propellerhead

POLAR DUAL PITCH SHIFTER

OPERATION MANUAL

propellerhead

Polar Dual Pitch Shifter

Introduction

The Polar Rack Extension from Propellerheads is a dual pitch shifter unit. The concept of Polar is based on the classic harmonizing effect units of the 80's - with added features and possibilities.

In this day and age a harmonizing effect is a vocal "magic box" that harmonizes and formant-corrects the pitched harmonies - and also has advanced analysis and "intelligent" pitch shifting. Since we have already got the Neptune Pitch Adjuster and Voice Synth in Reason, we didn't go that path with Polar. Instead, we added lots of real-time control, modulation capabilities, plus a Filter section to make it an inspiring device for spicing up boring sounds - and of course for creating weird and spooky effects!

The main applications for Polar are:

- **Micro pitch shifting/thickening/widening**
- **Pitch shifted delay**
- **Twisted and modulated pitch shift effects**
- **Dramatic pitch shifting (special FX)**

Setting up for pitch processing

The examples below describe how to connect and use Polar in typical pitch processing situations.

Setting up for pitch processing of recorded audio tracks

The most flexible method of applying a pitch shift effect is to apply it to already recorded audio tracks in the sequencer. Doing so will give you total freedom to edit and change the pitch shift settings afterwards without needing to re-record any audio.

To set up Polar for pitch shifting of recorded audio tracks, proceed as follows:

1. **Select the Audio Track in the sequencer for the track you want to pitch process.**
2. **Create a Polar device.**
   The Polar device is created as an insert effect in the Audio Track device and is automatically connected to the appropriate connectors.

► **If you want to control the pitch shifting via your MIDI keyboard, create a sequencer track for the Polar device and select the Polar track in the Track List.**

Now, you are all set for pitch shifting the audio on the Audio Track. For information about pitch shifting, see "Setting up for basic pitch shifting".

Setting up for pitch processing of "live" audio

Polar can of course also be used for pitch shifting live audio (not recorded on any audio track). This is great for live performances and can also be used for recording pitch-processed audio on an audio track.

To set up Polar for pitch shifting of live audio, proceed as follows:

1. **Create a Mix Channel device.**
2. **Create a Polar device, ether by double-clicking the device icon in the Devices tab in the Tool Window or by selecting it from the Create menu.**
   The Polar device is created and routed as an insert effect in the Mix Channel device.
3. **Connect a microphone to your audio interface and manually patch the appropriate Audio in jack on the Reason Hardware Interface to the Left Input on the Mix Channel device.**

► **If you want to control the pitch adjustment via your MIDI keyboard, create a sequencer track for the Polar device and select the Polar track in the Track List.**

Now, you are all set for live pitch shifting. For information about pitch shifting, see "Setting up for basic pitch shifting".

If you want to record your processed audio on an audio track in the sequencer, perform the following additional steps:

4. **Create an Audio Track.**
5. **Click the Rec Source button on the Mix Channel device you use for Polar.**
6. **Select the Mix Channel device as input source in the Input Selector on the Audio Track in the sequencer.**
   This will route the pitch-processed audio from the Mix Channel to the Audio Track. This setup is identical to what you would use when recording with effects, as described in the "Recording in the Sequencer" chapter in the Reason Operation Manual pdf.

Using Polar

Loading and saving patches

Loading and saving patches is done in the same way as with any other internal Reason/Reason Essentials device - see the “Sounds and Patches” chapter in the Reason/Reason Essentials Operation Manual pdf for details.

Setting up for basic pitch shifting

1. **Set up for pitch processing as described in “Setting up for pitch processing”.**
2. **Select the “Smooth” or “Classic” Pitch Algorithm:**

“Smooth” produces a high-quality pitched signal whereas the “Classic” algorithm often produces a little “rougher” sound that resembles the classic harmonizing effect sound of the ‘80s.

3. **Next, select Slow, Medium or Fast based on the audio material you will be running through Polar:**

- **“Slow” is best suited for polyphonic audio and complex sounds.**
  “Slow” mode has the highest latency.
- **“Medium” is best suited for bassy monophonic sounds and also for drums.**
  “Medium” mode has a little higher latency than “Fast” mode.
- **“Fast” is best suited for less bassy monophonic sounds, like vocals.**
  “Fast” mode also has the lowest latency.

► **As a rule of thumb, use the fastest mode that makes your audio still sound good. Feel free to experiment to find the optimal setting for your specific audio material.**

The Pitch Algorithms are described in detail in “Algorithm”.

4. Turn the Shift and Fine knobs in the Pitch Shifter 1 section to detune the pitch:

→ If you want the original Dry signal to be present as well, activate the Dry section and turn up its Volume knob:

→ If you like you could increase the latency of the dry signal to make it "sync" with the Pitch Shifter 1 signal by turning the Delay knob:

Setting up for dual pitch shifting

1. Follow the instructions in the "Setting up for basic pitch shifting" paragraph.
2. Activate the Pitch Shifter 2 section:

3. **Turn the Shift and Fine knobs in the Pitch Shifter 2 section to detune the second pitch:**

Now, the input signal is detuned in two separate pitches.

