3.1 Multi Envelopes

These are envelopes used for modifying parameters in the mod section. They function the same as the time envelope, so see that section for help. The only additional feature of these is that you can check "Free" to have them behave as LFO's. When in free mode, the MultiEnv runs continuously. In regular mode they are triggered everytime you hit a note. Also, in free mode the MultiEnv is "per voice". That is, each note has it's own virtual MultiEnv. In "Free" mode all the notes share one MultiEnv.

3.2 ENV1

This is an extra envelope you can use for modifying parameters.

3.3 XYPAD

This allows you to modify parameters with your mouse while your song plays. You can click in the XYPad area and your mouse movements are converted to modulation values. There are three modes.

Pos - The default mode

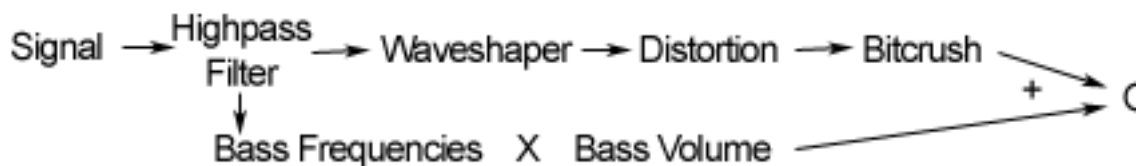
PlusMinus - Allows you to modulate up or down (not just up).

Circular - The angle of your mouse is used as the PadX value. Useful for turntable type beat playing?

3.4 FX Section

Distortion

Adds an aggressive edge to sounds. In this distortion module, the signal is split into two parts. The high half gets distorted while the lower part is preserved. This is done to alter the tone and also to preserve the bass frequencies. You can also use this as an "exciter" type effect by just distorting the high frequencies. The signal path looks like this:



Params:

On/Off - Toggles the effect

Drive - Volume gain before distortion

Bass Volume - volume of the bass frequencies. Can be used to boost the bass frequencies

Crossover Freq - Above this frequency, distortion occurs

BitCrush - Lower the number of bits to create a Lo-Fi sound

Waveshaper - Wrap the signal to radically distort the signal. Setting the knob to it's leftmost position turns off the effect. Increasing the knob's value increases the amount of wrapping.

Chorus/Flanger

Adds a detuned stereo copy of the sound to create a warmer, fuller sound.

Delay - Initial delay

Rate - Rate of delay modulation

Mod - Amount of delay modulation

Feedback - echo regeneration

Mix - dry/wet mix (right for more wet)

Phaser

Phasers add a spectral peak to sounds and can be used to create spacy or filtery effects.

Params:

On/Off - Toggles the effect

Freq - Min. Frequency for phaser sweep

Mod - The amount of frequency modulation

Rate - Rate of modulation

Feedback - Resonance of the peak

Delay

This is a simple static pingpong delay. It can be used to create a sense of spaciousness to sounds.

On/Off - Toggles the effect

Amount - mix of delay to dry signal

Feedback - affects the number of echoes

4.0 Trouble Shooting

I don't see how to load presets with my host

You can save and load single .FXP presets using your host. All DM's presets are in DMG format which is specific to DM. You can load these the same as you would load a .WAV, through the file browser. Just double click.

DM is not making any sounds!

- Make sure DM is set up properly in your host application, volume is turned off, the source is unmuted, etc.

- Make sure the Master Volume is not turned all the way down

- Check your formant filter settings - maybe you are filtering out all the sound?

When my sounds are ringy or have artifacts (happens often with timestretching too).

- Try increasing your Time resolution during analysis or changing the FFT size. Also, try to be sure your source material

is relatively reverb free for the best results. It's not always possible to get good results with all types of samples. Samples that have many instruments or are complex may not be recreated well. If the sound is a vocal or simple instrument, try setting the Pitch Detect option -> "Instrument" and reimport the sample.

Sounds are noisy

- It helps to filter out some of the inaudible high's from your sample .WAVS before importing. A good thing to do is use an Equalizer to have the frequencies above 1k gradually falloff to zero. Anything above 5k you can probably totally filter out. The goal is not to darken the sound, just filter out any low noise which might be getting picked up.

Beats are off time or don't sound as crisp

- Try setting FFTSize to 1024 and Time resolution to Very High. This will give you the best time resolution and sounds the best with beats.

I try to load a sound but I double click and nothing happens

- The sound may have failed to load. Make sure it is a 16 bit 44.1 or 22.05 khz sound (mono or stereo). Windows PCM format is the only format that can be loaded. Also make sure it's not longer than 30 seconds. Your machine may not have enough memory. Larger sounds require a lot of memory to analyze properly. Make sure you have the proper source selected - you may not hear any changes if you load a sound into slot 2 and aren't morphing.

DM takes up too much CPU!

- Resynthesis is naturally cpu consuming, but there are a few things you can do to save cpu. Try lowering the Number of partials used during analysis. This will create a sound that uses less CPU. You can also turn off FX.

Any other issues?

email: info@whitenoiseaudio.com

5.0 Thanks

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