Filter & Amp Sections

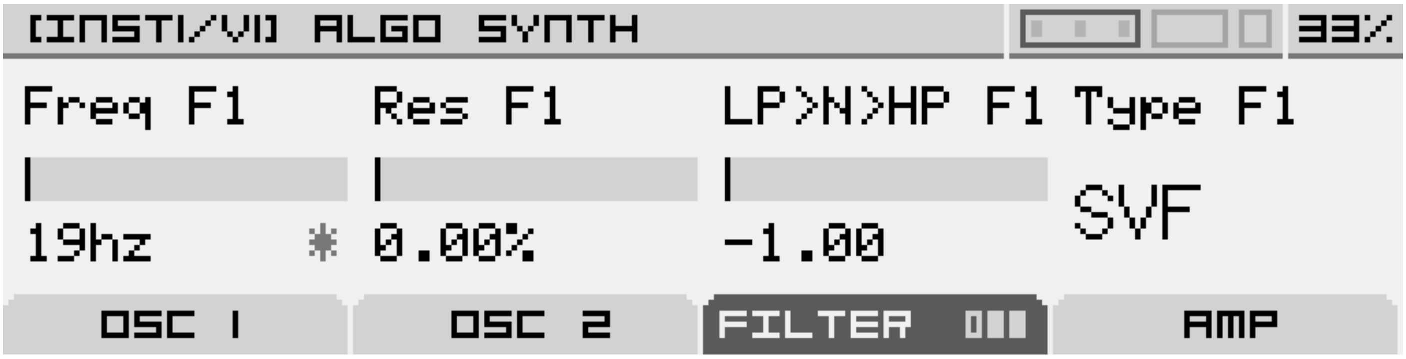
Filter Section

Within each Machine, the sound source is always routed through a similar Filter section, available on Tab 3.

Each Filter section then goes to an **[Amp section]()** before going to the **[Mixer]()**.

The Filter section is always on Tab 3 in every Machine.
This filter tab has 3 pages containing different controls. When a Tab header displays small bar icons, click its corresponding
button underneath to jump between its pages.

Main screen of Filter section



Page 1:

Filter 1 Frequency	**Filter 1 Resonance**	***(Filter Morph)*** ***-or-*** ***(Filter Gain)***	**Filter 1 Type**
Controls the filter's cutoff frequency	Controls the filter's resonance amount	*Morph between the filter types, from Low-Pass to Notch to High-pass. This parameter is only available when the filter type is set to SVF for State Variable Filter.* *-or-* *Set the EQ gain. This parameter is only available when the filter type is set to Bell EQ.*	Select a filter type and slope. Read the filter types reference below for details on every filter available. This parameter cannot be modulated.

Page 2 is the same as page 1, but controls Filter 2. In order to unlock it, you need to have the Routing on page 3 on a setting other than Single.

Filter types reference:

Off	The filter is bypassed
SVF	State Variable Filter model. Use Knob 3 to morph between filter types
K35 LP12 / HP6	Korg 35. Inspired by the MS-20 filter.
TLD LP 6/12/18/24	Transistor Ladder Filter model. Inspired by the classic Moog filter. Low-Pass with a selection of slopes from 6dB/oct to 24/dB/oct
TLD N 12/24	Transistor Ladder Filter model. Inspired by the classic Moog filter. Notch filter with 12dB/oct and 24dB/oct slopes
TLD BP 12/24	Transistor Ladder Filter model. Inspired by the classic Moog filter. Band-pass with 12dB/oct and 24dB/oct slopes
DLD LP24	Diode Ladder Filter model. Inspired by the TB-303 filter. Low-pass with a steep 24dB/oct slope.
COMB +/-	Comb filter for hollowed-out sounds and wooshes effects. With positive or negative feedback (resonance)
FORMANT	Formant filter for vowel sounds. Morph through A-E-I-O-U with Knob 1.
BELL EQ	Simple 1-band equalizer to increase or decrease a selected frequency region. Knob 2 will adjust the bell width and Knob 3 will set the gain

