MATERIA PRIMA

Sound Design Engine & Library

USER MANUAL

VERSION 1.0

*materia pri·ma | \ - ˈprīmə *

primary material | first matter | prōtē hylē

indeterminate matter viewed as the material cause of the universe

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REQUIREMENTS

- Materia Prima requires the **full version*** of Native Instruments software sampler, Kontakt.
  It **is not compatible** with the FREE version of Kontakt Player.
- Versions of Kontakt 6.7.1 and above are supported (tested up to version 8.1.0).
- All major DAWs supported.
- Mac OS 10.13 High Sierra or higher. Windows 10 or higher. 32-bit systems are not supported.
- 8GB RAM minimum.
- At least 3 GB of free drive space.

INSTALLATION

Materia Prima doesn't require any complex installation. Simply extract the zip file to the location of your choosing on your computer (an SSD drive is preferable for maximum performance) and then open a Instrument or Multi from the Files menu in the full version of Kontakt.

If you wish to import the library into the Kontakt 7 or 8 Library view, Copy the "Materia Prima v1" folder from the "Install/Library browser images" folder into the appropriate place, depending on your operating system.

WINDOWS:
Copy into "C:\Users\Public\Documents\NI Resources\image\"

MAC:
Copy into "Macintosh HD > Users > Shared > NI Resources > image"

You can follow instructions here to add Materia Prima to the Kontakt library browser:
https://support.native-instruments.com/hc/en-us/articles/6677339715741-How-to-Add-Non-Player-Libraries-to-Kontakt-7-8-s-Browser

Adding Materia Prima to the browser also enables you to use the audio preview feature of each patch within the library. You can also browser the library using Native Instruments Kontrol S series keyboards using the Komplete Kontrol software.

**Materia Prima requires the FULL version of Kontakt. It will not work with the FREE Kontakt Player. The FREE Kontakt Player will only load and play Materia Prima for 15 minutes in DEMO mode.*

INTRODUCTION

First, a sincere thank you for your purchase. Materia Prima is our first sample library (of many hopefully). We hope you find as much joy and inspiration from it as we've had making it! We know there are many incredible tools and libraries available to you, so it means a great deal that you chose to add Materia Prima to your collection. Your support not only makes projects like this possible, it motivates us to keep creating and pushing forward. We believe music is about connection—between sound and emotion, between artist and listener, and between creators like us and musicians like you. Thank you for letting Materia Prima be a part of your creative journey.

- Chad Seay, Seay Interactive

P.S. If you create something really cool with Materia Prima, share it with us and we'll let the world know in our Creators Showcase. Visit us online at seayinteractive.com/audio/creators-showcase for more details!

With that, let's dive in and explore everything Materia Prima has to offer.

INSTRUMENTS

Instruments can be found in the *Instruments* folder of the library in the following categories:

- Bass
- FX
- Keys
- Lead
- Organic
- Pad
- Percussion
- Pluck
- Pulse
- Synth

MULTIS

Multis are found in the *Multis* folder of the library. These are sounds created by combining different instrument presets together to create either a whole new sound, or a playable "scene". For example, a scene might have percussion or pulsing sounds playable on lower notes and solo or pad instruments in higher notes.

EXTRAS

In the *Extras* folder you will find…

- A starter "init" version of Materia Prima with default settings minus any samples.
- The BYOS ("Bring Your Own Samples") or "Drag and Drop" edition for those who would like to go beyond the presets and develop their own presets using their own sample and the Materia Prima engine.
- In the *Waveforms* folder, we've also included 266 WAV files to get you started developing your own sounds using the BYOS edition! These include basic synth waveforms, a selection of single cycle waveforms, as well as one shots and looped files from the Materia Prima library.

FEATURES

- Collection of 230 presets in 10 categories, and multis (204 Instruments and 26 Multis)
- 2.7 GB download, 2,525 NCW lossless compressed samples (5 GB uncompressed wav)
- Up to 3 voice layers per preset, each with individual volume, panning, and +/- 3 octave tuning
- Individual AHDSR (Attack, Hold, Decay, Sustain, Release) control over each voice layer (or simplified global control over all 3 layers)
- Global pitch envelope with AHDSR control
- Global low, high or band pass filters, with AHDSR and velocity control
- 16 independent, simultaneous effects modules (depending on CPU performance)
    - 4 distortion effects (Tube, Lo-Fi, Classic Amp, Modern Amp)
    - 3 modulation effects (Phaser, Flanger, Chorus)
    - 1 "Time" effect (Delay)
    - 4 "Space" effects (Plate, Room / Hall, Convolution)
    - 5 "Output" effects (Tape Saturation, Stereo Modeler, Transient Designer, Compressor, Equalizer)
- "Pulse" Engine (with DAW tempo sync or Hz rate selection)
    - Individual Gain, Pan, and Pitch LFOs
    - Step Sequencer
    - Pre and Post-FX Filters ,each with their own Step Sequencer
    - 2 additional Pre-FX LFOs
- Access to an extensive library of 713 Impulse Response files for the Convolution reverb module, including recordings of:
    - real rooms, chambers, halls, cathedrals, and auditoriums
    - plates
    - speaker cabinets
    - special effects
    - professional digital reverbs units from Bricasti, Lexicon, Quantec, and TCE
- 80 IRs exclusive to the Materia Prima library
- Aftertouch vibrato support (monophonic), with controllable depth and rate
- Assignable Pitch Bend range
- MIDI Learn (easily assign controls to a MIDI device and access them via MIDI controllers)
- Help system with guidance for every control
- "BYOS" (Bring Your Own Sample) version, with drag and drop sample support for 3 voice layers. Includes a selection of over 250 WAV files ready for drag and drop.
  Each "BYOS" voice layer features:
    - MIR support (music information retrieval / auto root note detection) or manual root note selection
    - One-shot or looped file support
    - Manual selection of voice key range (low key, high key)