→ **If you like you could pan the two pitch shifted signals hard left and right by turning the corresponding Pan knobs:**

→ **If you have detuned the pitches only slightly (using only the Fine knobs), you could turn up the Delay knob in the Pitch Shifter 2 section to widen the stereo image of the sound:**

The Delay parameter delays only the Pitch Shifter 2 signal. Set to a low value (but above 0) will give the impression of a very wide stereo image - if Pitch Shifter 1 and 2 are panned hard Left and Right respectively.

Controlling the pitch shifting from a MIDI keyboard

If you want to control the pitch shift(s) from your connected MIDI keyboard, this is how you do:

1. **Create a sequencer track for Polar by selecting "Create Track for Polar n" from the device panel context menu.**
   A sequencer track is created for the Polar device and is automatically selected in the sequencer Track List.
2. **Switch on the Keyboard Input button:**

3. **Play the connected MIDI Master keyboard to change the Pitch Shifter ratio.**

- **Both Pitch Shifter 1 and 2 are controlled from the same input MIDI Note in a monophonic fashion.**
- **The Pitch Shifters are tracked from the actual MIDI Note plus (or minus) the set Shift + Fine values.**
  Note that the pitch shift cannot exceed the default pitch shift limits +/-2 octaves relative to the input audio pitch.
- **The last played MIDI note holds the pitch shift ratio until you play a new MIDI note.**

Using the Delay Buffer

All audio to Polar passes through the Delay Buffer. This is a stereo delay which can be used in combination with the Feedback controls in the Pitch Shifter sections to create repeating pitch shifts.

1. **Turn up the Delay knob a bit:**

2. **Turn up the Feedback knobs in the Pitch Shifter section(s) to hear the pitch shifts repeat:**

- **Using feedback with delay on pitched-down audio produces continuously decreasing pitches.**
- **Using feedback with delay on pitched-up audio produces continuously increasing pitches.**

► **You can also lock the audio currently in the Delay Buffer and "sweep" through the audio material to create really far out effects, see "Lock (Buffer)".**

Parameters

Input section

On/Off/Bypass

This switch is located in the upper left corner of the device. The switch has three modes:

Mode	Description
Bypass	In this mode, the input signal is passed directly to the audio output, without being affected by the effect device. This is useful when the effect device is connected as an insert effect, and you want to compare the effect sound with the dry sound.
On	This is the default mode, in which the device processes the incoming signal.
Off	In this mode, the effect device is turned off and neither dry nor effect sound is sent out. This is useful when the device is connected as a send effect and you want to turn it off temporarily.

Algorithm

Select the type of algorithm you want to use for the audio analysis and resynthesis:

Smooth

This algorithm uses advanced real-time frequency analysis and resynthesis to generate smooth and high quality pitched signals out of complex input signals. There are separate analysis + resynthesis channels for stereo signals. The Smooth algorithm is best suited for polyphonic input audio.

Classic

This algorithm looks for suitable positions in the audio to generate grains (short waveform snippets). The grain lengths vary dynamically according to the input audio to preserve the pitch. The Classic algorithm is best suited for periodic monophonic waveforms like strings, vocal, etc. The Classic algorithm is what was used by the classic hardware harmonizing effect units of the '80s.

Slow/Medium/Fast

Here you determine how accurate the pitch analysis should be. The analysis is made continuously on dynamic snippets of the input audio. The more complex and/or bassy the input audio, the longer the snippets have to be for a correct pitch analysis. The detection time of these audio snippets is selected with the Slow/Medium/Fast knob.

► **Use the fastest option that makes your audio still sound good. Feel free to experiment to find the optimal setting for your specific audio material.**

- **Slow**
  This is best suited for polyphonic audio and complex sounds. “Slow” mode has the highest latency (typically around 96ms) because it needs time to correctly analyze the frequencies in complex sounds.
- **Medium**
  This is best suited for bassy monophonic sounds and also for drums and percussion. “Medium” mode has a little higher latency than “Fast” mode (typically around 48ms) because it should also be able to accurately detect lower frequencies.
- **Fast**
  This mode is best suited for less bassy monophonic sounds, like vocals. “Fast” mode has the lowest latency (typically around 24ms) because it’s not optimized to detect low frequencies.

! **Note that the Slow/Medium/Fast knob isn’t necessarily a “sound quality” control. E.g. for vocals the best sound quality is often achieved with the Classic+Fast combination.**

► **For polyphonic audio, the Smooth+Medium/Slow combination is often the best setting.**