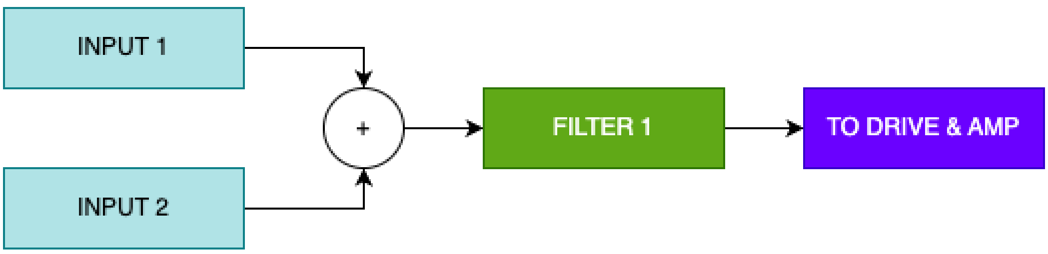
Page 3 allows for filter routing options:

Routing	**Balance between filter 1 and 2**	**-**	**-**

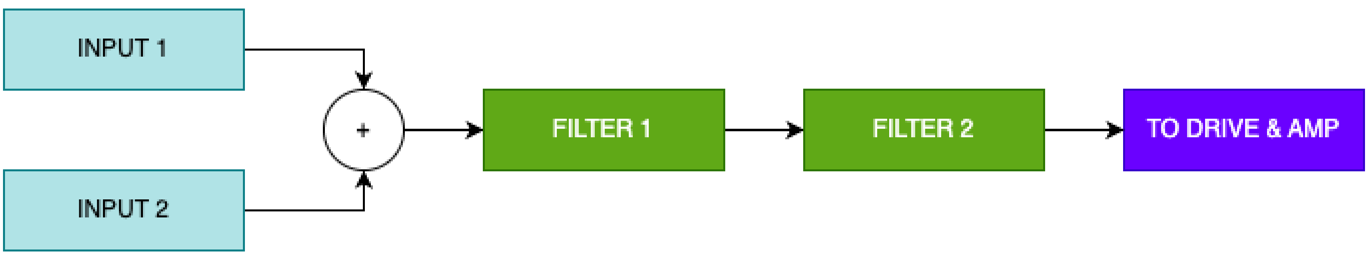
Select a routing configuration for the filters. Single enables only filter 1. Serial routes the output of Filter 1 to Filter 2. Para will route both filters in parallel Split will split the sound sources into the two filters, depending on the Machine select.	Controls the volume of both filters at the output. Fully clockwise only filter 2 will be heard, and fully counter-clockwise it will be only filter 1.	-	-

Filter routing

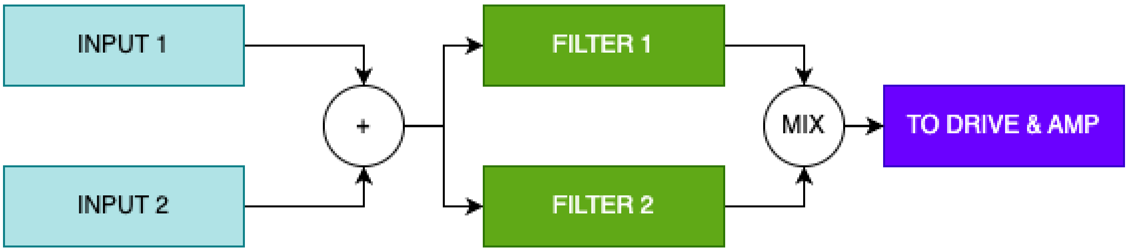
Single filter :



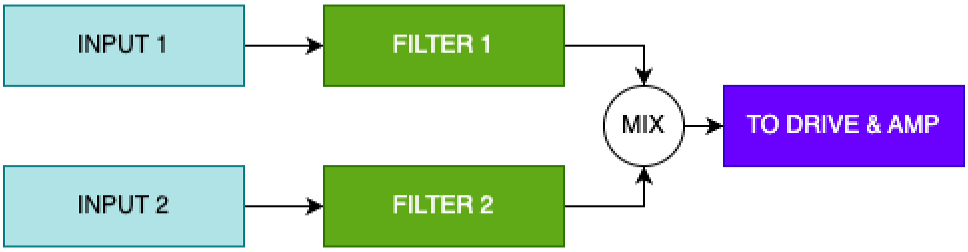
Serial filters :



Parallel filters :



Split filters :



Input reference (per machines types)