CONTROLS & USER INTERFACE

1. Knobs & Vertical Sliders
   Clicking and dragging up and down raises and lowers its value.
2. Horizontal Sliders
   Clicking and dragging left and right lowers and raises its value.
3. Buttons, checkboxes, and toggles
   Clicking these will toggle its value on or off.
4. Menus
   Clicking will open the menu and reveal a list of items to choose from. With the menu open, hover over an item and click to select it and close the menu.
5. Message Center
   Towards the bottom left corner of the instrument, above the Materia Prima logo, you will find the Message Center. When a control is moved in the instrument, this area will show the name of the control and its current value.
6. Instrument Navigation
   These buttons switch between the different sections of the instrument: Source, different FX types, the Pulse engine, and finally the Output section.
7. Options Menu
   Here you can enable, disable, and adjust the aftertouch vibrato feature. In the "BYOS" version of the instrument, this is where you can add and manage your own samples in each of the 3 layers.

- Continuous Controller Mapping
    - Almost every control within the instrument can be mapped to a MIDI CC value. Simply right click any control and select “Learn MIDI CC automation”. Immediately after, move the slider or knob on your controller and it will automatically be assigned to the control within Materia Prima.
- Help System
    - When hovering over any control within the instrument, help text will appear within the Info panel in Kontakt. NOTE: Make sure the Info panel in Kontakt is enabled and visible (F9) to be able to see the help text.

MAIN

The Main section is the initial and primary view of the instrument. Here you can change tuning, pan, and volume of each individual sound layer, as well as shape the overall sound details.

Layers

Each sound can be made up of up to 3 layers. Use this section to enable or disable layers, adjust their balance, and create complex tonal blends.

Amplifier

The amplifier section shapes the volume contour of the sound using an AHDSR envelope.

Pitch

The pitch section lets you apply an AHDSR envelope to pitch, for dynamic sweeps and bends.

Filter

The filter section sculpts the harmonic content of the sound using one of three filter types and its associated controls.

Layer Controls:

- PAN: Adjusts the stereo position of the selected layer.
- MUTE (M): Silences the selected layer.
- SOLO (S): Isolates the selected layer so only it is the only one heard.
- GAIN: Sets the output level of the selected layer.
- TUNE / FINE: Transposes the pitch of the selected layer in semitones (TUNE) or cents (FINE).

Amplifier Controls:

- (A) ATTACK: The time it takes the sound to reach full volume.
- (H) HOLD: How long the sound stays at full volume after attack.
- (D) DECAY: How quickly the volume falls after the hold stage.
- (S) SUSTAIN: The level held while a key is pressed.
- (R) RELEASE: How long the sound fades after the key is released.
- (V) VELOCITY: Adjusts how much playing velocity effects the amp settings.
- LINK: When selected, you can adjust all layers at once instead of individually.

Pitch Controls:

- AHDSR: This works the same as in the Amplifier but affects pitch instead of volume.
- AMOUNT: This sets how strongly the envelope affects pitch. Positive values bend the sound up; negative values bend the sound down.

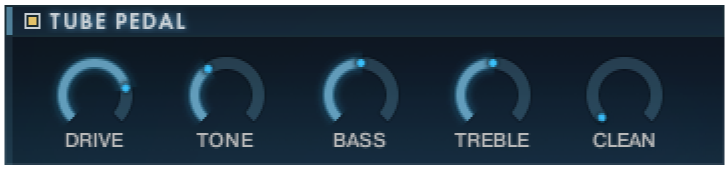
Filter Controls:

- TYPE MENU: choose filter mode (e.g., low-pass, high-pass, band-pass).
- CUTOFF: sets the frequency where the filter begins to take effect.
- RESONANCE: emphasizes frequencies around the cutoff point, adding sharpness or “bite.”
- VELOCITY: controls how note velocity affects cutoff. Higher velocities can open the filter more for expressive playing.
- AHDSR: envelope controlling how the filter cutoff evolves over time.
- AMOUNT: sets how much the AHDSR modulates the cutoff.

FX : DISTORTION

Tube Pedal

Inspired by the classic Ibanez TS-808 Tube Screamer, the Tube Pedal delivers a smooth, warm overdrive tone ideal for both lead and rhythm distortion sounds.



Controls:

- DRIVE: Controls the intensity of the overdrive effect.
- TONE: Shapes the overall brightness. Turning the knob to the right enhances treble frequencies, adding bite and clarity—great for sharp leads and cutting rhythms. Turning it left produces a softer, darker character.
- BASS: Boosts or cuts the low-end frequencies.
- TREBLE: Adjusts the presence of high frequencies.
- CLEAN: Blends the original, unaffected signal with the distorted tone. At 0%, only the distorted sound is heard; at 100%, it mixes in an equal amount of clean signal.

Lo-Fi

Lo-Fi introduces digital imperfections—such as quantization noise and aliasing—to otherwise clean audio signals. It's ideal for adding character and texture to sounds that might feel too sterile or flat.

Controls:

- BIT DEPTH: Reduces the signal's bit depth, with fractional settings (e.g., 12.4 bits) available to increase sonic grit. For reference, audio CDs use 16-bit resolution, vintage samplers often worked with 8 or 12 bits, and a 4-bit setting can recall the harsh tones of old electronic toys.
- SAMPLE RATE: Downsamples the audio to a user-defined rate without applying low-pass filtering, introducing intentional aliasing. The sample rate can be lowered to as little as 50 Hz, dramatically degrading the source audio.
- NOISE: Adds a layer of white noise or hiss to the signal.
- COLOR: Shapes the noise frequency spectrum, acting similarly to a low-pass filter.