Loop

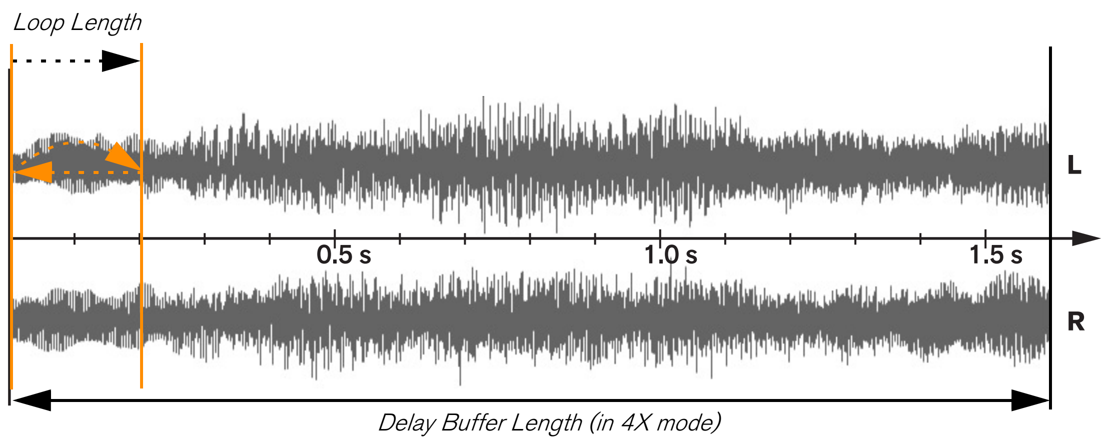
This algorithm lets you define your own fixed loop length (or “grain length” if you whish) with the “Length” knob. Since the loop length is fixed, using very short Loop Length settings can result in varying pitches depending on the pitch variations of the input audio material.

Reverse Loop

This algorithm works exactly like the Loop algorithm described above, except it plays back the audio loop reversed. This algorithm is perfect for creating really spooky vocal effects - especially if you turn the “Length” knob up a bit!

Length

If you have selected the “Loop” or “Reverse Loop” Pitch Algorithms described above, you set the loop length with the Length knob. The loop range in the audio input Delay Buffer is set according to the figure below:



Depending on if you selected the “Loop” or “Reverse Loop” Pitch Algorithm, the loop will run in different directions, forward or backwards.

The loop Length range is: 0-1.0s.

Delay Buffer section

All audio that is input to Polar passes via the Delay Buffer. The Delay Buffer can be used for setting the feedback delay time for both Pitch Shifters sections. You can also manually play back audio from the Delay Buffer by locking it and controlling the playback position within the Delay Buffer, see "Lock (Buffer)".

! **Note that changing the Delay time/Buffer Position does not affect the pitch of the audio, which is great if you want to create stutter effects, for example!**

Delay/Buffer Position and 4X

This controls the delay time for the input audio and for the Pitch Shifter 1 and 2 feedback parameters.

- **In the default mode the delay buffer range is 400ms.**

→ **Click the 4X button to the right of the Delay knob to quadruple the delay buffer range to 1600ms.**
The set Delay time/Buffer Position value is automatically quadrupled as soon as the 4X button is engaged.

! **Note that the Delay knob tool tip will always show a time in the 0-400ms range, regardless if the 4X button is on or off.**

! **Note that at the lower end of the range, the Delay knob will only have an effect once it has reached the latency time introduced by the Slow/Medium/Fast parameter (see "Slow/Medium/Fast"). E.g, if you have set the Slow/Medium/Fast parameter to Fast, then it will not be possible to set a delay of less than 24ms.**

In "Lock" mode, the Delay knob controls the playback position within the Delay Buffer, see "Lock (Buffer)".

LFO

→ **Turn this knob if you want the Delay time/Buffer position to be modulated by the LFO.**
The parameter is bipolar which means that you can define the initial "direction" of the delay time shift - shorter or longer.

The LFO parameters are set in the LFO section, see "LFO section".

Env

→ **Turn this knob if you want the Delay time/Buffer position to be modulated by the Envelope generator.**
The parameter is bipolar which means that you can define the "direction" of the delay time/position shift:

- **A positive value makes the Delay time shorter, or moves the Buffer Position forwards in "Lock Buffer mode, and then reverts to the current Delay knob setting.**
- **A negative value makes the Delay time longer, or moves the Buffer Position backwards in "Lock Buffer mode, and then reverts to the current Delay knob setting.**

The Envelope parameters are set in the Envelope section, see "Envelope section".

Lock (Buffer)

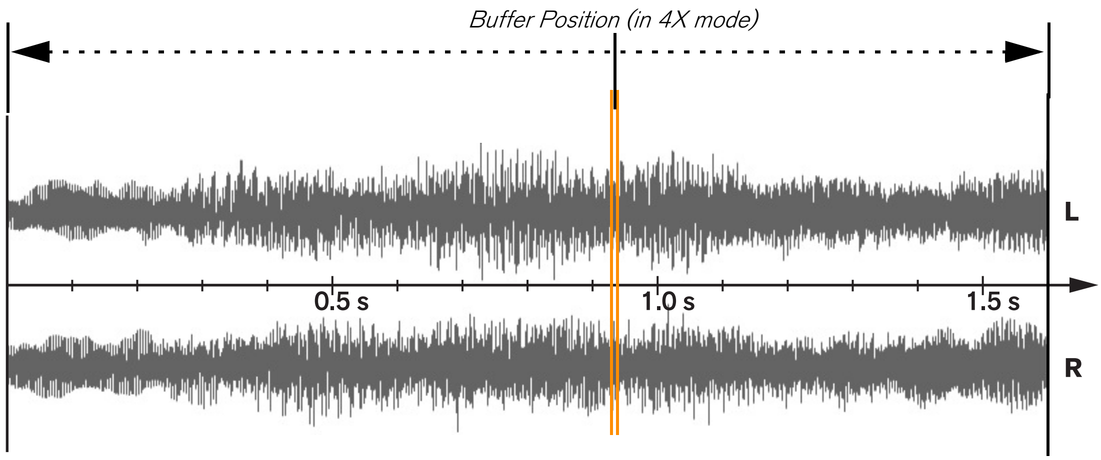
With the Lock button switched on, you temporarily "freeze" the audio that is currently in the Delay Buffer, and no new audio is let into the Buffer. You can now play back the audio from the Buffer by turning the Delay knob up and down.