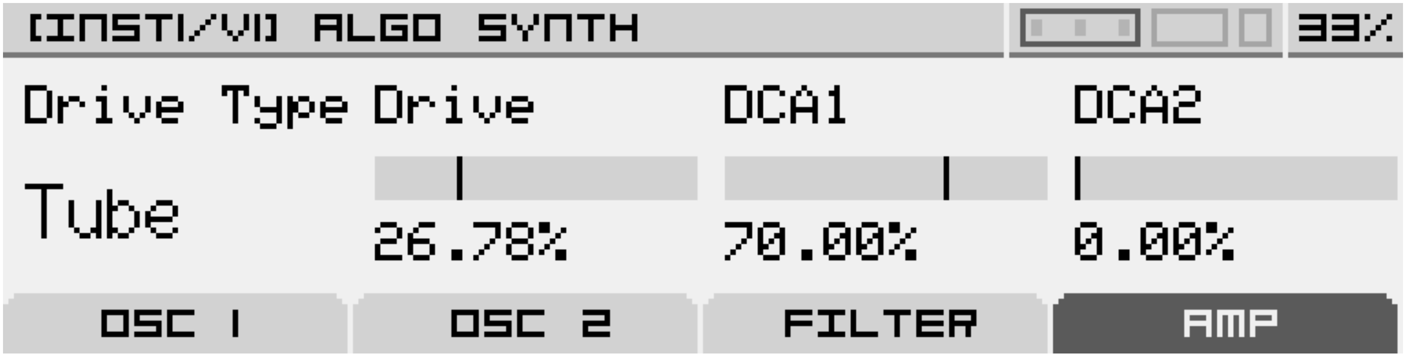
Machine	Input 1	Input 2
Algo synth	Oscillator 1	Oscillator 2
Wavetable synth	WT Oscillator + Noise	Sub oscillator
Sample player	Sample	Noise
Crossmod	Crossmodulated output	-

Amp Section

Within each Machine, the sound source goes through a Filter section and is then routed to the same Amp section.

Each Amp section is then routed to the **Mixer**.

Main screen of Amp section



The Amp section is always on Tab 4 in every Machine.
This amp tab offers settings related to amplitude and distortion:

Drive Type	**Drive**	***DCA1***	**DCA2**
Select one of the 29 distortion algorithms. Read the chart below for details regarding each one.	Controls the amount of distortion	The first Digitally Controlled Amplifier. DCA1 and DCA2 are routed in series, with 1 usually modulated by velocity and 2 modulated by an envelope. But you can set modulations as you wish.	The second Digitally Controlled Amplifier. DCA1 and DCA2 are routed in series, with 1 usually modulated by velocity and 2 modulated by an envelope. But you can set modulations as you wish.

Distortion algorithms reference

Type	**Description**
Off	Bypassed

Type	Description
Soft	Applies a soft saturation effect to an audio sample by adjusting the input sample's amplitude based on a calculated drive factor, using a hyperbolic tangent function for non-linear distortion
Medium	Applies a medium saturation effect to an audio sample by scaling the input sample's amplitude with a drive factor and using an arctangent function to achieve a smoother, non-linear distortion.
Hard	Applies a hard saturation effect to an audio sample by manipulating the sample's amplitude with a drive factor and employing a combination of hyperbolic tangent and arctangent functions for a more aggressive, non-linear distortion.
Diode	Applies a diode-based non-linear distortion to an audio sample. It scales the input sample by a drive factor, processes it through a diode model for non-linear distortion, and then blends the processed signal with the original signal based on the drive amount, including a volume compensation factor.
Demon	Applies a distortion effect to an audio sample by scaling the sample with a drive factor, processing it through a sine function, and then applying a diode-like non-linearity, blending the result with the original sample based on the drive amount
Soft Fold	Applies a soft folding distortion to an audio sample by scaling the sample with a drive factor, processing it through a sine function to create a folding effect, and blending the result with the original sample based on the drive amount
Diode Fold	Applies a diode-based folding distortion to an audio sample by scaling the sample with a drive factor, processing it through a diode model and a sine function to create a folding effect, and blending the result with the original sample based on the drive amount, including volume compensation
Dual Frequency	Applies a frequency-dependent distortion to an audio sample by splitting the sample into low and high frequency components using simple filters, applying different saturation levels to each band, and then blending the processed bands with the original sample based on the drive amount
Tube	Simulates tube-like distortion by scaling the input sample with a drive factor, applying a non-linear transformation to mimic the tube saturation effect, and blending the processed signal with the original sample based on the drive amount

