NEIL PEART DRUMS
VOLUME 1: THE KIT

USER MANUAL

DRUM MASTERS 2 Studio-ProFiles

infinite Player

SONIC REALITY

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NEIL PEART DRUMS
VOLUME 1: THE KIT

Thank you for purchasing Sonic Reailty's Neil Peart Drums Vol. 1: The Kit, an Infinite Player library for Kontakt! In this manual you will find detailed instructions on the installation, configuration and use of your product.

Neil Peart, legendary drummer of the band Rush, has collaborated with Sonic Reality and producer/engineer Nick Raskulinecz (Rush, Foo Fighters, Alice In Chains) to bring the authentic sound of Neil's "Snakes and Arrows" Custom DW® Drum Kit into the digital domain. With advanced features such as deep level dynamics, humanized random alternating hits, discrete multiple mic mixing and more, this deluxe sampled kit is an ultra-realistic recreation of Neil's actual acoustic set as used live on tour and in the studio.

For the first time, e-drummers, keyboardists and composers can play rock drum samples with the iconic signature sound of drum legend Neil Peart, and have a world-class, hard-hitting drum kit suitable for many styles of music from Progressive Rock to Heavy Metal! Producer Nick Raskulinecz engineered the sample sessions with over 20 channels of Neve® mic preamps, deluxe vintage tube mics and multiple room positions for one of the most elaborate sampled drum kits ever.

Drum Masters 2

Studio ProFiles

About Drum Masters 2

Drum Masters 2 is a virtual drum instrument packed with the tools to create the most realistic virtual studio drum sessions.

Drum Masters 2 is part of Sonic Reality's line of Infinite Player libraries for Native Instruments' Kontakt sampler. It uniquely allows Sonic Reality's extensive range of instruments, loops and sound effects samples to be purchased in a modular fashion to create a custom library specific to each user's needs. To take advantage of this cutting edge system, it important to first understand the individual components involved:

1. Kontakt Player 4
2. Infinite Player
3. Infinite Library Installer
4. Sample and patch zip files
5. Studio ProFiles Infinite Player_info.nkx

And the following file types:

.nki Instruments, .nkm Multis, .nkx Samples and Graphics, .nkc Temporary Cache Files

1. Kontakt Player 4

This is the sample playback engine that Sonic Reality uses for its Infinite Player libraries. It is made by Native Instruments and is one of the most popular sample platforms available for Mac and PC. It allows multi-sampled instruments, loops, and sound effects to be played in RAM or streamed from your hard drive for use either as a standalone application or as an RTAS, VST or AudioUnit plug-in for all major DAWs.

The Kontakt Player component of this bundle is available free from Native Instruments and is also provided here for convenience by Sonic Reality. You may also use the full version of Kontakt (optional) with Infinite Player products. In this case, the Kontakt Player is not required. **Please note that in order for Kontakt Player or Kontakt to play any Infinite Player-based sounds from Sonic Reality, the Infinite Player must be authorized with a serial number and activated using the Native Instruments Service Center application**. Please read the installation instructions for more information.

2. Infinite Player

Sonic Reality's Infinite Player is a uniquely expandable and customizable third party library module that works inside Kontakt. It is the gateway to being able to use all Infinite Player-based sound collections and virtual instruments by Sonic Reality. It is important that the Infinite Player Library is activated with Native Instruments using the supplied Infinite Player serial number to unlock the sounds. If not unlocked, the sounds will play in demo mode.

One of the great things about the Infinite Player is that once you activate the Infinite Player, you are a drag and drop away from adding more and more sound collections to your setup efficiently and easily. Knowing this, it is worth taking this moment to understand all of the components involved so that you can take full advantage of this powerful, modular, customizable sound station.

3. Infinite Library Installer

This is an exclusive sound installer from Sonic Reality that allows a simple drag and drop of patches and / or samples in special .zip files. Whether you obtain the sound .zip files from discs or via download, you must keep the .zip intact (do not unzip these files manually) and let the Infinite Library Installer take care of putting all of the patches and samples into the correct places within your Infinite Player folder. Note that all of your Infinite Player based sounds will reside in an Infinite Player folder that you can place on any hard drive. Click “Add Library” within Kontakt to choose your Infinite Player folder location.

4. Sample and patch .zip files

For installation purposes, samples and patches are provided as specially constructed .zip files. It is important that you do NOT unzip these files manually. They are to be used by simply dragging and dropping onto the Infinite Library Installer so that all types of Instruments, Multis, samples, and graphics are automatically placed in the proper folder hierarchy. **IMPORTANT!** Kontakt wants to see sounds in a specific location all within the correct folder hierarchy. We’ve made that part easy—just drop the .zip files onto the Infinite Library Installer, choose the Infinite Player folder as the destination for the sounds to be installed, and the Installer will take care of the rest!

5. Studio ProFiles Infinite Player_info.nkx

This particular file, which we refer to as the “info file,” is very important because it contains the graphic interface skins (.tga files) that characterize Infinite Player sounds. It also contains the Infinite Player Library ID that Kontakt requires. In order to be up to date with the latest graphics, you can check http://www.sonicreality.com for a free download of the latest version of this file.

DVD Products: Important Installation Instructions

If you are **NEW** to the Infinite Player, please follow the NEW user instructions. If you are an **EXISTING** Infinite Player user, follow the EXISTING user instructions on the next page.

Installation Instructions for NEW Infinite Player users

1. Locate and install Kontakt Player 4. Follow ALL onscreen instructions.

2. Locate the **Infinite Player** folder on disc and drag it to a location of your choice. NOTE: Due to default OS security settings, Windows Vista / 7 users must choose the Documents folder or a secondary drive (ie, external) for the library location.

3. Locate the **Infinite Library** Installer appropriate for your operating system (located in the Infinite Library Installers folder). Drag the Library Installer to your desktop and double-click to open it.

4. Drag each of the .zip files from the **Product Discs** onto the Infinite Library Installer. **DO NOT** unzip the files. When asked for the location of your Infinite Player folder, browse to the location chosen in Step 2.

5. Launch Kontakt Player 4. Once open click the "add Library" tab and navigate to your Infinite Player folder. Once you have located the Infinite Library folder, a message will pop up saying "Kontakt will now add the library to the Service Center, please provide admin rights if prompted". Hit the "Ok" button. Kontakt Player will be running in DEMO mode. Demo mode works for 30 minutes at a time.

6. You must activate the library in the Native Instruments Service Center application (installed in Step 1). Click "Activate" on the Infinite Player (as seen below) to launch the Native Instruments Service Center. Use the included serial number and follow the Service Center instructions to activate your Infinite Player.

Installation Instructions for EXISTING Infinite Player users

1. Locate the **Infinite Library** Installer appropriate for your operating system (located in the Infinite Library Installers folder). Drag the Library Installer to your desktop.

2. Double Click to launch the Library Installer

3. Locate and drag each of the .zip files from the **Product Discs** onto the Infinite Library Installer interface. You will then be asked to locate your **Infinite Player** folder; select the location where you have installed your previous Infinite Player products.

4. After the Library Installer completes installation of each .zip file(s), your sounds will be available and ready to use in your Infinite Player.

Getting Started

In this manual we will go into detail about using the Drum Masters 2 sound library and the features associated with it.

If you have not yet installed the Kontakt Player, please read the "Important Installation Instructions" section of the manual to learn the quickest way to get your Infinite Player up and running.

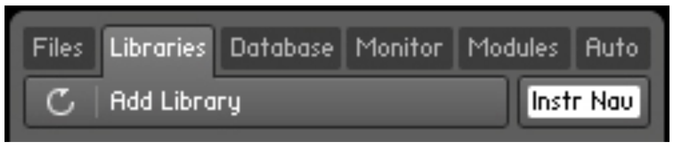
Separate Kontakt Player manual
Please note that this document is NOT the Kontakt Player manual. For detailed information on issues specific to Kontakt, please see the Kontakt User Manual for the current version you are using.

Missing samples?
If you load a Patch and see a "missing samples" error, it is possible that one of your sound library .zip files did not install entirely and that samples were left out of the installation. In this case, please re-drag all of the .zip files for that particular set of sounds onto the Infinite Player Library Installer interface to re-install.

Note for Downloadable Versions:
If you purchased the download version, compare the size of the downloaded .zip with the size showing in your download area to ensure your download is complete. If the download is incomplete, samples may be missing and you will need to re-download. A good download manager application with "resume" is recommended to avoid internet interruptions for downloads.

After Installing the Kontakt Player

After installing Kontakt Player you will need to link or add the Drum Masters 2 library to the Kontakt Player "Libraries" menu.



Click "Add Library" and select the "Infinite Player" folder from its location on your computer or HD.

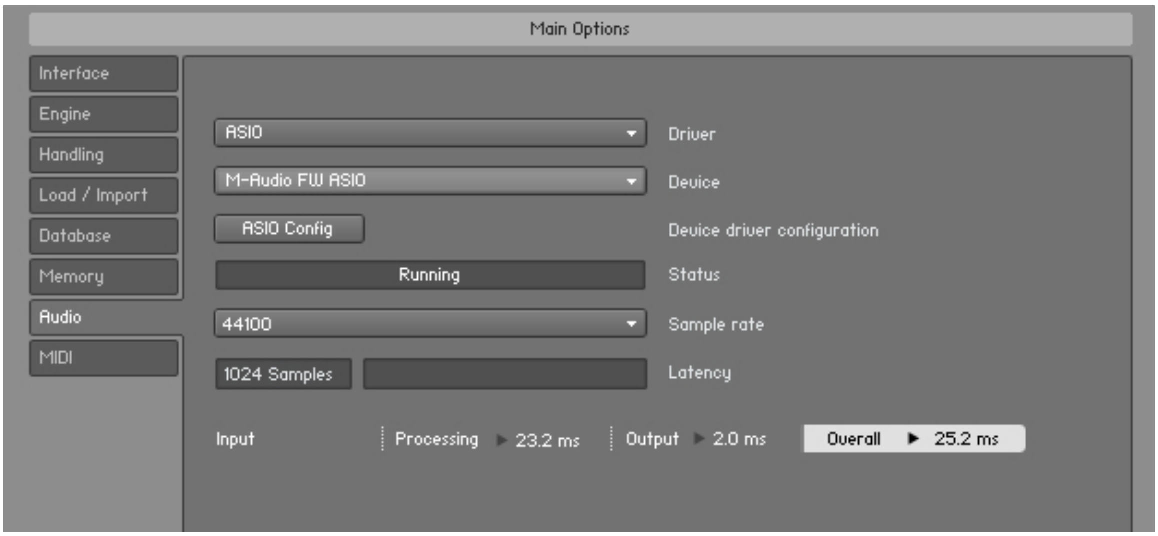
Note: If you have previously installed an Infinite Player library, the Infinite Player may already be visible in your library menu.