! **Note that as soon as you turn off the Lock Buffer button, the "freezed" audio is replaced in real-time by the audio coming in to Polar at the moment.**

! **Note that the Delay Buffer range can be extended from 400ms to 1600ms by pressing the 4X button to the right of the Delay knob.**

Depending on which Pitch Algorithm (see "Algorithm") you have selected, the playback result will be different. The figures below show what happens in the different scenarios:

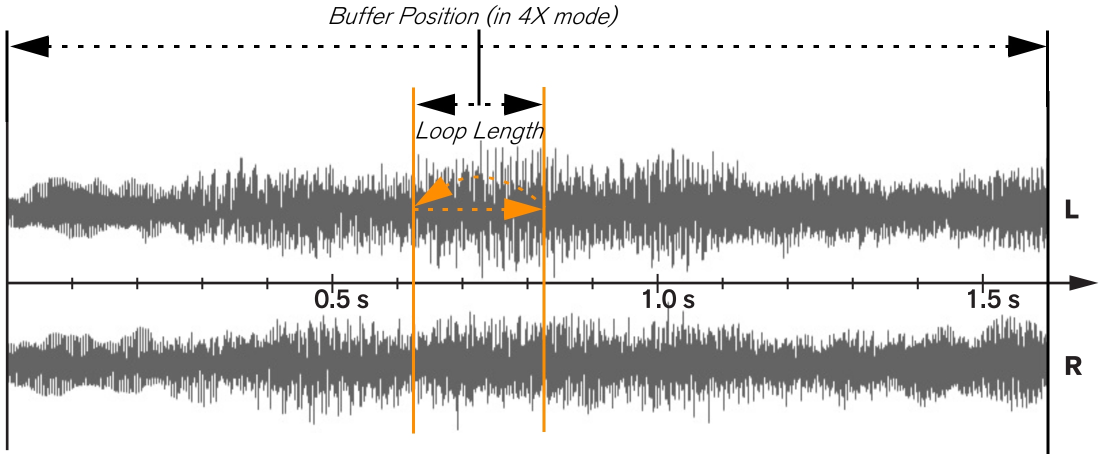
- **The "Classic" and "Smooth" pitch algorithms:**



By turning the Delay knob you can move the playback position within the Delay Buffer. When you stop turning the knob, the playback is performed by looping a grain (Classic mode) or by performing real-time frequency analysis + resynthesis (Smooth mode) at the current position in the buffer.

! **Note that turning the Delay knob steadily up (clockwise) will play back the audio backwards, and turning the knob steadily back down again (counter-clockwise) will play back the audio forwards.**

- **The "Loop" and "Loop Reverse" pitch algorithms:**



By turning the Delay knob you can move the loop position within the Delay Buffer. You set the loop length with the "Length" knob and define the loop type with the "Loop" and "Reverse Loop" buttons.

! **With longer loops there might not be any audible changes of the sound when you start to reach the end positions of the Buffer. This is because the loop start/end will reach the Buffer's start/end position before the Delay knob does.**

Sum L+R

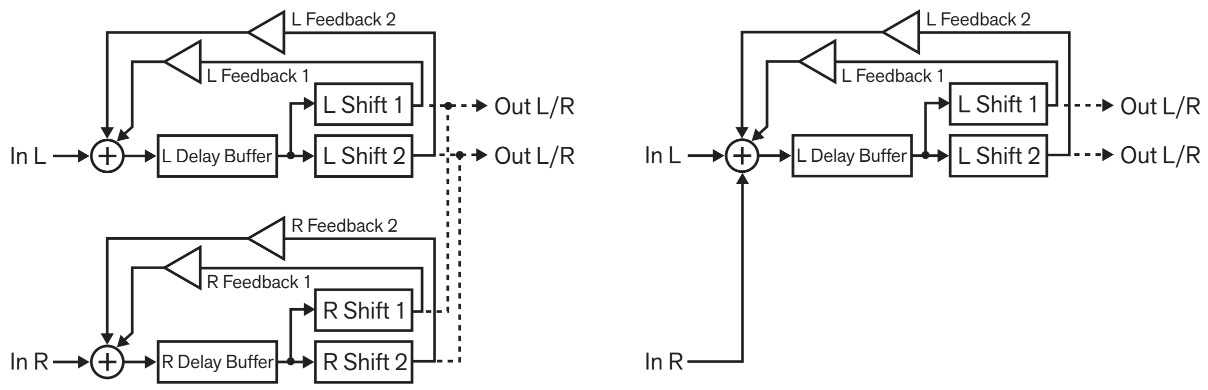
Click this button to make Polar perform the pitch analysis and resynthesis on the sum of the Left and Right input signals. This saves processing power since only one (merged) input signal is analyzed and processed. The other processing channel is automatically switched off. You can still pan the processed sound on the Polar outputs as desired to create a stereo output effect.