Classic Amp

The Classic Amp emulates the iconic sound of Marshall JMP guitar amps, making it a great choice for achieving smooth, expressive lead tones.

Controls:

- HIGH GAIN: Engages a higher gain stage in the preamp, ideal for producing rich saturation and heavy distortion.
- MONO: Sums the input signal to mono before processing. When disengaged, each channel is processed separately, which may result in higher CPU usage.
- GAIN: Controls the amount of gain introduced by the preamp section. Increasing this adds warmth, grit, and harmonic content.
- BASS: Shapes the low-end response of the amp.
- MID: Adjusts the amp's response in the midrange frequencies.
- TREBLE: Controls the high-frequency content of the tone.
- PRESENCE: Enhances clarity and definition in the upper mids.
- MASTER: Sets the overall output level of the amplifier.

Modern Amp

Modeled after the Peavey 5150 amp, a staple in the world of high-gain amplification, it's aggressive, punchy tone delivers a full spectrum of gritty, in-your-face guitar sounds.

Controls:

- LEAD/RHYTHM: Switches between Lead and Rhythm channels.
- MONO: When enabled, combines all input channels into a single mono signal before processing. Disabling this allows separate processing of each channel, which may increase CPU usage.
- HIGH GAIN: Switches between standard gain and a more aggressive high-gain mode.
- BRIGHT: Adds a high-frequency boost to the Rhythm channel when enabled.
- CRUNCH: Introduces heavy distortion to the Rhythm channel for a more saturated tone.
- RHYTHM: Adjusts preamp drive for the Rhythm channel.
- LEAD: Adjusts the preamp drive level for the Lead channel.
- BASS: Modifies the amp's low-end response.
- MID: Sets the midrange tonal characteristics.
- TREBLE: Controls the presence of high frequencies.
- RESONANCE: Alters the low-end resonance in the power amp section.
- PRESENCE: Enhances the upper midrange frequencies for added clarity and bite.
- GAIN: Regulates the overall output volume and contributes to power amp saturation.

FX : MODULATION

Phaser

The Phaser enhances the audio signal by introducing dynamic tonal movement and intricate spectral shaping. It operates using a series of all-pass filters whose center frequencies are modulated over time. These filters create a series of peaks and dips across the frequency range, and as the modulation shifts these positions, the harmonic content of the sound is reshaped in a fluid, animated manner. The result can span from vintage rock textures to experimental, swirling effects. Phasers have long been a staple in recording studio hardware and guitar pedalboards.

Phaser includes a flexible bank of all-pass filters, capable of generating up to twelve matched peak/dip pairs. The built-in modulation system influences the filters' central position (Center) and their spacing (Spread), making it possible to create vocal-like resonances. An ULTRA mode pushes the modulation and filter ranges further, enabling audio-rate modulation and introducing harmonics akin to frequency modulation (FM) synthesis.

Controls:

- ULTRA: Expands the modulation and frequency range of the Speed and Center controls. At high Speed values (into the audio range), this can generate complex harmonics similar to FM tones.
- INVERT PHASE: Reverses the modulation phase for Spread, altering how it moves relative to Center.
- INVERT MOD MIX: Inverts the modulation signal itself, effectively flipping the positioning of the spectral features.
- SPEED: Sets the speed at which the modulation affects the *Center* and *Spread* parameters. Increasing the *Depth* intensifies the modulation's effect.
- DEPTH: Controls the depth of modulation applied to Center and Spread. The balance between the two is determined by the Mod Mix control.
- NOTCHES: Determines how many peak/dip pairs are used in the effect.
- CENTER: Moves the all-pass filter frequencies, effectively shifting all spectral peaks and notches together to new frequency ranges.
- SPREAD: Controls the distance between spectral peaks and notches. Lower values pack them closer together; higher values spread them farther apart, enabling formant-like effects.
- MOD MIX: Balances the modulation amount between Center and Spread. Left bias emphasizes Center, right bias emphasizes Spread. Center position applies equal modulation to both.
- STEREO: Introduces a phase offset between left and right channels in the modulation signal, enhancing stereo movement. The effect direction depends on the knob position. Has no impact when Amount is set to 0.
- FEEDBACK: Adds resonance to the effect by feeding the signal back through the all-pass filters. Higher settings increase the prominence of the frequency notches.
- MIX: Controls the blend between the unprocessed (dry) signal and the effect (wet) signal. Full left outputs only the dry signal.

Flanger

A flanger is an audio effect used to enhance and transform sounds by adding rich, modulated textures. Built on the principle of comb filtering, it employs a short delay with feedback that generates peaks and notches in the frequency spectrum. This interaction introduces resonant filtering effects that can range from shimmering metallic tones to the sweeping roar reminiscent of a jet engine spooling up.

Widely adopted in both studio gear and guitar pedals, flanging has long been a staple in audio production. This particular Flanger unit reimagines the traditional effect with an expanded feature set, allowing for more creative and extreme sonic shaping while maintaining intuitive control and familiar usability.

It offers three distinct flanger modes, each with its own approach to modulation and harmonic treatment:

- Standard: Delivers the classic flanging sound by producing frequency-dependent phase cancellations and reinforcements using harmonically spaced comb filters.

- Thru Zero: Emulates the vintage tape-flanging technique where delayed and non-delayed signals pass through zero phase difference, causing dramatic cancellation effects. Each flanger voice is paired with a static version, and modulation causes them to shift against each other. The Offset control allows fine-tuning of their alignment in the frequency domain, influencing the rhythmic feel and intensity of cancellation.

- Scan: Instead of layering multiple voices at once, this mode sequentially transitions through them, much like an arpeggiator cycles through notes in a chord. A Scan Mode selector provides options for triangle, upward saw, and downward saw modulation waveforms.