Activate your Infinite Player
If this is your first time using the Infinite Player, you must activate the library in the Native Instruments Service Center application. Click "Activate" on the Infinite Player (as seen below) to launch the Service Center application. Follow the Service Center instructions to activate your Infinite Player.

Note: Failing to activate the library in the Native Instruments "Service Center" will cause the product to run in Demo Mode which limits usage to 30 minutes.

Audio and MIDI Setup in Kontakt

All audio options can be found in the “Main Options” menu under the “Audio” tab. Here you can set your audio card and adjust the following settings:



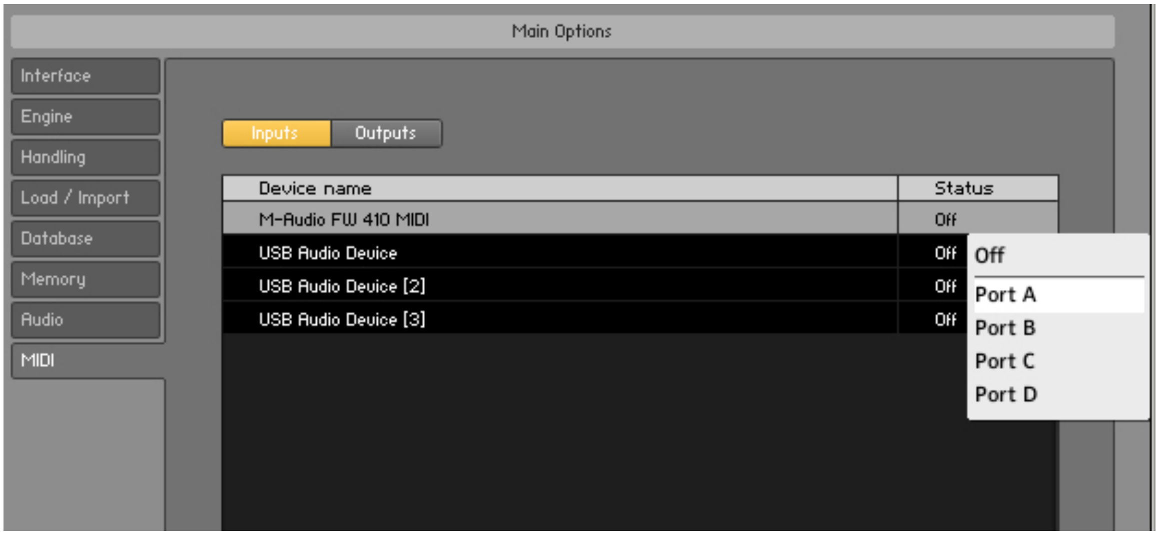
Driver: This drop-down menu selects which of your operating system’s device driver architectures KONTAKT should use. Most professional audio devices provide ASIO™, CoreAudio™ (Mac), or WASAPI™ (Windows) drivers.

Device: This menu lists all connected audio interfaces that match the driver architecture chosen above. Select the audio interface that you would like to use for playback here.

Sample rate: This drop-down menu allows you to set the global playback sample rate at which KONTAKT will operate. Common values are 44100 Hz for music and 48000 Hz for film production. Note that this doesn’t correspond with the sampling rate at which your samples have been recorded – if the playback rate doesn’t match a sample’s recording rate, KONTAKT will handle all necessary conversion steps transparently for you.

Latency: The size of the audio playback buffer in samples. Small values will shorten the delay between pressing a key and hearing the resulting sound (this is called “latency”), but may cause drop-outs and stuttering when playing many voices at the same time. Setting this to a higher value will make playback more reliable, but this will create a longer delay between the key and hearing the sound. For more info on latency please refer to the Kontakt Player manual.

MIDI: To select your MIDI controller in Kontakt Player, click the “Options” tab then the MIDI tab. Your MIDI controller should show up here and allow you to assign it to a MIDI port.



Once selected, your controller will trigger your loaded patch.

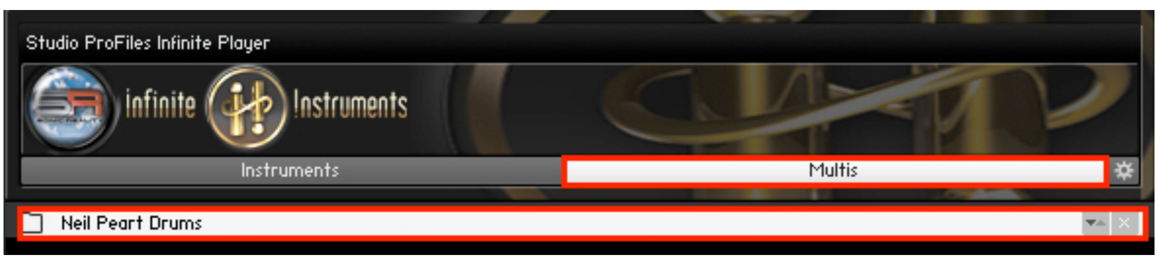
Note: If your controller is not visible under the “MIDI” tab you may need to re-install your controller's MIDI Drivers. Please see the Kontakt Player manual for more information regarding MIDI.

Loading Drum Kits

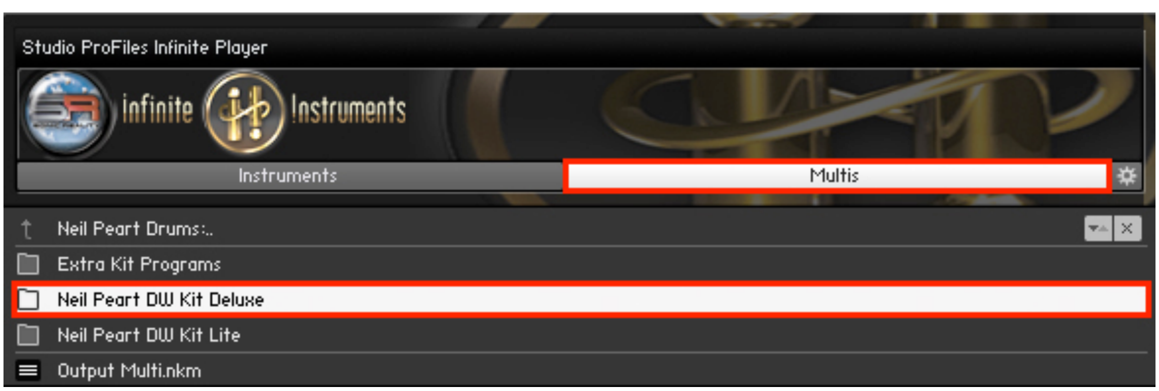
Sonic Reality's multitrack playable drum kits have discrete control of the mic channels just like you'd find in today's popular drum plug-ins. They offer many of the same features, but they also differ in that they use very particular styles of instruments and recording techniques, and they often include the artist's personal equipment that is either rare, vintage, custom, or top of the line hand-selected instruments from the factory. Besides the unique character of these drum kits, all Drum Masters 2 products include several different instrument mappings: Sonic Reality's unique IMAP format, GM, V-Drum, Performance, and Custom mappings. (See page 21 for mapping formats.)

Drum Masters 2 Drum kits consist of **Multis** and **Instruments** (These are Kontakt's terms for the two types of patches, one that is a combination of Instruments and the other that is just the single Instruments). A Drum Masters 2 "Multi" is a set of kit pieces (Instruments) that make up the particular kit. To load a kit, simply select one of the multis from the browser, and then drag it to the right into the Kontakt Player Instrument window. You can also load individual Instruments of a particular kit or replace a piece from an existing kit with a piece from another kit. We'll discuss this further a little later on in the manual in the "Making Your Own Custom Drum Kits" section.

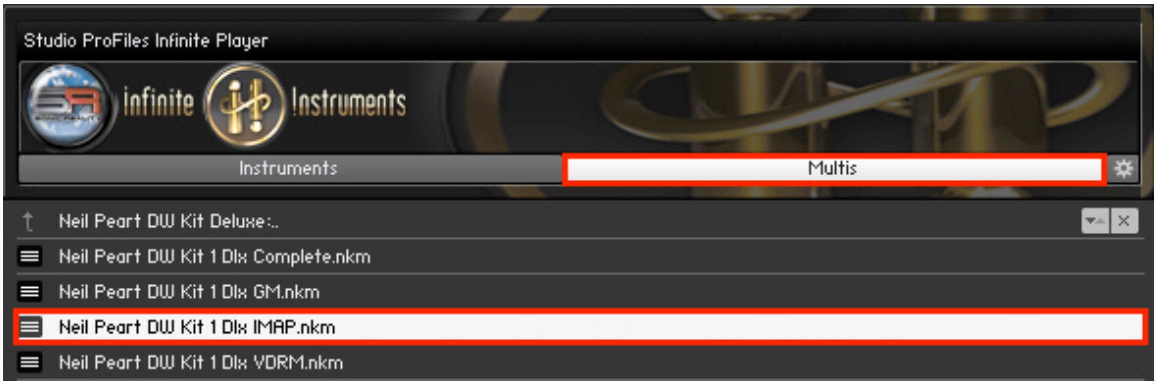
Kits are located under the Multis tab of your Infinite Player.



1. Navigate the "Neil Peart Drums" folder in the Multis section of your Infinite Player menu



2. Select the type of kit you wish to load.



3. Choose the kit's mapping format you want to work with.

Note: When you load a drum kit from under the Multis menu it loads all the Instruments for that particular kit. It also loads an "Output Multi" to which all your individual mic signals will be routed.

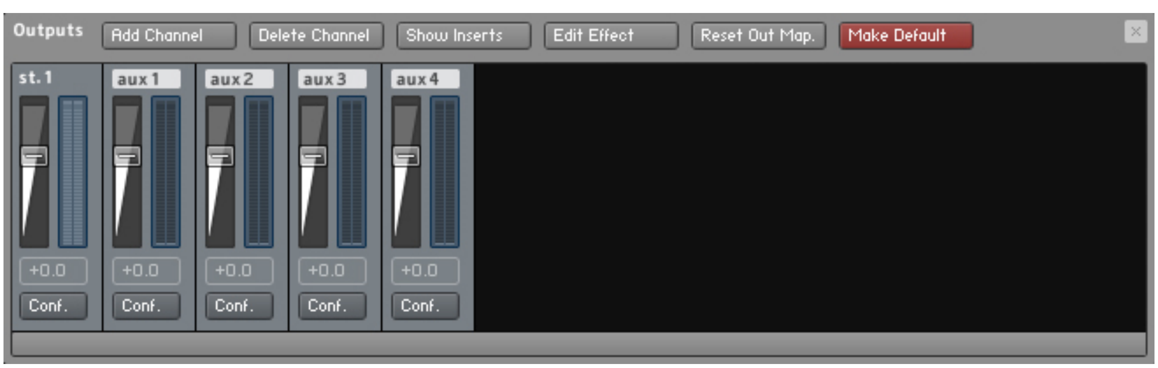
Making Your Own Custom Drum Kits

Drum Masters 2 provides the ability to swap out kit pieces to make your own custom drum kits. With a multitude of kit pieces available and the tweak functions provided, virtually endless kit combinations are possible. There are several ways you can make custom drum kits. One is to swap out kit pieces after you've loaded a drum kit Multi and then saving back your new custom Multi kit. That is the easiest and quickest method. Building multitrack kits piece by piece from scratch is also possible, just be sure to load the "Output Multi" first when working with multitrack kit pieces as the frame so that the individual mic channels of the kit pieces are routed properly.