The Sum L+R mode is suitable in two scenarios:

- **If only the Left Audio Input of Polar is connected because you are processing a mono sound.**
- **If you are processing a stereo sound that has similar character on the left and right channels.**
  For example, this could be a stereo synth pad, a stereo piano or basically any "single instrument" sound.

! **The Sum L+R mode is not suitable if the stereo sound you are processing has different character on the left and right channels. Also, if you want to preserve the original panning of the input stereo sound you are processing, make sure the Sum L+R button is Off.**

The schematic illustration below shows the signal flow in the default “true stereo” mode and in the Sum L+R mode:



The default “true stereo” mode to the left and the “Sum L+R” mode to the right.

Pitch Shifter 1&2 common parameters

Shift

→ **Set the pitch shift amount with the Shift knobs.**
Range: +/- 2 octaves (relative to the current input audio pitch).

Fine

→ **Fine tune the pitch shift with the Fine knobs.**
Range: +/- 0.5 semitones, in steps of 1 cent.

Feedback

→ **Set the feedback amount of the pitch shift with the Feedback knobs.**
Range: 0-100%.

→ **Set the feedback delay time with the Delay knob, see “Delay/Buffer Position and 4X”.**

- **Using feedback with delay on pitched-down audio produces continuously decreasing pitches.**
- **Using feedback with delay on pitched-up audio produces continuously increasing pitches.**

LFO

→ **Turn this knob if you want the pitch shift to be modulated by the LFO.**
The parameter is bipolar which means that you can define the initial direction of the pitch shift - up or down.

The LFO parameters are set in the LFO section, see “LFO section”.

Env

→ **Turn this knob if you want the pitch shift to be modulated by the Envelope generator.**
The parameter is bipolar which means that you can define the “direction” of the pitch shift:

- **A positive value makes the pitch go up and then down to the Shift setting.**
- **A negative value makes the pitch go down and then up to the Shift setting.**

The Envelope parameters are set in the Envelope section, see “Envelope section”.

Pan

→ **Set the position of the pitch shifted signal in the stereo panorama.**

Auto (Pan)

→ **Set the amount of auto pan for the pitch shifted signals.**
The parameter is bipolar which means that you can define the initial direction of the panning effect. This is very useful if the other Pitch Shifter and/or the Dry signal also use the auto pan function.

The Auto Pan rate is set with the Auto Pan Rate control, see “Auto Pan”.

Volume

→ **Set the level of the pitch shifted signals.**
Range: -inf to 0dB

Pitch Shifter 2 exclusive parameters

On/Off

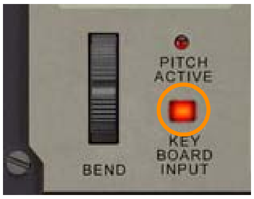
→ **Press this button to switch on/off the Pitch Shifter 2 section.**

Delay

The Delay parameter delays the Pitch Shifter 2 signal. This can be used for generating very wide pseudo-stereo effects when Pitch Shifter 1 and 2 are panned hard left and right respectively and are only slightly detuned.

Range: 0-500ms.

Keyboard Input



Polar responds to MIDI Note data. MIDI Notes can be used for controlling the pitch shift amount and also for triggering the Envelope section (see "Envelope section").

To control the Pitch Shifter pitch(es), proceed as follows:

1. **Create a sequencer track for Polar by selecting "Create Track for Polar n" from the device panel context menu.**
   A sequencer track is created for the Polar device and is automatically selected in the sequencer Track List.
2. **Switch on the Keyboard Input button.**
3. **Play the connected MIDI Master keyboard to change the Pitch Shifter pitches.**

- **Both Pitch Shifters are controlled from the same input MIDI Note in a monophonic fashion.**
- **The Pitch Shifters are tracked from the actual MIDI Note plus (or minus) the set Shift + Fine values.**
- **The last played MIDI note holds the pitch shift ratio until a new MIDI note is received.**

Dry section

Here are the controls for the dry, unprocessed signal.

On/Off

→ **Click this to let the dry signal through to the outputs.**

Delay

→ **Set the delay time of the dry signal.**
By turning the Delay knob you manually apply latency to the dry signal. This is useful if you want to "sync" the dry signal with the signals of the Pitch Shifters.

Range. 0-250ms.

Pan

→ **Set the position of the dry signal in the stereo panorama.**

Auto (Pan)

→ **Set the amount of auto pan for the dry signal.**
The parameter is bipolar which means that you can define the initial direction of the panning effect. This is very useful if the Pitch Shifters also use the auto pan function. The Auto Pan rate is set with the Auto Pan Rate control, see "Auto Pan".