Type	Description
Sigmoid	Applies a tube-like distortion using a sigmoid function to achieve smooth non-linear saturation, scaling the input sample with a drive factor and blending the processed signal with the original sample based on the drive amount.
Tape Dynamics	Applies dynamic saturation by using a high-pass pre-emphasis filter, followed by a hyperbolic tangent saturation, and then a low-pass de-emphasis filter, blending the processed signal with the original sample based on the drive amount.
Tape Hysteresis	Models tape hysteresis by simulating magnetic hysteresis behavior, adjusting the input sample based on coercivity and remanence factors, and applying a hyperbolic tangent function to saturate the magnetization, blending the processed signal with the original sample based on the drive amount.
Tape Curve	Applies a tape saturation effect by approximating a saturation curve, scaling the input sample with a drive factor, and using a non-linear transformation to mimic the characteristic response of tape saturation
Tape Noise	Simulates tape saturation with added noise by generating white noise, applying a hyperbolic tangent saturation to the noisy signal, and blending the processed signal with the original sample based on the drive amount, including volume compensation for higher drive levels.
Hard Clipping	Applies a hard clipping distortion to an audio sample by limiting the sample's amplitude to a threshold determined by the drive factor, normalizing the output, and blending the processed signal with the original sample based on the drive amount
Fuzz	Applies a fuzz distortion effect to an audio sample by scaling the sample with a drive factor, using an exponential function to create a non-linear distortion, and blending the processed signal with the original sample based on the drive amount
Chebyshev	Applies a series of Chebyshev polynomials to an audio sample, using a normalized drive factor to create a complex harmonic distortion effect, and then blends the processed signal with the original sample based on the drive amount
Half Rectifier	Applies a half-wave rectification effect to an audio sample by zeroing out negative values, scaling the rectified signal with a drive factor, and blending the processed signal with the original sample based on the drive amount
Full Rectifier	Applies a full-wave rectification effect to an audio sample by taking the absolute value of the input sample, scaling it with a drive factor, and blending the processed signal with the original sample based on the drive amount

Type	Description
Transistor	Simulates transistor-like saturation by scaling the input sample with a drive factor, applying a non-linear transformation to mimic transistor saturation characteristics, and blending the processed signal with the original sample based on the drive amount.
Dynamic	Applies a dynamic distortion effect to an audio sample by scaling the sample with a drive factor that is modulated by the sample's envelope, using a hyperbolic tangent function for non-linear distortion, and blending the processed signal with the original sample based on the drive amount.
Asymmetric	Applies an asymmetric clipping distortion to an audio sample by limiting the sample's amplitude to different positive and negative thresholds based on a scaled drive factor, normalizing the clipped signal, and blending it with the original sample based on the drive amount.
Feedback	Applies a feedback-based distortion effect to an audio sample by adding a feedback signal, scaled by a gain factor derived from the drive, to the input sample and then applying a hyperbolic tangent function for non-linear distortion, updating the feedback with the processed sample
Zero Crossing	Introduces distortion at zero crossings by adding a small spike to the audio sample whenever it crosses zero, with the spike's magnitude determined by a normalized drive factor, and updates the last sample for future comparisons.
Bit Reaper	Applies a bit reduction effect to an audio sample by scaling the drive factor, using it to determine a decimation factor, and then applying a bitwise reduction to the sample, followed by a non-linear saturation using a hyperbolic tangent function, blending the processed signal with the original sample based on the drive amount
Sample Reaper	Applies a sample rate reduction effect by holding the last sample value for a duration determined by the drive factor, updating the sample only when a counter exceeds a threshold, and blending the processed signal with the original sample based on the drive amount
Sample Reduction	Reduces the sample rate of an audio signal by holding the current sample value for a number of iterations determined by a drive-scaled reduction factor, effectively lowering the perceived sample rate
Bitwise	Applies a bitwise distortion effect to an audio sample by performing an XOR operation between the sample and a drive-scaled value, then normalizing the result and scaling it based on the drive amount

Type	Description
Ring Modulation	Applies a ring modulation effect to an audio sample by multiplying the sample with a sine wave at a frequency determined by the drive factor, updating the phase of the modulating signal to maintain continuous modulation