Output Section and the Output Multi

Before diving into the details about customizing kits, we need talk about the Output Section and the Drum Masters 2 default Output Multis.

<u>Output Section</u>: Your Output section can be found by clicking on the "Outputs" tab located at the top of Kontakt Player. This is where you have control over the levels of all your individual mics. When loading the Kontakt Player Infinite Player you will see the Outputs looking like the following image. If you were to try to load an Drum Masters 2 Instrument at this point you would not have control over your individual mic mixing (as seen below). For this you **MUST** load a Multitrack Output Multi first.



The Drum Masters 2 MT Output Multi

The Output Multi is the default output you must load before loading ANY multitrack Instruments. Multitrack drum kits (full kits) on the other hand already have the Output Multi built into the “Multi” or .nkm. These Output Multis are always found under the Multis tab in their product specific folders. Once loaded, you will see an entire set of faders specific to the Outputs for any Drum Masters 2 product.

The Drum Masters 2 Output Multi is 16 faders as follows: Kick, Sub Kick, Snare Top, Snare Bottom, Hat, Tom 1, Tom 2, Tom 3, Tom 4, Tom 5, Tom 6, OH, Room 1, Room 2, Misc. 1, Misc. 2. No matter how many mic signals any Drum Masters 2 multitrack kit or has, these Outputs will ALWAYS show up the same. In some cases a drum kit may not have the same number of mics, etc.

Combined Signal: Having an Output Multi like this allows you to have kit pieces with shared loaded at the same time, with both being routed through the same set of faders. This gives you control over your mix like a recording console would. Ex: Turning the OH fader up would raises the level of the OH mic on all instruments containing an OH signal.

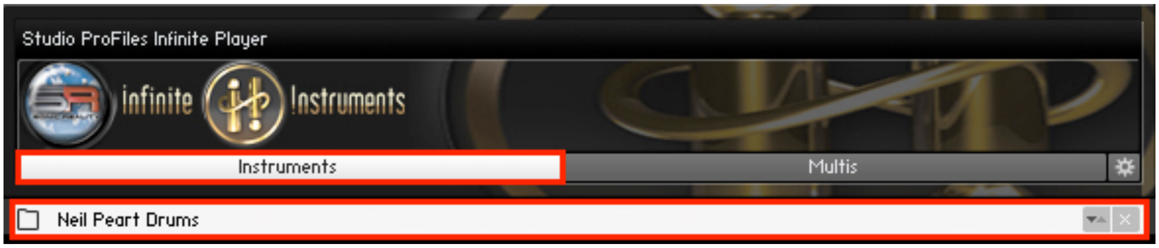
Swapping Instrument Kit Pieces

Let's say, for example, that after loading a kit, you want to change out the snare drum with another Drum Masters 2 snare. Simply close (**x-out)** the snare drum you wish to get rid of (*see below*).

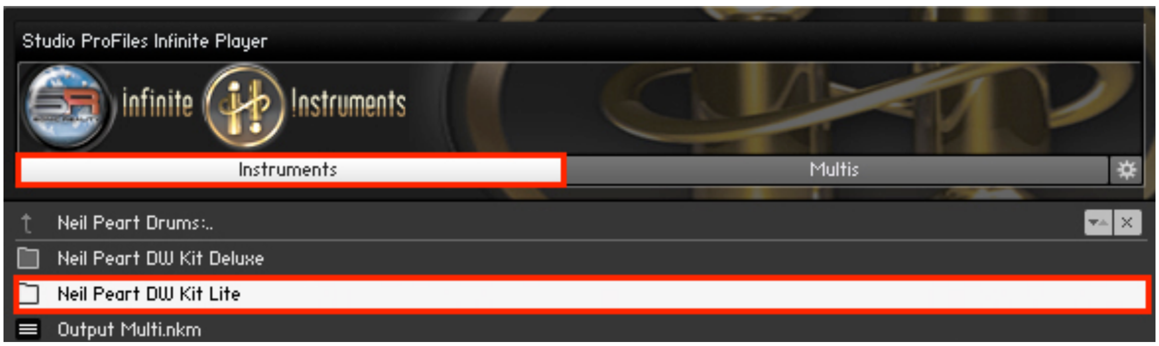
Navigate (as shown in the next section) to your individual Instruments under the "Instruments" tab on the Infinite Player. Click on the kit piece you want to add into your kit, and wait for it to load. Remember you can mix and match any individual Instruments (Kit Pieces) with one another.

Individual Instruments (Kits Pieces)

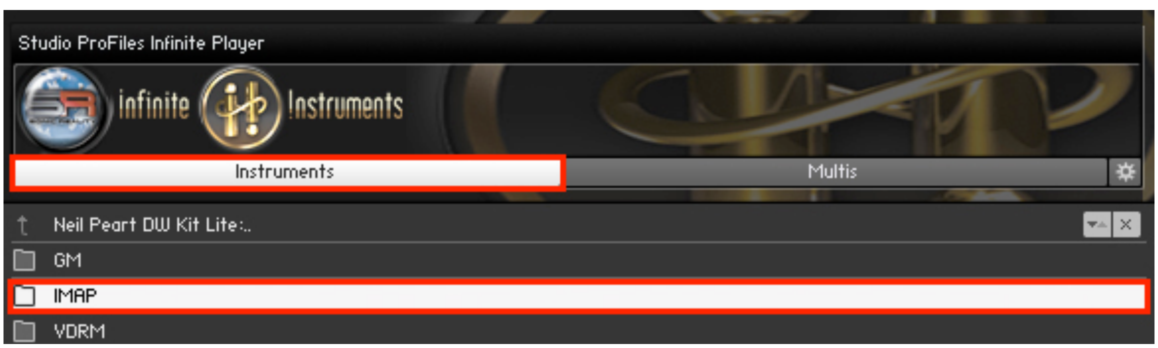
Individual Instruments can be found by clicking on the “Instruments” tab of the Infinite Player. Simply navigate to the kit piece you want to load and click. The kit piece will merge into your existing drum kit Multi.



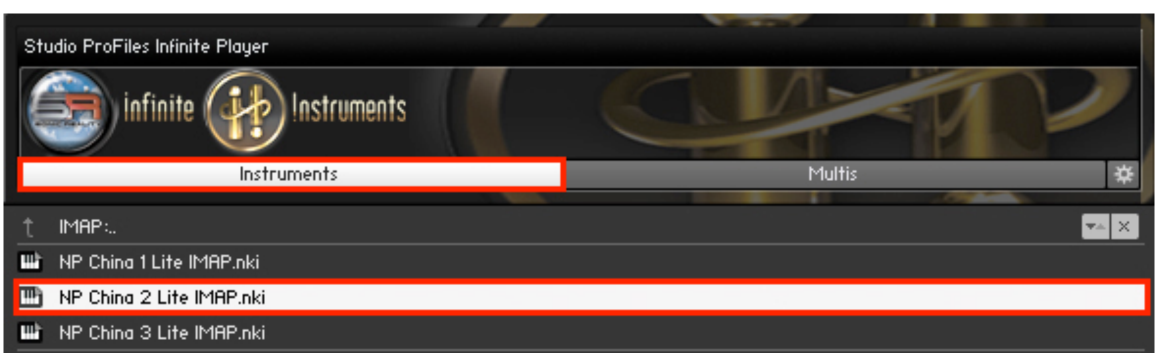
1. Click Instruments / Neil Peart Drums



2. Select the kit’s Instruments folder you want to choose from.



3. Choose which mapping you want your kit piece to open as.



4. Select the kit piece you want to load.

Note: If you load two of the same kit piece type (ex: two snares) they will overlap. In some cases this may cause problems. We recommend using one of each kit piece at a time.

Saving Back User Multis

After you've created your custom drum kit you may want to save it back so you can use it again. Saving a kit back is as simple as clicking on the Load/Save tab at the top of the Kontakt Player window and scrolling to "Save Multi as…".

Choose or create a folder to save your new Multi into. Most users will want to create a new folder aside or inside the *Multis / Drum Masters 2 Multitrack Kits / Specific artist folder /,* but any location is fine.

Please note that it ***is*** *possible to save over the original versions of your multis and instruments. Be very careful when saving, and we recommend making a backup of all original multis and instruments.*

Mapping Formats: Drum Kits

The Drum Masters 2 kits are set up to be used in a several mapping modes: iMap, GM, V-Drum, PERFORMANCE and CUSTOM.

Below, each of these modes are explained in detail. The modes can be easily changed per Instrument or for the entire Multi at once. Simply choose the mode from the "Mapping Mode" menu (as seen above) or toggle through the modes with a keyswitch by hitting C-2 or D-2 on your MIDI controller. *Note that this keyswitch is global; all instruments' mapping modes will change.*

Slots:
For any of the mappings, tom and cymbal Instruments have a special feature called slots. Tom and cymbal slots are positions on each of the maps where you can slot in toms or cymbals. This has been done to easily customize your mappings for your personal playing taste. There are some general differences between the mappings regarding this:

iMap: It is important to remember that iMap kits are made up of the original number of toms in that particular kit. If it was a 3 tom kit, the IMAP version will ONLY have 3 toms included in the Multi. By default these 3 toms will be mapped to slots 4, 5 and 6. To change this, simply switch the slot number under the mapping panel per tom.

GM: The GM mapping is always made up of 6 toms no matter how many the original kit had. In the case of a 3 tom kit, these toms would be brought back into the GM mapping and set to the other 3 open slots, then tuned up or down. Tuning can easily be achieved at the top of each Instrument. See page 43 for tuning Instruments.

V-Drum: This mode by default loads with 4 toms. Just like the GM version, if the original kit had 3 toms, a fourth is made, put in the last slot and pitched up or down.

Note: Setting two Instruments to the same slot will cause both Instruments to trigger from the same key. Only one Instrument should be mapped to each slot.

iMap:
iMap is Sonic Reality's proprietary map designed for "finger drumming" that is more expressive than general MIDI mapping. By having more performance articulations spread across the keyboard, Sonic Reality's iMap kits are able to offer more realism and performance nuances when played from the keyboard.

Since iMap has the largest amount of performance articulations of all 5 mapping types, here is a list of the performance articulations of iMap that other maps also pool from:

Kick: 2 (plus 2 "extra keys"*)

Snare: 7 (plus 4 "extra keys"*)

Center, Edge, Rimshot, Sidestick, Roll 1, Roll 2, Ghost

Hi Hats: 8 (plus 2 "extra keys"*)

Inside Closed, Bell, Edge, Foot Open, Foot Shut, Open, Open Edge (formerly referred to as "Bonham-style Open"), Choke.