Volume

→ **Set the level of the dry signal.**
Range: -inf to 0dB

Auto Pan

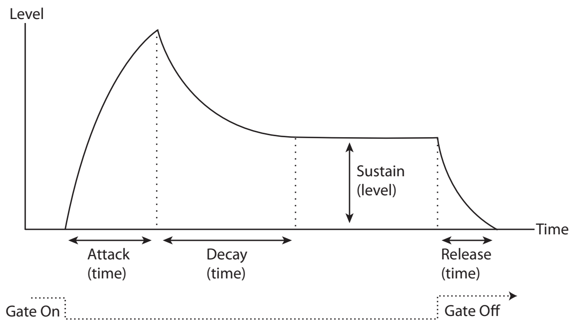
Here you set the rate for the Auto Pan function. The auto pan waveform is a fixed sinewave and the LEDs above the Rate knob indicate the current rate. The modulation amount and initial panning direction can be set individually in the corresponding destination sections. Range: 0.10-13.3Hz

► **If you like you could route a CV signal to the Auto Pan CV input on the rear panel and completely override the default waveform and rate, see "Auto Pan".**

Envelope section

The Envelope can be used for modulating various parameters in Polar. The ADSR (Attack-Decay-Sustain-Release) envelope can be triggered/gated by clicking/holding the Gate button. The envelope can also be triggered/gated by using the CV Gate In modulation input on the rear panel, or by sending MIDI Notes from a connected MIDI keyboard.

The figure below shows the principle of an ADSR envelope, with the stages marked out:



The different stages of an ADRS envelope

Gate

Click the Gate button to gate the envelope. The envelope can also be gated from a connected MIDI keyboard - if you have created a sequencer track for Polar and the track has keyboard focus - or by sending a CV signal to the Gate In input on the rear panel, see "Gate".

! **If the envelope is retriggered before all envelope stages are completed, the envelope will simply restart at the current level (similar to how a monophonic synthesizer works).**

A(ttack)

When you trig the envelope the envelope level rises from zero to maximum. The time it takes to reach maximum level is set with the A(ttack) slider.

Range: 0.0ms-3.00s.

D(ecay)

After the maximum value has been reached, the value starts to drop. How long this should take is set with the D(ecay) slider.

Range: 0.0ms-10.0s.

S(ustain)

The Sustain parameter determines the level the envelope should rest at after the Decay stage, given that the Trig button is still depressed. If you set Sustain to full level, the Decay setting is of no importance since the level is never lowered.

Range: 0-100%

R(elease)

This works just like the Decay parameter, except it determines the time it takes for the value to fall back to zero *after* releasing the Trig button.

Range: 0.0ms-10.0s.

Env to Amp

Click this button if you also want the output volume from Polar to be modulated by the Envelope.

LFO section

The LFO can be used for modulating various parameters in Polar. Here you set the LFO parameters:

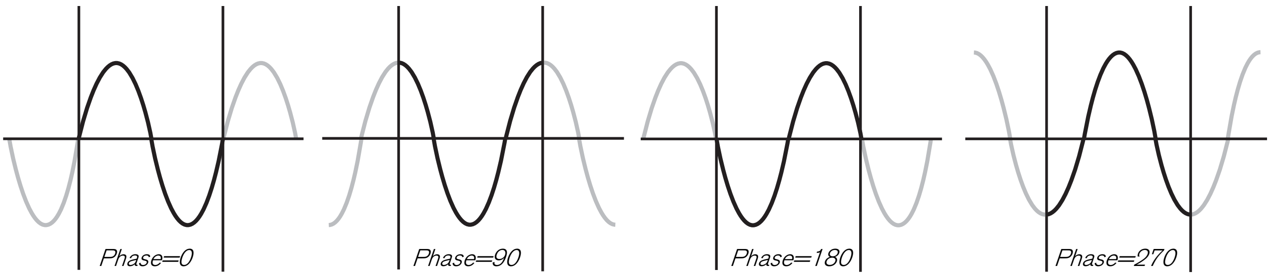
Rate

Sets the LFO rate. The LEDs above the Rate knob indicate the current rate. The rate can also be synced to the sequencer tempo by clicking the Tempo Sync button below the Waveform selector (see “Tempo Sync”).

Range: 0.10-13.3Hz (synced: 32/4 to 1/64th)

Phase

The Phase control lets you offset the phase of the LFO cycle, i.e. decide where in the cycle the waveform should start. The range of the Phase control is 0-360 degrees:



Waveform selector

Here you select one of six LFO waveforms. The waveforms are:

- **Triangle**
- **Ramp up**
- **Ramp down**
- **Square**
- **Random**
- **Soft random**

The shape of the selected waveform is shown in the display.