Controls (some are only available in certain modes):

- MODE MENU: Switches between three flanger modes (Standard, Thru Zero, Scan)
- CHORD MENU: Selects a chord structure that determines the pitch relationship between the flanger voices, creating harmonically rich textures.
- INVERT PHASE: Flips the phase of the effect signal, effectively reversing the filter's peaks and dips. This results in a perceived pitch shift—typically one octave lower—and in Thru Zero mode, can intensify rhythmic phase cancellations.
- SPEED: Sets how fast the modulation moves. When sync is enabled, modulation aligns with your DAW's tempo using a combination of note values and timing subdivisions.
- DEPTH: Determines how much modulation is applied to the pitch of the flanger voices, affecting the depth and intensity of the effect.
- OFFSET: When Thru Zero mode is active, this allows you to shift the duplicates of the flanged voices in the frequency range.
- PITCH: Shifts the base pitch of the primary flanger voice in semitone steps, moving all harmonic peaks and troughs up or down accordingly.
- VOICES: Adds up to four harmonically tuned flanger voices. In Standard and Thru Zero modes, additional voices are layered; in Scan mode, voices are faded between one at a time in a progressive sequence.
- DETUNE: Fine-tunes each flanger voice individually (within ±60 cents), simulating oscillator drift and creating a richer ensemble effect, especially effective when Chord is set to Unison.
- WIDTH: Expands the stereo image by panning modulated voices across channels and introducing phase offsets and cross-feedback for a more animated spatial effect.
- DAMPING: Reduces high-frequency content in the feedback loop, allowing for smoother, darker flanging tones even at high feedback levels.
- FEEDBACK: Controls how much of the processed signal is fed back into the flanger, enhancing resonance and metallic characteristics.
- MIX: Balances the dry input with the processed signal. Turning fully left yields only the original sound; fully right outputs only the flanged signal.

Chorus

The chorus effect enriches audio by introducing subtle timing and pitch variations, giving the impression of multiple sources playing simultaneously. This creates a sense of spatial depth and movement, often resulting in a warm, ensemble-like texture. The effect relies on short delay lines that are modulated over time, which introduces small pitch fluctuations due to shifting delay times. Depending on the settings, this can range from mild tonal changes to immersive, vibrant stereo layers.

A staple in both guitar rigs and studio setups, chorus has been implemented in numerous formats—rack units, pedals, and synthesizers. This particular chorus module draws inspiration from the iconic synth and studio gear of the late 70s and early 80s but offers modern, hands-on parameter control for greater flexibility.

This chorus includes four distinct modes that emulate classic hardware tones and behaviors. It uses up to three pairs of delay lines, referred to as Voices, to generate its effect. These voices maintain the stereo character of the original input but can be independently panned using the Width control for a broader image. Modulation is applied uniquely to each voice, reducing the chance of repetitive or predictable motion.

A unique Scatter mode extends the typical chorus architecture to produce reverb-like effects without the metallic resonance often caused by high feedback levels.

Controls:

- MODE MENU: Selects one of four chorus types:
    - Synth: Modeled after vintage analog synth choruses. Characterized by a dark, classic tone with spacious modulation.
    - Ensemble: Inspired by string machine choruses of the '70s, delivering a lush, animated sound.
    - Dimension: Echoes the transparent, wide choruses found in '80s rack gear. Clean and consistent.
    - Universal: A flexible, modern interpretation suitable for a wide variety of sounds depending on the number of voices.
- SCATTER: Activates a special feedback structure that mimics reverb behavior, softening high-feedback artifacts.
- INVERT PHASE: Flips the phase of the effect signal, altering the tonal characteristics of the chorus interaction.
- SPEED: Sets the modulation speed, from slow, gentle pitch drifting to rapid, vibrato-like motion. The intensity of this effect increases with the Amount setting.
- DEPTH: Controls how much modulation is applied to the delay times. Because delay modulation affects pitch, this parameter directly shapes the chorus intensity.
- DELAY: Adjusts the base delay time for each voice, influencing the perceived depth and interaction with feedback.
- VOICES: Crossfades between one and three chorus voices. More voices create a richer and more complex sonic character. The second and third voices are modulated differently for a more animated stereo image.
- WIDTH: Controls the stereo spread of the voices. At 0%, the original stereo image is maintained, increasing the value pans voices outward.
- FEEDBACK: Routes a portion of the chorus output back into the input path, enhancing sustain and spaciousness.
- MIX: Crossfades between the dry (unaffected) input and the processed signal. At full left, only the dry sound is heard.

FX : TIME / SPACE

Time

This delay effect module provides five versatile modes—Modern, Analog, Tape, Vintage, and Diffusion—each designed to add its own distinct sonic character. All modes include core controls for delay time, feedback, low and high cut filters, and a ping pong stereo effect. Additionally, each mode offers a unique set of parameters to help you shape the sound to your creative needs.

Modes:

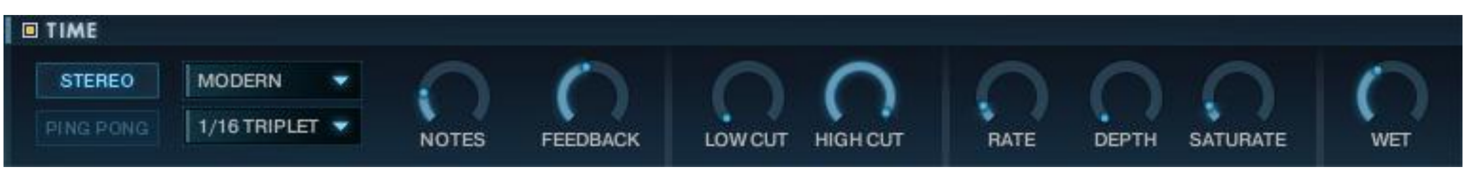
- Modern delivers a crisp, uncolored delay with optional saturation and filtering, ideal for retaining clarity while adding subtle warmth. It's especially well-suited for instruments like acoustic guitar or piano and maintains a consistent pitch when adjusting delay time.