Toms: 2 (plus 1 "extra key"*)

Rim, Center

Note: Toms can now be assigned to one of 6 tom "slots" across the keyboard so you can make any custom combination you want and both articulations get moved to the right positions depending on the slot you choose. For example, in iMap the tom "Rims" are always on the black keys so when choosing different slots (sets of 3 keys - for the two articulations plus extra key) the rim always ends up on the black key of that slot. This makes it a more intelligent feature than a simple transpose function.

Ride: 4 (plus 2 "extra keys"*)

Bell, Center (aka "Edge"), Crash (refers to "crashing" the ride), Special (additional misc performance or position on the ride).

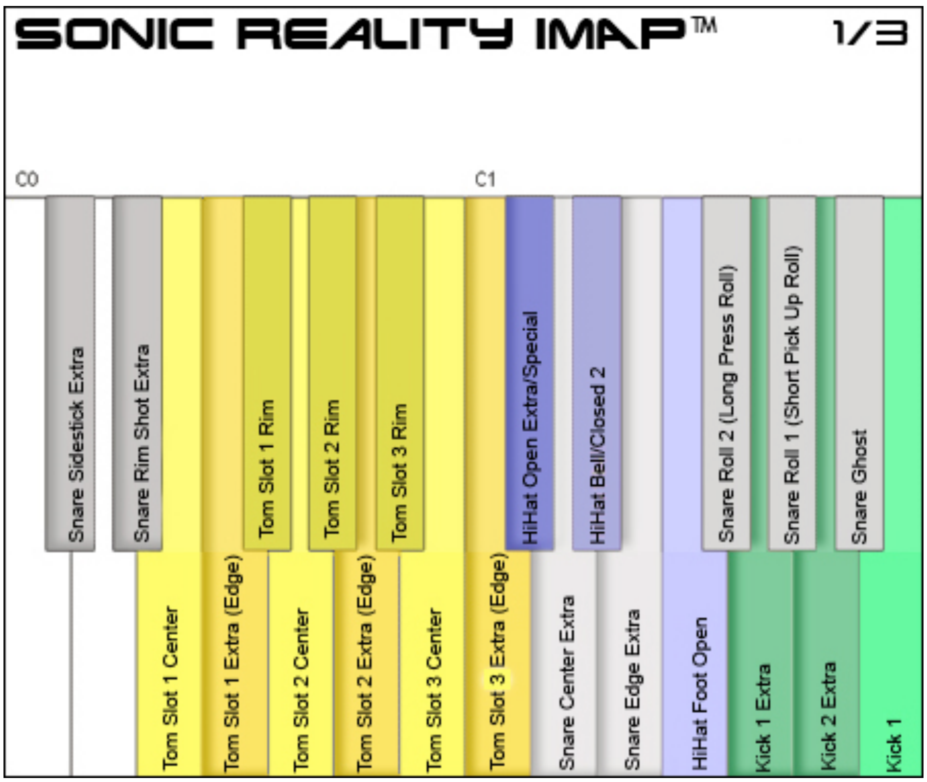
Crash/Splash/China: 2

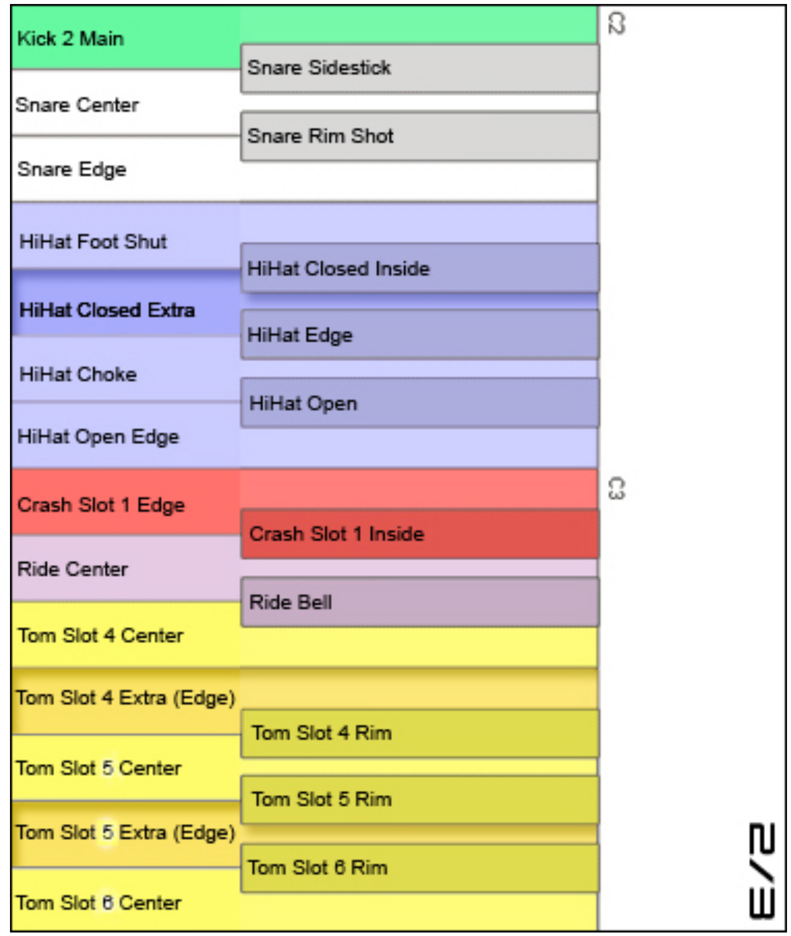
Inside, Edge

Note: Crashes, Splashes or Chinas can now be assigned to one of 5 different Cymbal "slots" across the keyboard so you can make any custom combination you want. Additionally, both articulations get moved depending on the slot you choose.

The iMap layout is shown below split into three sections.

**Middle C = C3 (Yamaha Convention)*

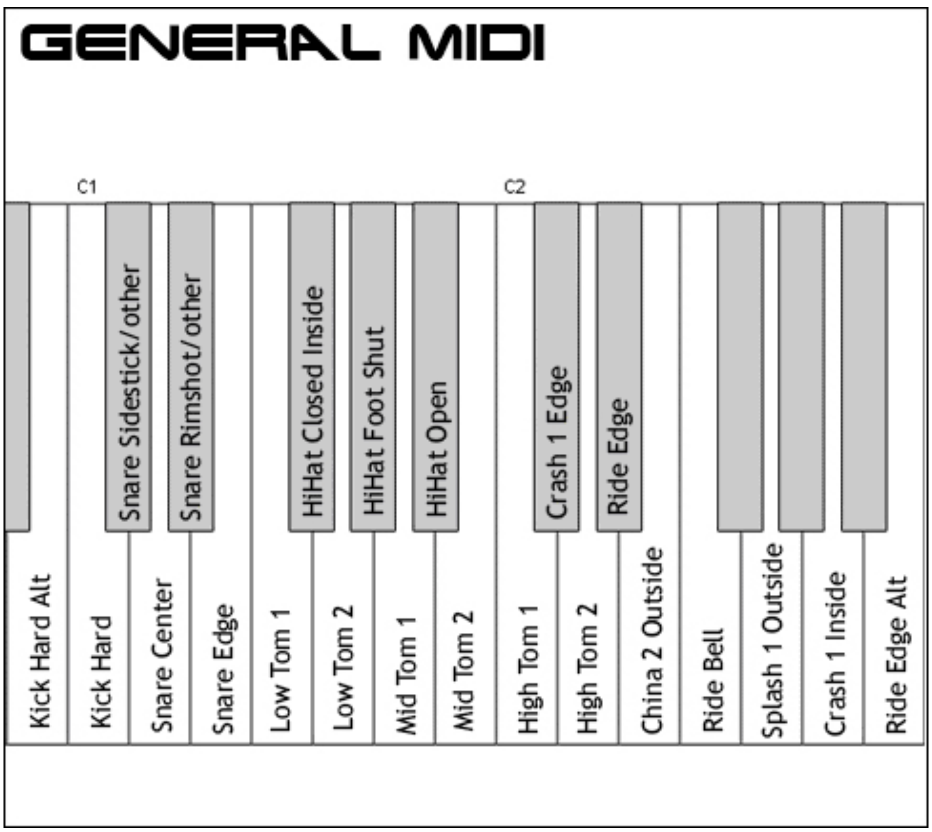



2/3
C2
C3
Kick 2 Main
Snare Sidestick
Snare Center
Snare Rim Shot
Snare Edge
HiHat Foot Shut
HiHat Closed Inside
HiHat Closed Extra
HiHat Edge
HiHat Choke
HiHat Open
HiHat Open Edge
Crash Slot 1 Edge
Crash Slot 1 Inside
Ride Center
Ride Bell
Tom Slot 4 Center
Tom Slot 4 Extra (Edge)
Tom Slot 4 Rim
Tom Slot 5 Center
Tom Slot 5 Rim
Tom Slot 5 Extra (Edge)
Tom Slot 6 Rim
Tom Slot 6 Center


3/3
C4
C5
Tom Slot 6 Extra (Edge)
Ride Special
Ride Center Extra
Ride Bell Extra
Ride Crash
Crash Slot 2 Edge
Crash Slot 2 Inside
Crash Slot 3 Edge
Crash Slot 3 Inside
Crash 1 Swell
Crash 1 Bell
Crash 1 Choke
Crash Slot 4 Edge Splash
Crash Slot 4 Inside (Splash)
Crash Slot 5 Edge China
Crash Slot 5 Inside (China)
Crash 3 Choke
Crash 5 China Ride
Crash 5 China Bell
Crash 4 Choke Splash
Crash 4 Bell Splash
Crash 2 Swell
Crash 2 Bell
Crash 2 Choke

GM:
GM, or General MIDI, is the most popular format for drums used in MIDI grooves, sequencers, standard MIDI files and more. It may not be the most powerful map in terms of performance realism but it is the most universally compatible. For those that are used to this popular map or want instant compatibility with most hardware and software MIDI devices, we offer the GM Mapping format option.