! **Note that all waveforms are bipolar, i.e., they generate both positive and negative levels.**

→ **Select waveform by clicking the up/down arrow buttons or by clicking and holding in the waveform display and dragging up/down.**

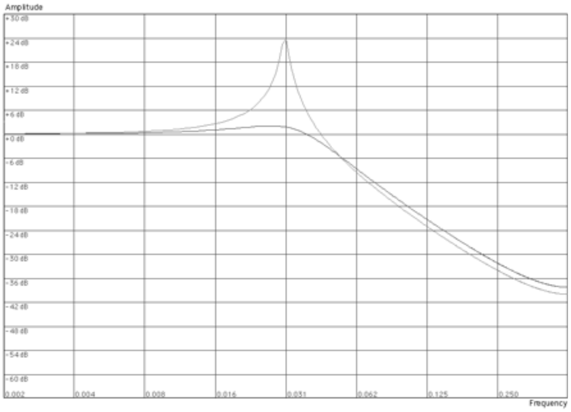
Tempo Sync

Click the Sync button to sync the LFO Rate to the sequencer tempo. In Sync mode, the Rate knob controls the sync resolution, see “Rate”.

Filter section

The Filter section comprises a multi-mode stereo filter with four different filter types:

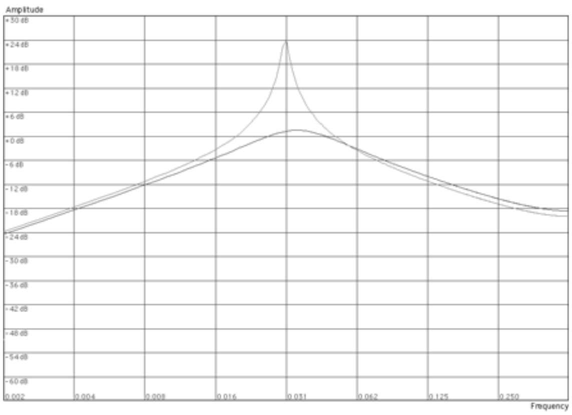
- **Lowpass (LP)**



A 12 dB Lowpass filter with low and high Resonance settings

Lowpass filters let low frequencies through and cut off high frequencies. This filter type has a roll-off curve of 12dB/Octave.

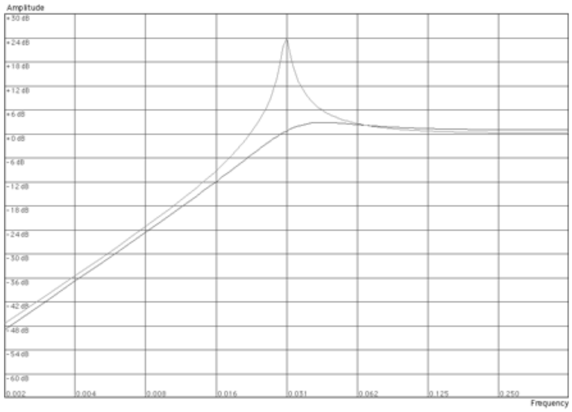
- **Bandpass (BP)**



A 12 dB Bandpass filter with low and high Resonance settings

Bandpass filters cut both high and low frequencies, leaving the frequency band in between unaffected. Each slope in this filter type has a 12 dB/Octave roll-off.

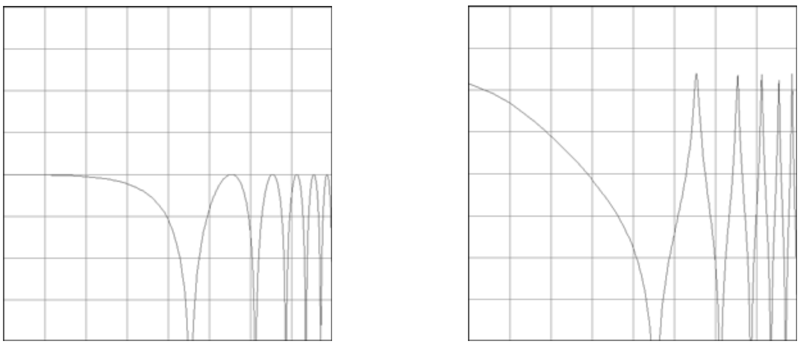
- **Highpass (HP)**



A 12 dB Highpass filter with low and high Resonance settings

Highpass filters let high frequencies through and cut off low frequencies. This filter type has a roll-off curve of 12dB/Octave.

- **Comb**



A Comb filter with low and high Resonance settings respectively

Comb filters are basically delays with very short delay times with adjustable feedback. A comb filter causes resonating peaks at certain frequencies. The Resonance parameter controls the shape and size of the peaks. The Comb filter in Polar is positive (+), which means that low frequencies are let through.

On/Off

→ **Click this to enable the Filter section.**

Freq

→ **Set the cutoff frequency (LP, HP and Comb modes) or the center frequency (BP mode) with the Freq knob.**
Range: 61.0Hz to 12.69kHz

Res

→ **Set the resonance level with the Res knob.**
Range: 0-100%

LP/BP/HP/Comb

→ **Select filter type by clicking the corresponding button.**

Kbd

With the Kbd (Keyboard Track) knob you define if, or how much, you want the filter cutoff to be affected by incoming MIDI Note values. At maximum value the keyboard tracking is 1:1, which means the filter cutoff tracks the pitch of the incoming MIDI Notes to 100%.