- Analog recreates the tone of classic Bucket Brigade Device (BBD) delays, offering four different circuit emulations. Each model introduces varying levels of warmth, grit, and smoothness reminiscent of vintage hardware.

- Tape emulates classic tape delay systems, simulating the physical quirks of analog tape. You can control parameters such as Tape Age, Flutter, and Saturation to introduce hiss, wow, and analog coloration.

- Vintage simulates early digital delay units. It includes four quality modes that reduce sampling resolution as delay time increases, mimicking the artifacts of memory-limited digital hardware. Each quality setting varies in sample rate and interpolation characteristics.

- Diffusion can function as a delay, but excels at creating lush, reverb-like textures. Parameters like Density, Modulation, and Size allow for expansive ambient sounds perfect for cinematic or atmospheric music.



Controls (some are only available in certain modes):

- STEREO: Introduces a time offset between left and right modulation for a widened stereo image. When off, both channels modulate identically.
- PING PONG: When enabled, delay echoes alternate between the left and right channels for a bouncing stereo effect.
- NOISE: Adds tape hiss. The level of hiss is affected by the Tape Age setting.
- DENSE: Toggles between two reflection density settings, changing the perceived complexity of the effect's space.
- MODE MENU: Choose between the five available delay types: Modern, Analog, Tape, Vintage, or Diffusion.
- NOTES: Sets the delay length in note values. This delay unit is always synced to the host's tempo.
- FEEDBACK: Controls how much of the delayed signal is fed back into the input. Higher settings increase the number of repeats, with values over 100% capable of creating self-oscillation.
- LOW CUT: Shape the tone of the feedback loop by filtering out bass frequencies. Turning the knobs fully disengages their respective filters.
- HIGH CUT: Shape the tone of the feedback loop by filtering out treble frequencies. Turning the knobs fully disengages their respective filters.
- RATE: Sets the speed of delay modulation.
- DEPTH: Adjusts how much modulation affects the delay time.
- SATURATE: Adds analog-style saturation at the input stage. Increase to move from subtle warmth to full overdrive; fully counterclockwise disables it.
- BBD TYPE type: Choose from four Bucket Brigade Device models (Grunge, Dark, Warm, Clean), each offering varying degrees of filtering and saturation.
- FLUTTER: Adds pitch instability by simulating tape speed fluctuations.
- TAPE AGE: Emulates the tonal degradation of aging tape, including high-end roll-off and increased hiss when NOISE is active.
- QUALITY: Selects the digital fidelity of the delay: Crunch (0–24%), Low (25–49%), Medium (50–74%), or High (75–100%).
- TAPE AGE: Emulates the tonal degradation of aging tape, including high-end roll-off and increased hiss when Noise is active.
- SIZE: Sets the perceived space by controlling the reflection buildup and decay of the reverb tail.
- AMOUNT: Controls the intensity of diffusion applied, blurring delay taps into a more reverb-like sound. Extreme values can impact sync precision.
- MOVEMENT: Adjusts modulation depth and speed for the diffusion algorithm, affecting pitch and timing of reflections.
- RETURN: Controls the amount of effect applied to the signal.

Plate

Replicates the characteristics of a classic plate reverb. Originally developed as one of the first artificial reverb techniques, plate reverb uses a suspended sheet of metal to recreate the way sound reflects in a room. Because the vibrations travel across a flat surface rather than through a 3D space, the resulting reflections are denser and more consistent. Plate reverbs are known for their smooth, bright tails, making them ideal for enhancing vocals and snare drums without pushing them back in the mix. The Damping parameter shapes the tonal quality—on brighter sources, increasing damping can create a warmer, more controlled reverb.

Controls:

- PRE-DELAY: Adds a brief delay before the reverb begins, helping to separate the dry signal from the effect.
- DECAY: Determines how long the reverb lingers.
- LOW SHELF: Boosts or cuts low frequencies in the reverb signal.
- HIGH DAMP: Controls how quickly high frequencies decay, softening the brightness over time.
- HIGH DAMP: Controls how quickly high frequencies decay, softening the brightness over time.
- STEREO: Adjusts the width of the stereo field; higher settings produce a broader image.
- RETURN: Controls the amount of effect applied to the signal.

Room / Hall

This reverb includes two distinct modes: Room and Hall. The Room setting produces a realistic, tight reverb with pronounced early reflections and a quick decay—ideal for simulating the acoustics of smaller spaces. Its fast response time makes it particularly effective for enhancing drums and guitars.

The Hall mode, in contrast, recreates the expansive ambiance of a large concert hall, offering a lush and immersive reverb tail. It's well-suited for tonal instruments and broader soundscapes, thanks to its spacious character and extended decay. Both modes support modulation of the Room Size and Pre Delay parameters, allowing for expressive spatial effects.

Controls:

- MODE: Toggle between the Room and Hall reverb types.
- TIME: Controls how long the reverb tail persists.
- PRE-DELAY: Adds a short delay before the reverb begins, creating separation between the dry and wet signal.
- SIZE: Alters the perceived size of the virtual space. Higher values simulate larger environments.
- DAMPING: Simulates how much sound is absorbed by the environment. Higher damping reduces high-frequency reflections.
- DIFFUSION: Adjusts how densely the reflections are distributed within the reverb tail.
- MOD: Introduces movement or pitch variation into the reverb. Turning it fully left disables modulation.
- STEREO: Sets the stereo width of the reverb. Higher values increase the perceived spatial spread.
- LOW SHELF: Boosts or reduces the low-end frequencies of the reverb signal.
- HIGH CUT: Rolls off high-frequency content for a warmer reverb sound.
- RETURN: Controls the amount of effect applied to the signal.