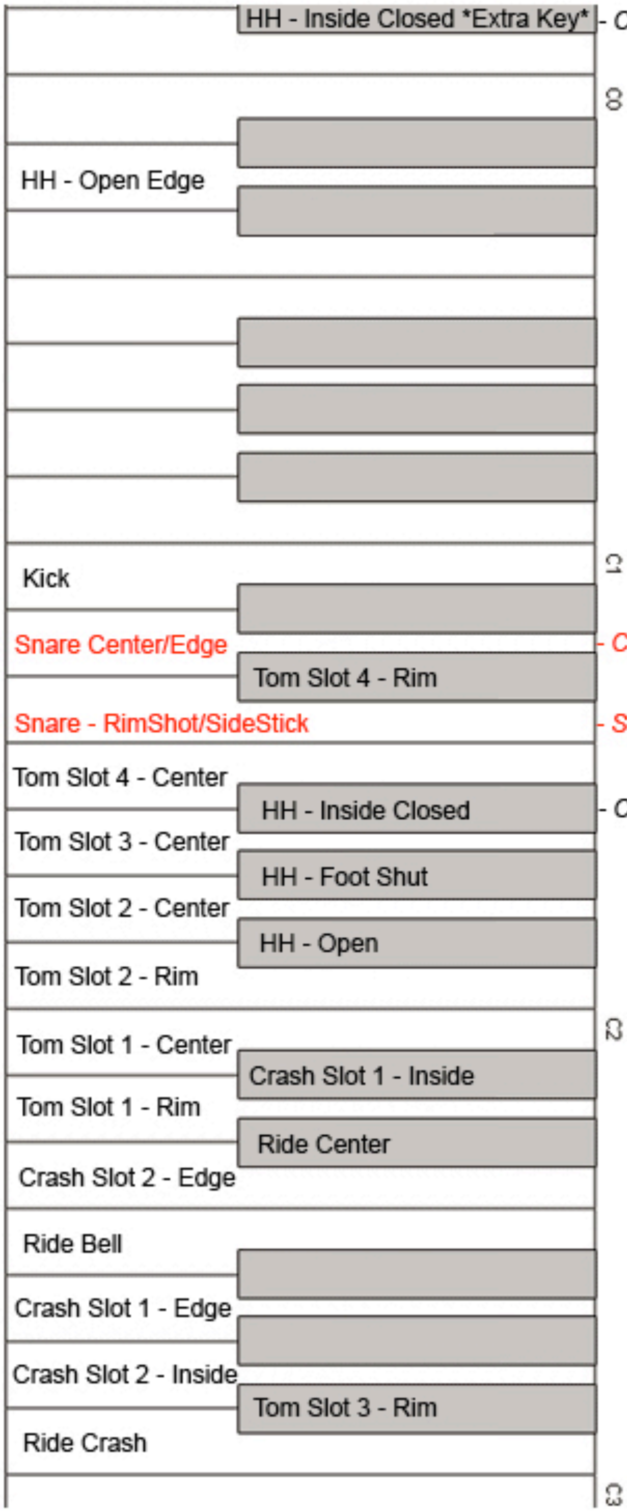
Note: In Sonic Reality's GM kits we replace the usual "Clap" with a more Drum Kit-oriented "Rimshot" and use the kick and snares for one kit as opposed to a variety of kits. GM Percussion is often not included in Drum Masters 2 Signature Kits but will be available separately as an add-on.



V-Drum:
This mapping mode is set up to respond to the Roland® TD20 V-Drums including advanced "position sensing" and other advanced features. However, it can also be used with other e-Drum kit brands and models as long as that hardware controller's pads are assigned to play the V-Drum map of Drum Masters 2. This mapping utilizes all of the main Drum Masters 2 hit articulations. Unlike the iMap, no extra hits or rolls have been added to the V-Drum patches, virtually keeping the characteristics of a real drum kit. Sonic Reality has also developed a new Hi-Hat script for V-Drum users which This new technology is explained further in the “Controls” section.

The Neil Peart kit has more kit pieces than the V-Drum brain allows for. The V-Drum maps supplied will work for the stock Roland set-up right off the bat, but we have also mapped two additional toms to the Aux 1 and 2 inputs. If you have extra triggers for these inputs, the tom inputs high-to-low would be Aux 1, Tom 1, Aux 2, Tom 2, Tom 3, Tom 4, Tom 5, Tom 6.

For maximum playability we recommend adjusting the sensitivity of your snare drum pad to suit your playing style. A setting of 10-12 has worked well for Sonic Reality’s developers. It is also recommended that the curve of the snare drum pad be changed to “SPLINE” (S curve). Please note that these are suggestions solely based on our in-house testing findings; customizing settings for all pads for each drummer’s playing style will increase realism and playability. Consult your user manual for your specific brand and model of e-drums for more.



- *CC4 controls Inside Closed *Extra Key* (tight edge samples) from medium open to open edge.*

- *Control the Center/Edge switch point on the V-drum Snare from your Snare mapping panel.*

- *Select RimShot or SideStick on the Snare mapping panel.*

- *CC4 controls Inside Closed from medium open to open*

Kick Drum

Snare Drum

Tom Slot 1

Tom Slot

Tom Slot 3

Tom Slot 4

Ride Cymbal

Crash Cymbal Slot 1

Crash Cymbal Slot 2

Hi-Hat

<u>Performance</u>:
Performance mode is a lighter version of Sonic Reality's iMap that is shifted an octave higher and has only 3 tom slots instead of 6 tom slots. The purpose of this map is to fit smaller sized keyboards such as 61 and 49 key controllers, as well as to offer larger sized controllers the ability to split the keyboard between a drum kit in performance mode and audio grooves, where applicable.

<u>Custom:</u>
If you don't want to touch your hardware set up's mapping and want to do it all from the software side, this could be more tedious because it would have to be done per kit. But the good news is that it is not only possible, but extremely easy to do with Sonic Reality's proprietary Custom map mode. Here you can assign ANY performance articulation of a kit piece to ANY incoming MIDI note via "MIDI Learn."

All you have to do is choose the performance articulation (be sure you are in CUSTOM mode by changing the Mapping Mode drop down to CUSTOM). Then click on "MIDI Learn" and hit the pad or MIDI note you want that articulation to be assigned to. It will "learn" the incoming note and automatically assign it (you can see that note show up visually below the MIDI learn button).

If you want to reassign it, just hit MIDI learn again, and the next incoming MIDI note will then be assigned. You would have to do this for every articulation you want played from your e-Kit controller and then save the whole kit back as a "Multi." (See page 19.)

*Please note that it **is** possible to save over the original versions of your multis and instruments. Be very careful when saving, and we recommend making a backup of all original multis and instruments.*

Neil Peart Complete Map:

Neil Peart's drum kit famously contains many kit pieces – a wide array of toms and cymbals. The sampled Neil Peart DW kit exceeds the size of all previous mapping conventions, so we have created an expanded map containing *all* of the kit pieces. It is essentially the iMap, extended in both directions as follows:

Note #	Note Name	Instrument	Articulation
021	A-1	Snare 2	Left Roll
022	A#-1	Snare 2	Right Roll
023	B-1	Snare 2	Ghost
024	C0	Snare 2	Center
025	C#0	Snare 2	Sidestick
026	D0	Snare 2	Edge
027	D#0	Snare 2	Rimshot
028	E0	Tom 1	Center
029	F0	Tom 1	Edge
030	F#0	Tom 1	Rim
031	G0	Tom 2	Center
032	G#0	Tom 2	Rim
033	A0	Tom 2	Edge
034	A#0	Tom 3	Rim
035	B0	Tom 3	Center
036	C1	Tom 3	Edge
037	C#1	Hi Hat 1	Open Extra
038	D1	Hi Hat 2	Inside
039	D#1	Hi Hat 1	Bell
040	E1	Hi Hat 2	Edge
041	F1	Hit Hat 1	Foot Open
042	F#1	Snare 1	Long Press Roll
043	G1	Kick	Extra 1
044	G#1	Snare 1	Short Pickup Roll
045	A1	Kick	Extra 2
046	A#1	Snare 1	Ghost
047	B1	Kick 1	Main 1
048	C2	Kick 1	Main 2
049	C#2	Snare 1	Sidestick
050	D2	Snare 1	Center
051	D#2	Snare 1	Rimshot
052	E2	Snare 1	Edge
053	F2	Hi Hat 1	Foot Short
054	F#2	Hi Hat 1	Closed Inside
055	G2	Hi Hat 1	Closed Extra
056	G#2	Hi Hat 1	Edge
057	A2	Hi Hat 1	Choke
058	A#2	Hi Hat 1	Open

059	B2	Hi Hat 1	Open Edge
060	C3	Crash Cymbal 1	Edge
061	C#3	Crash Cymbal 1	Inside
062	D3	Ride Cymbal	Center
063	D#3	Ride Cymbal	Bell
064	E3	Tom 4	Center
065	F3	Tom 4	Edge
066	F#3	Tom 4	Rim
067	G3	Tom 5	Center
068	G#3	Tom 5	Rim
069	A3	Tom 5	Edge
070	A#3	Tom 6	Rim
071	B3	Tom 6	Center
072	C4	Tom 6	Edge
073	C#4	Ride Cymbal	Extra
074	D4	Ride Cymbal	Center Alt
075	D#4	Ride Cymbal	Bell Alt
076	E4	Ride Cymbal	Crash
077	F4	Crash Cymbal 2	Edge
078	F#4	Crash Cymbal 2	Inside
079	G4	Crash Cymbal 3	Edge
080	G#4	Crash Cymbal 3	Inside
081	A4	Crash Cymbal 4	Edge
082	A#4	Crash Cymbal 4	Inside
083	B4	Tom 7	Center
084	C5	Tom 7	Rimshot
085	C#5	Tom 8	Center
086	D5	Tom 8	Rimshot
087	D#5	China Cymbal 1	Edge
088	E5	China Cymbal 1	Inside
089	F5	China Cymbal 2	Inside
090	F#5	China Cymbal 2	Edge
091	G5	China Cymbal 3	Edge
092	G#5	Splash Cymbal 1	Inside
093	A5	Splash Cymbal 1	Edge
094	A#5	Splash Cymbal 2	Edge
095	B5	Splash Cymbal 3	Edge
096	C6	Cowbell 1	
097	C#6	Cowbell 2	
098	D6	Cowbell 3	
099	D#6	Cowbell 4	
100	E6	Cowbell 5	
101	F6	Crash Cymbal 3	Choke
102	F#6	Crash Cymbal 3	Swell
103	G6	Crash Cymbal 4	Choke
104	G#6	Crash Cymbal 4	Swell

Controls for Drum Kits

The Drum Masters 2 GUI has a wide range of controls. The Menu drop-down menu is how you switch between your Volume, Pan, Envelope, FX, Mapping and Velocity panels. Each of these panels are home to a number of different controls. In this section we will define each panel.

Volume Panel

Under the volume panel you can adjust the level of each microphone channel for that specific Instrument. Stereo mics are combined for this panel .

Pan Panel

Under the pan panel, you can adjust the panning of each microphone channel independently. This includes overheads and room mic L/R channels. These were combined for volume but made independent for panning. This was done so you can flip or adjust the stereo image if you want.

Envelope Panel

The "Envelope" is for shaping the amplitude of the kit piece for the sharpness or softness of attack, the shaping of the decay, or more importantly for drums, the RELEASE setting which determines how long the sound plays out after being struck (on a keyboard it is how long it plays out after the key is released and on e-Drum kits it is how long it plays out after the pad is hit). While the envelopes are already set for natural response, it is possible to adjust the envelope for creative variation and / or shaping to suit a different "tightness or looseness" that may be desirable.

Under any envelope panel you will see a **Select All** and a **Solo Edit** button along with mic buttons towards the bottom of that particular Instrument. These are accompanied by the ADSR knobs (Attack, Decay, Sustain, and Release).