Range: 0-100%

LFO

→ **Turn this knob if you want the cutoff/center frequency to be modulated by the LFO.**
The parameter is bipolar which means that you can define the initial direction of the cutoff modulation - up or down.

The LFO parameters are set in the LFO section, see “LFO section”.

Env

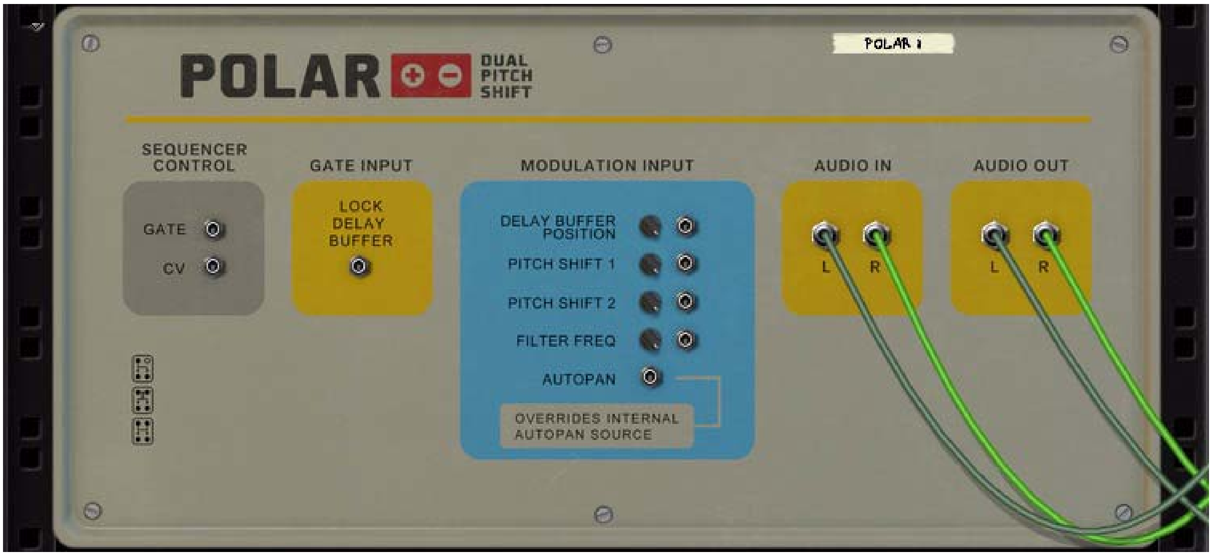
→ **Turn this knob if you want the cutoff/center frequency to be modulated by the Envelope generator.**
The parameter is bipolar which means that you can define the “direction” of the cutoff modulation - up or down:

- **A positive value makes the cutoff frequency go up and then down to the current Freq setting.**
- **A negative value makes the cutoff frequency go down and then up to the current Freq setting.**

The Envelope parameters are set in the Envelope section, see “Envelope section”.

Connections

! **Remember that CV connections will not be stored in the Polar patch!**



Sequencer Control

The Sequencer Control CV and Gate inputs allow you to control Polar from another CV/Gate device (typically a Matrix or an RPG-8).

Gate

The signal to the Gate input delivers note on/off.

CV

The signal to the CV input controls the pitch shift of Pitch Shifter 1 and 2 chromatically. This means you can “play” the pitch shift with Note CV signals from other devices. The control signal modulate Pitch Shifter 1 and 2 together, in a monophonic fashion, so that the pitch shift relationship is always maintained.

Gate Input

Lock Delay Buffer

A Gate signal on this input activates the Lock function, see “Lock (Buffer)”.

Modulation Input

These control voltage (CV) inputs (with associated trim pots) can modulate following parameters in Polar:

Delay/Buffer Position

A CV signal on this input controls the Delay time or - if Lock Buffer is active - the playback position in the locked Buffer.

! **A CV signal on this input can modulate the full Delay/Buffer Position range (0-1600ms) regardless if the 4X button is on or off, see “Delay/Buffer Position and 4X”.**

Pitch Shift 1 and 2

CV signals on these inputs modulate the pitch shift amount of Pitch Shifter 1 and 2 respectively.

Filter Freq

A CV signal on this input controls the Frequency parameter in the Filter section.

Auto Pan

A CV signal on this input completely overrides the Auto Pan rate and default sinewave waveform. This means you can take a signal from an external LFO (or other device), for example a square wave, route it to the Auto Pan input and control the auto pan characteristics and rate completely from the external source.

The Auto Pan CV input accepts bipolar input signals, where the level 0 represents the center position in the panorama.

Audio In

These are the stereo audio inputs. If you want to process a mono signal, connect only to Audio Input L.

Audio Out

These are the stereo audio outputs.

► **If you process a mono signal on the left Audio Input, you can still get the sound out in stereo if you connect both Audio Outputs.**

Additional external control

MIDI Notes

Polar responds to MIDI Note information, see “Keyboard Input”.

Pitch Bend

Polar responds to Pitch Bend data from the pitch bend control of your MIDI master keyboard.

The range is fixed at +/-1 octaves.