Convolution

Convolution is an advanced mathematical technique used to accurately replicate the acoustic characteristics of a linear system — such as a physical space, a loudspeaker, or an outboard reverb unit — and apply them to your own audio. This is done by capturing a brief recording of a broadband signal as it passes through the system, resulting in what's known as an impulse response (IR). This file, which is typically a standard audio format, is then processed to recreate the original environment's sonic fingerprint.

While convolution is most commonly associated with ultra-realistic reverb effects, it's equally effective for emulating the tonal coloration of speaker cabinets and similar systems.

Controls:

- PRESET BROWSER: This window lets you choose from a broad selection of over 700 carefully categorized impulse responses.
- REVERSE: Plays the impulse response backward for creative or non-traditional effects. Long reverbs may not produce any sound for a bit of time when reversed.
- PRE-DELAY: Adds a slight delay between the dry signal and the processed output. This is especially useful for reverb IRs, mimicking the natural delay before early reflections arrive in large spaces.
- SIZE: Time-stretches or compresses the impulse response, altering its perceived spatial scale.
- LOW PASS: Filters out high-frequency content.
- HIGH PASS: Filters out low-frequency content.
- RETURN: Sets the return level of the processed signal.

PULSE : MODS

Gain, Pan, and Pitch LFOs

These 3 Low Frequency Oscillators can modify the gain, pan, and pitch of the signal.

Controls:

- SYNC: Syncs RATE to hosts tempo. When selected, RATE is in note values, otherwise its value is in Hz.
- PRESETS: A menu of LFO presets.
- AMOUNT: Controls how much LFO affects signal.
- RATE: Adjust the speed of the LFO.
- PULSE WIDTH: Changes pulse waveform duty cycle (only applicable when the pulse wave ins in use).
- WAVEFORMS: Each of the individual waveforms can be added to the LFO to modify its shape.

Step Sequencer

A sequencer that rhythmically modulates gain prior to the effects chain.

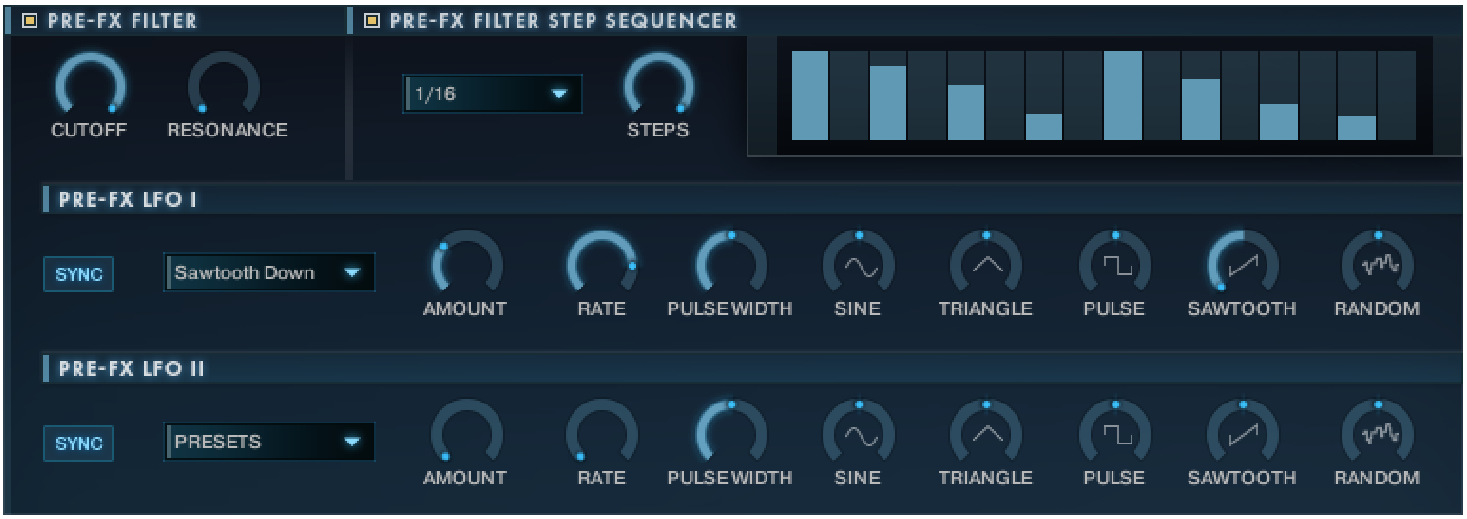
Controls:

- NOTE MENU: Set duration for each step, in note values.
- SHORTNESS: Set the resolution of step timing (e.g. how short each note is in each step).
- STEPS: Adjusts the total number of steps to play.
- NOTE DISPLAY: Click or drag to edit step values. 0 value = no step/note. You can click/drag across multiple steps to quickly set values.

PULSE : FILTERS

Pre-FX Filter, Step Sequencer, and LFOs

This is a 2-pole filter that attenuates frequencies above the cutoff at a rate of -12 dB/octave. It affects the signal before the effects chain and before the POST-FX FILTER and the OUTPUT section. The Step Sequencer modulates the cutoff frequency of the filter. Low values lower the cutoff frequency; high values raise it. In addition, the 2 LFOs can further modulate the cut-off frequency as well.



Filter Controls:

- CUTOFF: Sets the filter's cutoff frequency.
- RESONANCE: Applies a gain boost near the cutoff frequency.

Step Sequencer Controls:

- NOTE MENU: Set duration for each step, in note values.
- STEPS: Adjusts the total number of steps to play.
- NOTE DISPLAY: Click or drag to edit step values. 0 value = no step/note. You can click/drag across multiple steps to quickly set values.