To set the ADSR for that particular Instrument, simply select the mics you wish to effect or **Select All**. Adjust your ADSR to affect the signal like you would any envelope.

Snare Instruments only: Along with the regular envelop parameters each Instrument shares, snare Instruments have separate ADSR for Snare Rolls 1 and 2. For these Instruments simply select the **Snare Roll 1** or **2 button** and adjust the ADSR separately for each roll.

FX Panel

The FX panel is home to another drop down menu which lets you add three 1 band EQs, two Compressors and a Limiter. Each effect can be added to any number or combination of mic channels for that Instrument. Simply select the mic channels you wish to affect by choosing them at the bottom of each FX panel, and adjust that effect.

Note: that these insert effects only affect that kit piece. Any additional effects added at the output mixer still do not replace the Instrument's insert effects but are "in addition to" and applied after in the chain.

FX Panel - EQ

Select the engage button to activate the EQ.

Freq. (Frequency): Adjusts the center frequency at which the boost or cut will occur.

Bandw. (Bandwidth): Adjusts the width of the frequency range that will be affected in octaves.

Gain: Adjusts the amount of boost (positive values) or cut (negative values) at the center frequency.

FX Panel – Compressor

Compressors are dynamic tools which automatically reduce the level of loud passages in a signal, thereby affecting the signal's dynamic range. They are invaluable tools for many common tasks. For instance, they can be used for reducing level peaks, thereby allowing the overall signal volume to be turned up without making it clip. In other words, increasing the average volume of a signal. With careful adjustment of the attack and release times, compressors can also modify signal transients, allowing you to add punch to weak-sounding drums or tame exaggerated "clicking" in percussion sounds. However, there is a point of diminishing returns; too much compression can result in a rather strained and weak sound.

Note: To engage (turn on) the compressor you only need to hit "Select All" or the button at the bottom of the Instrument interface with the name of the kit piece.

Thresh: Sets a level threshold above which the Compressor starts working. Only levels that rise above this threshold will be reduced by the compression; signals that stay below it will be left unprocessed.

Ratio: Controls the amount of compression, expressed as a ratio of “input level change” against “output level change”. A Ratio of 1:1 means that no compression will occur. A Ratio of 2:1 means that a level increase of 2 dB at the input will raise the output level by only 1 dB (keep in mind, though, that this applies only for input levels above the threshold). A 4:1 Ratio results in more aggressive compression, with a 4 dB level increase at the input causing a 1 dB increase at the output. Typical ratios for natural compression of Instruments are between 2:1 and 4:1.

Attack: Adjusts the time the Compressor will take to reach the full Ratio value after an input signal exceeds the Threshold level. If you’re using compression mainly for transparent dynamic reduction, values between 5 and 10 ms are a good starting point. Longer attack times can be useful for emphasizing transients and adding “punch” to a signal.

Release: Adjusts the time the compressor will take to fall back to non-compression after the input signal falls below the threshold. Typical values range from 50 to 250 ms.

Output: Controls the module’s output level. This knob acts as a make-up gain control, which allows you to bring the output signal up to the same peak level as the input signal after compression. After you’ve found a compression setting, it’s good practice to adjust the input and output signals so they have comparable levels, and then compare them via the Bypass button. This way, you can make sure your adjustment really made the signal sound better (and not just louder).

FX Panel – Limiter

Limiters are actually a special form of compressors with a ratio of one to infinity, a threshold just below the maximum level, and a very short attack time. They act as a "safety net" to keep short signal peaks from overloading the system, which would result in audio clipping. While compressors have a range of artistic applications, limiters are usually used for technical reasons – they can tame signals with peaks which would otherwise overload the output, without requiring you to turn the signal's overall volume down.

Note: To engage (turn on) the compressor you only need to hit "Select All" or the button at the bottom of the Instrument interface with the name of the kit piece.

In Gain: Sets the gain of the input signal. The Limiter is different from the Compressor in that it has a fixed threshold; to achieve a sensible peak reduction, use this control to adjust the input gain until you see the Attenuation meter responding only to occasional level peaks.

Release: Just like the Compressor's control of the same name, this knob adjusts the time it takes the Limiter to return to an unprocessed signal after the input level falls below the threshold.

Output: Adjusts the module's output level.

The Mapping Panel

The Mapping Panel is home to some of the most important features of Drum Masters 2. Most of these features have already been set up for standard users with default settings. We have also saved out multiple patches per drummer (iMap, VDRM, GM) to prevent you from having to switch the mapping manually.

With that said, this section will define the different areas of the mapping panel. One thing to note is that all Instruments' mapping panels will look basically the same other than a few exceptions. Some mapping modes have additional features that others do not have. One example is the "Settings" menu under any HiHat Instrument when in V-Drum Mode. These extras will be noted in this section:

Mapping Panel / Mapping Mode

Under the Mapping Mode drop down you can choose between Performance, iMap, GM, V-Drum and Custom mappings. For a detailed list and the features of these mapping modes please see page 26 of this manual. Some general controls are spread across the Mapping panel. You will always see Articulation, Alt, Round Robin, Mapped Key, and Trim tabs which are all explained in this section. When switching to certain modes on some Instruments, the controls may change. Here we will list which Instruments and in what modes have these additional controls.

Hihat Instruments in V-Drum Mode:
When any hihat instrument is put into the V-Drum Mapping Mode you will see a "Settings" button which does not appear in other Mapping panels.

Click on this "Settings" button to reveal the V-Drum hihat sample range settings. Here you have a Closed Range Start/End, Mid Range Start/End, Open Range Start/End. They are accompanied by On/Off buttons for each. There is also a Mid/Open Switch point knob. We suggest you turn all Ranges on or off. If they are turned off your hihat will be using the regular envelopes settings from the envelope panel.

The V-Drum hihat responds to V-Drum TD-20 hihat controllers automatically. In addition, although it is not required, there are settings for advanced users to fine-tune the performance of your hihat. Note that controller message 4 (CC4) sends the Open to closed message in the TD-20 (fully open HH will send velocity 0; a fully closed hihat will send velocity 127).

Closed Range:
This controls the release times according to CC4 of the closed group of samples.
Suggested settings: Start = 188.7, End = 1.1

Mid Range:
This controls the release times according to CC4 of the half open group of samples.
Suggested settings: Start = 98.7, End = 919.9

Open Range:
This controls the release times according to CC4 of the fully open group of samples.

Suggested settings: Start = 484.7, End = 3.0k

Mid / Open:
This controls the switch point between the mid and open range samples. This is also determined by CC4.

Note: When working with a V-Drum Hi-Hat an "extra key" MUST be assigned for the "Inside – Closed" articulation. This is important because the v-drum mapping calls from the "extra key" samples. If no "extra key" samples are assigned, the V-Drum Hi-Hat will not work properly. To read more info on "extra keys" please see page 48 of this manual.

Snare Drum in V-Drum Mode:
In the V-Drum snare you have the ability to choose if you want the snare drum rim to trigger a Sidestick or Rim Shot simply choose by hitting the SideStick/Rim button on this panel.

You can also adjust the amount of Center to Edge from the position sensor of the snare pad of a TD20. [The switch between Center and Edge is done via MIDI CC]

Tom Drums in GM Mode:
When in the GM mapping mode for any tom Instrument you will see a “Set Articulation” button and a “Slot” menu (see page 26).

When using a GM tom drum you must always choose which articulation you want to trigger, Center or Edge. Not doing so would cause this tom Instrument not to trigger when playing the slots mapped key.

Choose the articulation you wish to trigger.

Mapping Panel - Articulation

You can select which performance articulation you want to edit with the drop down menu under **"Articulations"**. You then see a full panel of knobs and buttons that adjust and save settings for that articulation (note: you must save back the entire kit as a Multi to save the settings permanently at the end of your session).

Mapping Panel - Alts

To the right of each articulation you will see the **"Alt"** section. An "Alt" is an alternate hit of that performance articulation. These alternate hits can be used either in a "round robin," which is a cycling or randomization of hits with subtle changes each time you hit the drum, or they can be assigned to be played from an "extra key*" (when an extra key option is offered). The number of Alts available varies from articulation to articulation, kit piece to kit piece and even kit to kit. Generally speaking, larger sized kits will tend to offer more variation in alternate hits.

When turned on (button turned white) per ALT, Round Robin causes the ALTs that have it on to "cycle" changing which ALT hit is played so that each time you play that note it changes the sound to another alternate hit. This is to not only offer more animation and realism, but also to avoid the "machine gun" effect when played in fast succession. Below it there is a special "Randomize" button which only has to be turned on ONCE and all of the ALTs of that articulation will play in random order instead if in a sequential cycle. This helps avoid predictable repeats of tones and often adds even more realism and variation to animate the sound of the drum.

*Note: You can adjust the **"Trim"** of the articulation to get the right balance of snare sidestick vs. snare center for example and other volume level adjustments you may want to do per articulation.*

Extra Keys Explained: The "Extra Key" feature is only used in iMap mode. It allows any of the ALTS of a performance articulation to be assigned to an additional key on the keyboard from the "Extended iMap". These are usually a key that is an octave away on the same note. For instance, a CENTER snare articulation is on D2. The "Extra Key" for the CENTER snare would be on D1 (an octave below). The EDGE snare is on E2 and the "Extra Key" for the edge would be on E1. In the case of the Rimshot and Sidestick, the "Extra Keys" are two octaves away. Everything is shown on the iMap image below which has all of the extras (which will only have samples there if assigned to the "Extra Key" button which is now optional).

These "Extra Keys" from the "Extended iMap" are now optional. In previous iMap products that had "Extended iMap" their purpose was to allow the user to have tonal variation via the use of additional keys on the keyboard. This was to imitate having a left and right stick position of the drum (a center and slightly off center snare for example). It allowed there to be a deliberately played tonal variation just by switching which key was hit for the articulations that had extra keys (such as snare, hi hat closed and open edge, ride center and bell, tom center/edge). Now, with the new Round Robin feature these tonal variations can optionally be used as part of the main articulation's round robin group simply by turning round robin ON for that articulation ALT and turning OFF the "Extra Key" button (so it is not white).

Note: As an added bonus enhancing the Extended iMap "Extra Key" feature, any ALTS that are assigned to an Extra Key are automatically put into their OWN "Round Robin" group separate from the main articulation key.

Velocity Panel

On this panel you can adjust the dynamic response of that whole kit piece by adjusting the "Velocity to Amp" parameter. As an extra tool for expression you can also adjust the "Velocity to Pitch," which is particularly useful for Instruments such as toms that rise in pitch when they are hit harder (with a real kit, not virtual). While the pitch variance in a real drum might depend on the looseness or elasticity of the drum head, by adjusting the "Velocity to Pitch" anything from realistic to drastic variation can be achieved for either enhanced expression or for a cool dynamic effect.

Tune

The tune knob on top of each Instrument make it’s easy to quickly tune your toms or any Instrument.