LFO Controls:

- SYNC: Syncs RATE to hosts tempo. When selected, RATE is in note values, otherwise its value is in Hz.
- PRESETS: A menu of LFO presets.
- AMOUNT: Controls how much LFO affects signal.
- RATE: Adjust the speed of the LFO.
- PULSE WIDTH: Changes pulse waveform duty cycle (only applicable when the pulse wave ins in use).
- WAVEFORMS: Each of the individual waveforms can be added to the LFO to modify its shape.

Post-FX Filter and Step Sequencer

This is a 2-pole filter that attenuates frequencies above the cutoff at a rate of -12 dB/octave. It affects the signal after the effects chain (everything in the FX section as well as the PULSE : SOURCE section) and before the OUTPUT section. This Step Sequencer automates the movement of the cutoff frequency of the filter to give the filter a rhythmic pulse. Low values lower the cutoff frequency; high values raise it.

Filter Controls:

- CUTOFF: Sets the filter's cutoff frequency.
- RESONANCE: Applies a gain boost near the cutoff frequency.

Step Sequencer Controls:

- NOTE MENU: Set duration for each step, in note values.
- STEPS: Adjusts the total number of steps to play.
- NOTE DISPLAY: Click or drag to edit step values. 0 value = no step/note. You can click/drag across multiple steps to quickly set values.

OUTPUT

Tape Saturation

Simulates the natural compression and harmonic distortion produced by analog tape machines. When applied subtly, it adds warmth and character to audio; when pushed harder, it delivers pronounced distortion for a more aggressive tone.

Controls:

- HQ: Enables internal oversampling, which improves sound fidelity at the cost of higher CPU usage.
- GAIN: Adjusts the signal level going into the effect, directly impacting the intensity of saturation and compression.
- WARMTH: Boosts or reduces low-end frequencies to shape the tonal warmth of the signal.
- HF ROLLOFF: Sets the cutoff point for high-frequency attenuation, softening the top end beyond this threshold.

Stereo Modeler

Gives you precise control over your signal's stereo image. You can adjust its stereo width, reposition it within the stereo field, or simulate stereo from a mono input.

Controls:

- PSEUDO: Enables a stereo simulation algorithm for mono sources. NOTE: This should only be used with mono signals, as it can produce sounds that may not translate well in mono playback and could vanish from a mix when summed to mono.
- SPREAD: Adjusts the width of the stereo image. Turning it fully left collapses the signal to mono; while turning it right increases the stereo width. When widened artificially, the signal may seem to extend beyond the speakers but be cautious—this can lead to mono compatibility issues.
- PAN: Moves the signal left or right within the stereo field, functioning identically to the Pan control on the Source page.

Transient Designer

A dynamics processor specifically designed to shape the attack and sustain characteristics of audio signals. Unlike conventional compressors that respond directly to amplitude, this tool reacts to the overall envelope of the signal, making it less sensitive to fluctuations in input volume. It is particularly effective on sounds with pronounced transients, such as drums, pianos, and plucked instruments like guitars. Due to its powerful processing, it can produce dramatic results—so it's best to use it judiciously.

Controls:

- SMOOTH: If you're working with non-percussive sources (e.g., acoustic guitar) and the processing seems too aggressive or uneven, activating this option can help produce a more natural result.
- ATTACK: Modifies how the initial transients are emphasized or subdued. Raising this value enhances punch; while lowering it softens sharp onsets.
- SUSTAIN: Alters the tail portion of the sound. Increasing sustain can add fullness and presence; reducing it shortens the decay.

Compressor

A compressor is a type of dynamic processor that automatically reduces the volume of louder portions of an audio signal, effectively narrowing its dynamic range. This compressor is inspired by the iconic SSL G-Series Bus Compressor.

Controls:

- STEREO LINK: When enabled, the compressor applies identical processing to both the left and right channels, helping to maintain a consistent stereo image. When disabled, each channel is processed separately, allowing for independent dynamics control (dual mono operation).
- THRESHOLD: Determines the level above which compression begins. Only signals that exceed this level will be affected; anything below remains unchanged.
- RATIO: Sets the degree of compression applied to signals that exceed the threshold. For example, a ratio of 4:1 means that for every 4 dB over the threshold, the output level increases by just 1 dB. A 1:1 ratio results in no compression.
- ATTACK: Controls how quickly the compressor engages after the input signal crosses the threshold. Faster attack times result in more immediate compression.
- RELEASE: Determines how long it takes for the compressor to stop compressing once the input level falls back below the threshold.
- MAKEUP: Adjusts the gain of the output signal to compensate for any reduction in volume caused by the compression process.
- MIX: Balances the processed (wet) signal with the unprocessed (dry) signal. This allows for parallel compression techniques—where quieter elements are enhanced without overly squashing louder ones. A 100% mix outputs only the compressed signal, while 0% outputs only the dry signal.

Equalizer

This equalizer is inspired by the famous analog design of the SSL G-series console's channel strip. It features a versatile 4-band parametric EQ, with both bell and shelving options available for the high and low frequency bands.

Controls:

- GAIN: Boosts or attenuates the frequency selected by the FREQ control.
- FREQ: Sets the specific frequency where GAIN adjustments are applied.
- Q: Adjusts the bandwidth of the frequency band. Unlike typical EQs, a higher Q widens the band—mirroring the behavior of the original hardware.
- SHELF / BELL SWITCH: Switches the high of low bands between notch and shelf modes.
    - NOTCH will center boost/cut around chosen frequency.
    - SHELF will boost/cut everything below (for LOW) or above (for HIGH) the chosen freq.

OPTIONS

Aftertouch Vibrato

This option allows you to add natural pitch modulation by pressing harder on your MIDI keyboard keys (if your keyboard supports monophonic aftertouch). If you have a keyboard that supports *polyphonic* aftertouch, like the MK3 series of Native Instruments Komplete Kontrol keyboards, you will need to change it to send *monophonic* aftertouch signals instead of *polyphonic* to use the effect in this library.