<u>Assigning Discrete Outputs in a DAW</u>

<u>Note</u>: These instructions have been described using Pro Tools. All DAWs work similar to one another with regards to multiple outputs with Kontakt. To find more information regarding individual outputs in your particular DAW please check:

www.SonicReality.com/support

If you have Kontakt Player open and don't want to record your multitrack Drum Masters 2 Kits to a stereo track, you have the ability to route each individual output to separate audio tracks.

- Open an instance of Kontakt Player in Pro Tools.

- Click on the Outputs button to bring up your Outputs. This is where you are going to assign your outputs.

- If you don't have a Drum Masters 2 kit or multitrack Output Multi loaded, open the "MT Output Multi" in the product specific "Multis" folder. This will bring up the correct outputs.

- Click the 

button.

- Save your session and quit Pro Tools.

- Re-open Pro Tools and your session. Your outputs in Kontakt Player are now configured with Pro Tools.

- Click the 

button at the bottom of each audio channel to assign the correct outputs.

NOTE: The "Kick" Channel (1&2), only can be played through the track it was instantiated on. So if you have Kontakt Player on an aux track, channels 1 and 2 will only be routed to that aux track. Because of the limitations of Kontakt simply bus that Aux track to an Audio track to record your Kicks audio.

Now in Pro Tools create 16 audio tracks and select your discrete outputs from your "Input/plug-in" audio input tab for each track. Note: Don't forget to input your bussed kick track onto a new audio track.

Now you have discrete outputs from Konakt Player into Pro Tools. Each Kontakt channel should now be routed to individual audio tracks as well as your kick (which is being bussed from your Aux containing your Plug-in instance) to a discrete audio track.

Drum Replacement with Drumagog

Another popular use of drum libraries and virtual Instruments is for “Drum Replacement”. This is when you have a live multitrack drum recording that you want to improve the sounds of the individual kick, snare and other drums by using the original discrete tracks to “trigger” other samples to replace them. There are a variety of different methods and programs out there to help facilitate doing them. One popular program for this is called Drumagog. Here is a quick guide to getting started using Drum Masters and Drumagog together. All DAWs are similar when it comes to using Drumagog. This guide has been made using Drumagog within Protools. For more information about using Drumagog with your particular DAW please refer to your DAW/Drumagog manual

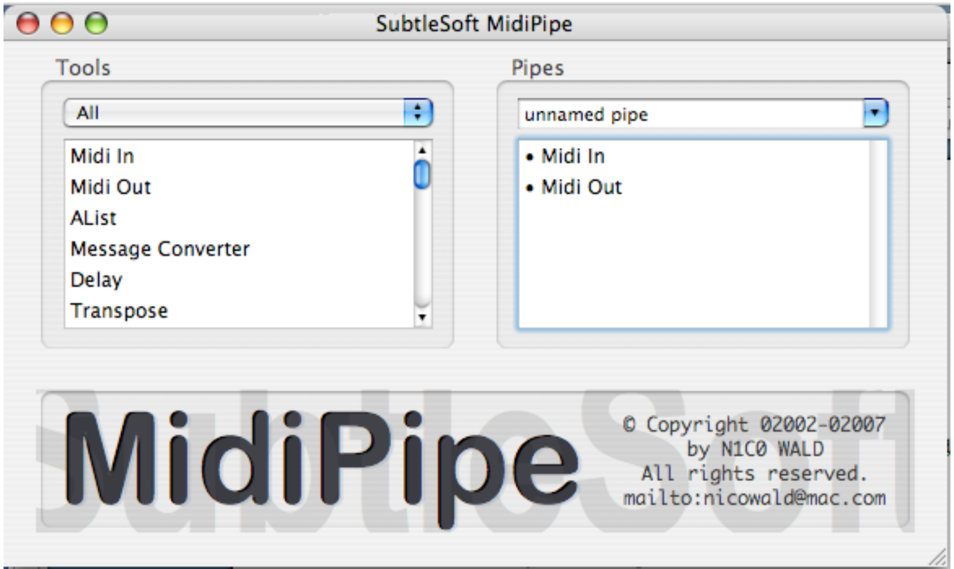
- Mac Users:

1. Download/install MidiPipe.
http://homepage.mac.com/nicowald/SubtleSoft

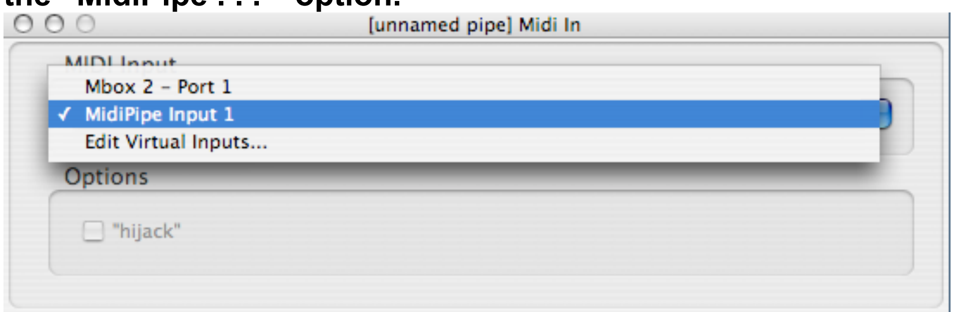
2. Launch MidiPipe.

3. Drag/drop an input and output from the left window to the right window. The right window MUST list “MIDI In” and “MIDI Out” in THAT ORDER.

You can set up multiple pairs of MIDI Ins and Outs, so that once you are working in Pro Tools you can use Drumagog on more than one audio track at a time. It is recommended to make a MidiPipe setup of ten to twenty virtual ins and outs, following these directions, and then save in MidiPipe. By saving a configuration in MidiPipe, you can then load this configuration for each new session.



4. Double-click "MIDI IN" and "MIDI OUT" in the "Pipes" list and choose the "MidiPipe . . . " option.



Repeat steps 3 and 4 for each desired set of virtual inputs/outputs, always selecting the next Input or Output number. (For your second set of virtual in/out, you will select "MidiPipe Input 2," "MidiPipe Input 3" for the third, etc.)

LEAVE MidiPipe OPEN.

5. Open Pro Tools. Start a new session or open an existing one.

6. In ProTools, go to Setup -> MIDI -> Input Devices. Select all "MidiPipe Output" options. (Note that MidiPipe reverses terminology for input/output in this case.)

7. Load the audio data you wish to replace.

8. Insert Drumagog into track. Open the inserted instance of Drumagog.

9. In the "Advanced" tab of Drumagog, enable MIDI Out and select the "MidiPipe Input 1." (See screenshot below)

10. Select appropriate MIDI channel. (1 by default)

11. Select the MIDI Note that corresponds to the note in Kontakt you wish to trigger - for example, in a bass drum track you might select “C2” to trigger one of Drum Masters 2 bass drum keys in an iMap mapped kit.

Repeat this for all tracks you wish to replace with Drumagog, each time choosing the next consecutive MidiPipe Input (MidiPipe Input 2 for the second instance of Drumagog, MidiPipe Input 3 for the third, etc.)

By adjusting the knobs on the “Main” tab of Drumagog you can further control Drumagog’s interpretation of the audio data. The sensitivity and resolution settings can be set visually by selecting “Visual” at the bottom right corner of the “Main” tab. For Instruments mic’d directly with sufficient isolation, the settings will be fairly cut and dry and easily manipulated. However, for

Instruments mic'd with insufficient isolation, such as overhead mics, the results will be less accurate and consequently harder to manipulate. Use of a gate (not included with Drumagog) will help isolate the desired hits.

Also note that, while triggering Kontakt via MIDI, Drumagog will also replace the selected track with the chosen (or default) audio sample from the Drumagog library. To avoid hearing this, completely turn down the Output knob in the "Main" tab of each Drumagog instance.

12. Set up an instrument track in Pro Tools. Insert Drum Masters 2 into this track. Load your desired kit.

13. Press play. This will trigger the Kontakt instance.

To record real-time replacement, send and record the instrument track to a new stereo audio track.

To manually edit Drumagog's output, record Drum Masters 2 output into the instrument track. From there, using Pro Tools' editing tools, notes can be deleted, moved, added, or embellished. This instrument track can later be bussed to a stereo audio track for audio recording.

- <u>Windows Users</u>:

1. Download/install MIDI-OX *and* the MIDI Yoke plug-in.
http://www.midiox.com
If you have Windows XP, no additional instructions are needed. If you have NT/98/ME/etc., follow the directions on the MIDIOX website. If you have Vista, disable the User Access Control (UAC) by following the following instructions.

1. Download MIDI Yoke installer
2. Run MSCONFIG from the Run menu
3. Go to the "Tools" menu and select "Disable UAC"
4. Press "Launch" and close the window. Restart.

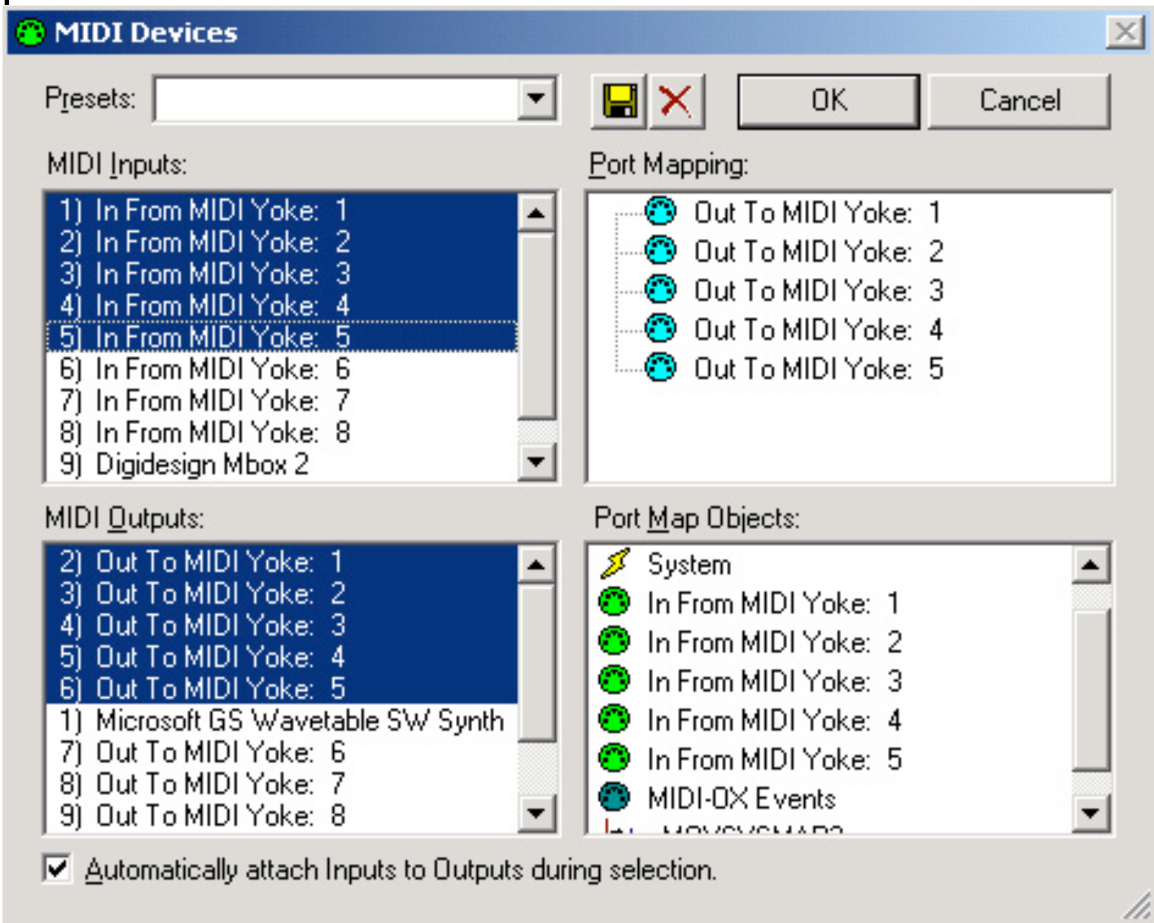
MIDI-OX will be installed as a standalone program, MIDI Yoke is a plug-in (and therefore will not be seen on your desktop, start menu, etc.)