Controls

- DEPTH: Sets the depth of the modulation amount, from 1 to 12 semitones.
- RATE: Sets the modulation speed, synchronized to note values.

Pitch Bend Range

This option allows you choose the range in semitones for pitch bend.

Controls

- DEPTH: Sets the depth of the pitch bend, from none to 12 semitones.

“BYOS” EDITION

The “BYOS” (Bring Your Own Samples) edition, found in the *Extras* folder, has drag and drop sample support (WAV, AIF, AIFF, and NCW) for the 3 voice layers. Also included in the *Extras/Waveforms* folder is a selection of 122 simple waveforms for some basic subtractive synthesis.

Use “File > Save as” to save your creations as NKI instrument files (snapshots will not work)!

Each “BYOS” voice layer features:

- MIR (music information retrieval / auto root note detection) or manual root note selection
- One-shot or looped file support
- Manual selection of voice key range (low key, high key)

Controls

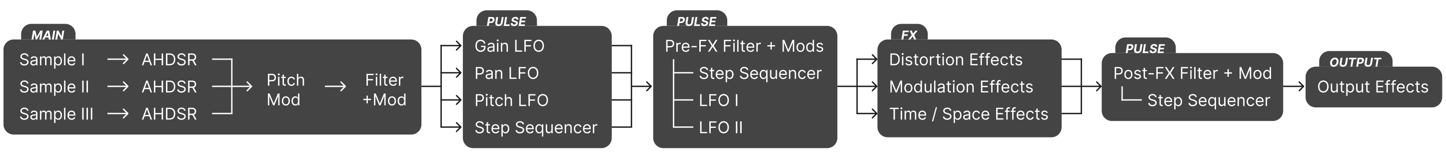
1. MODE MENU: Choose either One Shot or Sustain to loop the sample file.
2. ROOT MENU: Choose to use the MIR system to identify the root note of the file or Manual to reveal the ROOT KEY control and manually set it.
3. LOW KEY: Sets the lowest note available for the layers sample
4. HIGH KEY: Sets the highest note available for the layers sample
5. MIR RESULT: Displays the resulting root note of the MIR analysis
6. ROOT KEY: This control allows you to manually choose the root note
7. WAVEFORM DROP AREA & DISPLAY: Drag and drop your sample in this area to load it into the layer. Once loaded, a representative view of the sample will show in this area.

TIPS

- Many controls in the Convolution effect do not reflect real-time value changes due to a known issue in Kontakt. While the knob's visual position is accurate, the displayed value may not update immediately. To view the correct value after adjusting a Convolution control, click it again.
- CRTL/click on PC (CMD/click on Mac) on any control to reset it to its default value and position.
- For delicate or precise control over any knob or slider, utilize the MIDI CC controller mapping feature. Right click any control to map it to a knob or slider on your physical controller.
- This is a high-quality library that includes some CPU-intensive effects. If you encounter performance issues, the following tips can help improve efficiency:
  - Refer to Native Instruments' Official Optimization Guide: https://support.native-instruments.com/hc/en-us/articles/210275605-How-Can-I-Optimize-the-Performance-of-Kontakt
    - Suggested optimizations include:
      - Latency Settings: Adjust your audio interface buffer size to balance responsiveness and stability.
      - Multiprocessor Support: Enable this in the Kontakt settings to distribute processing across CPU cores.
      - Voice Limit: Reduce the maximum number of simultaneous voices to lighten the processing load.
  - Modify the Preload Buffer Size:
    - Navigate to the "Memory" tab in the Kontakt Options menu. Reducing the preload buffer size (commonly 30–60 KB for SSDs) can conserve RAM but increases dependency on disk streaming. HDDs may need a higher buffer (e.g., 60 KB or more).
  - Perform a Batch Resave:
    - This process can resolve file path issues and significantly speed up loading times. Right-click the library folder and select "Batch Resave." Follow the prompts carefully.
    - Note: Libraries batch-resaved in later versions of Kontakt will no longer work in earlier versions. To retain backward compatibility, perform the resave in the specific Kontakt version you intend to use.
  - Purge Unused Samples:
    - Use the "Purge" feature to remove unused samples from memory. You can either purge manually via the "Purge" button or choose "Purge All Samples" from the instrument's menu. This ensures only the samples required by your current MIDI data are loaded—ideally used once your track is finalized.
  - Use a Dedicated SSD:
    - For best performance, store your Kontakt libraries on a separate SSD from your operating system or DAW projects. Avoid running projects and libraries from the same drive, as this can slow down both playback and loading times.

- If you are not hearing any difference when using the Pre-FX of Post-FX Filter Step Sequencers, make sure you have the Pre-FX or Post-FX Filter Cutoff engaged. Meaning, make sure the Cutoff is not fully turned to the right. The value of the Cutoff sets the lower values of the steps in the sequencer.
- If you are an experienced Kontakt user, you can experiment with changing the mode from "DFD" to "Sampler" to see if that helps with performance with your particular computer configuration. There is a known bug within Kontakt 6 that can sometimes happen where a sample will not play fully. Native Instruments to date has not patched this bug, but if you experience it, switching from "DFD" to "Sampler" will fix it/
- Suggested Continuous Controller Mappings:
    - CC 1 (Mod Wheel) : Source Filter Cutoff
    - CC 112 : Source Filter On/Off

SYNTH FLOW CHART



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"If I have seen further, it is by standing on the shoulders of Giants."
— Sir Isaac Newton, 1675

Special thanks to Joanna Pena for their excellent Kontakt template "Photosynthesis". It served as the starting point and initial inspiration for this library. Check it out if you are interested in creating your own Kontakt instruments, as well as their libraries based on the Photosynthesis engine.

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