Restart your computer after installation of both of these components.

2. Launch MIDI-OX.

3. In MIDI-OX, go to Options -> MIDI Devices. Double-click the same number of Inputs and Outputs so that they're added to the "Port Mapping" and "Port Map Objects" menus (see screenshot below). It is recommended

that you set up ten to twenty sets of in/out so that you will have ample ports to work with.



After setting up your desired virtual inputs/outputs, you can save the configuration as a preset, which can then be loaded as a template for each new session.

LEAVE MIDI-OX OPEN.

4. Open Pro Tools. Start a new session or open an existing one.

5. In Pro Tools, go to Setup -> MIDI -> Input Devices. Select all MIDI Yoke inputs.

6. Load the audio data you wish to replace.

7. Insert Drumagog into track. Open the inserted instance of Drumagog.

8. In the “Advanced” tab of Drumagog, enable MIDI Out and select the “Out To MIDI Yoke: 1” MIDI Port.

9. Select appropriate MIDI channel. (1 by default)

10. Select the MIDI Note that corresponds to the note in Kontakt you wish to trigger – for example, in a bass drum track you might select “C2” to trigger one of Drum Masters 2 bass drum keys in an I-Map kit. (See screenshot below)

Repeat this for all tracks you wish to replace with Drumagog, each time choosing the next consecutive MIDI-OX output (Out To MIDI Yoke: 2 for the second instance of Drumagog, Out To MIDI Yoke: 3 for the third, etc.)

By adjusting the knobs on the “Main” tab of Drumagog you can further control Drumagog’s interpretation of the audio data. The sensitivity and resolution settings can be set visually by selecting “Visual” at the bottom right corner of the “Main” tab. For instruments mic’d directly with sufficient isolation, the settings will be fairly cut and dry and easily manipulated. However, for instruments mic’d with insufficient isolation, such as overhead mics, the results will be less accurate and consequently harder to manipulate.

Also note that, while triggering Kontakt via MIDI, Drumagog will also replace the selected track with the chosen (or default) audio sample from the Drumagog library. To avoid hearing this, completely turn down the Output knob in the "Main" tab of each Drumagog instance.

11. Set up an instrument track in Pro Tools. Insert Kontakt into this trackand load your desired kit.

12. Press play. This will trigger the Kontakt instance.

To record real-time audio replacement, send and record the instrument track to a new stereo audio track.

To manually edit Drumagog's output, record Drum Masters 2 output into the instrument track as MIDI data. From there, using Pro Tools' editing tools, notes can be deleted, moved, added, or embellished. This instrument track can later be bussed to a stereo audio track for audio recording.

Effects

Drum Masters 2 offers a multitude of effects to create additional depth and variety of the products content. There are three places where you can add DSP FX:

1. Instrument panel FX Menu: As seen above. In the FX menu you can add DSP Insert FX per instrument.

2. Mixer Output Section: Like the FX Drum Masters 2, here you can add, remove or edit effects that will be global over any sounds passing through those outputs.

3. Mixer Aux Sends: Kontakt gives you four Aux channels in your Output section. Here you can add up to four different FX per Aux and have discrete Aux send levels per Drum Masters 2 Instrument.

This section provides descriptions of all effects modules that are available in KONTAKT, as well as explanations of their parameters. The term “effects” encompasses dynamics tools, such as compressors, as well as audio processors that change the signal in a usually non-linear way, such as reverb, flanger, or distortion effects.

Compressor

Compressors are dynamic tools which automatically reduce the level of loud passages in a signal, thereby affecting the signal’s dynamic range. They are invaluable tools for many common tasks. For instance, they can be used for reducing level peaks, thereby allowing the overall signal volume to be turned up without making it clip. In other words, increasing the average volume of a signal. With careful adjustment of the attack and release times, compressors can also modify signal transients, allowing you to add punch to weak-sounding drums or tame exaggerated “clicking” in percussion sounds. However, there is a point of diminishing returns; too much compression can result in a rather strained and weak sound.

Controls

Mode: Choose between Classic, Enhanced, and Pro mode. Each of these settings provides a different flavor of compression; if you feel you can't make a setting work with your sound, you should experiment with the other modes in this menu.

St.Link (Stereo link): When enabled, this causes the compressor to always act on the left and right channel in unison; this preserves the stereo image. When disabled, the Compressor becomes a dual mono processor, which means that both channels will be processed independently.

Thresh: Sets a level threshold above which the Compressor starts working. Only levels that rise above this threshold will be reduced by the compression; signals that stay below it will be left unprocessed.

Ratio: Controls the amount of compression, expressed as a ratio of "input level change" against "output level change". A Ratio of 1:1 means that no compression will occur. A Ratio of 2:1 means that a level increase of 2 dB at the input will raise the output level by only 1 dB (keep in mind, though, that this applies only for input levels above the threshold). A 4:1 Ratio results in more aggressive compression, with a 4 dB level increase at the input causing a 1 dB increase at the output. Typical ratios for natural compression of instruments are between 2:1 and 4:1.

Attack: Adjusts the time the Compressor will take to reach the full Ratio value after an input signal exceeds the Threshold level. If you're using compression mainly for transparent dynamic reduction, values between 5 and 10 ms are a good starting point. Longer attack times can be useful for emphasizing transients and adding "punch" to a signal.

Release: Adjusts the time the compressor will take to fall back to non-compression after the input signal falls below the threshold. Typical values range from 50 to 250 ms.

Output: Controls the module's output level. This knob acts as a make-up gain control, which allows you to bring the output signal up to the same peak level as the input signal after compression. After you've found a compression setting, it's good practice to adjust the input and output signals so they have comparable levels, and then compare them via the Bypass button. This way, you can make sure your adjustment really made the signal sound better (and not just louder)

Limiter

Limiters are actually a special form of compressors with a ratio of one to infinity, a threshold just below the maximum level, and a very short attack time. They act as a “safety net” to keep short signal peaks from overloading the system, which would result in audio clipping. While compressors have a range of artistic applications, limiters are usually used for technical reasons – they can tame signals with peaks which would otherwise overload the output, without requiring you to turn the signal’s overall volume down.

ControlsIn Gain: Sets the gain of the input signal. The Limiter is different from the Compressor in that it has a fixed threshold; to achieve a sensible peak reduction, use this control to adjust the input gain until you see the Attenuation meter responding only to occasional level peaks.

Release: Just like the Compressor’s control of the same name, this knob adjusts the time it takes the Limiter to return to an unprocessed signal after the input level falls below the threshold.

Attenuation: This LED-style meter shows the amount of gain reduction that the Limiter imposes on the signal. Limiting works best if this meter responds only to occasional level peaks; if it indicates permanent action, it’s a sure sign that your In Gain is set too high. This can considerably degrade the quality of your signal.

Output: Adjusts the module’s output level.

Inverter

With this module, you can invert the phase of your audio signal, or swap the left and right channels. Since the Inverter only makes sense as an insert effect, you can use this module only in the Group Insert Effects and Instrument Insert Effects chains.

Phase: Inverts the signal phase polarity.

Pan: Swaps the stereo channels.

AET Filter

This module constitutes the core of the powerful Authentic Expression Technology (AET) in KONTAKT 4, which allows you to “morph” continuously between the timbral characteristics of multiple samples. The module is designed to work on the Group level, so it can only be placed in the Group Insert FX chain.

As the process of creating and using a morph extends to elements of the user interface other than this module itself, we’ll start with a high-level explanation of the idea and technology behind AET; if you’re just interested in a reference of the involved dialogs and the module’s controls, you can find those at the end of this section.

About Authentic Expression Technology

When sampling acoustic instruments, one of the common difficulties to overcome stems from the fact that most of them change their tonal characteristics radically throughout their range of dynamic and expression – a French horn played *mezzo forte* can sound very different from one played *piano*. This makes it more difficult to capture the whole range of their sonic character in a convincing way with static samples alone. The traditional approach has been to tackle this problem with sheer size: Sample libraries that contain several dozen velocity layers per note have now become a common occurrence, and indeed, this is a good way to increase the sampled instrument’s level of detail. In a lot of cases, though, the shortcomings of turning highly dynamic instrumental sounds into static samples, such as noticeable timbre changes when moving from one velocity layer to the next, still persist.

With the Authentic Expression Technology (AET) in KONTAKT 4, we’d like to introduce a radically different approach. The core of this technology is an FFT filter with a very high resolution, which is able to “imprint” frequency responses of almost any complexity on your signal. These frequency responses are derived from other samples in your patch via spectral analysis. For instance, the currently playing velocity layer could be filtered with the spectral information of the layer just above it to sonically shift it closer to the latter. By dynamically varying the amount of processing with the help of a modulation source (such as the velocity), you can thereby pass through any number of intermediate stages between two samples, thus “morphing” between them in real-time.

The process is not limited to dynamic layers, either: You can morph between different playing techniques of an instrument (such as a trumpet playing with and without mute) or even between different signals. This opens up a whole range of sound design possibilities – fancy a morph between a piano and a celesta!

The functionality of AET is split up into two logically separate parts. In the analysis phase, you tell KONTAKT which Samples should be taken into consideration; it will then generate spectral “fingerprints” of these and save them for later use. This selection and setup process takes place in the Mapping Editor. Once the fingerprints have been created, they become available for selection within the AET Filter module, which you can insert into a Group Insert FX slot just like any other effect module. Here’s where the actual filtering takes place: With just a single parameter, you can control which of the stored frequency responses will be imparted to which amount on the currently playing sample.
Before we dive head-first into an actual use case of AET, we should explain two of the logical building blocks you’ll encounter in the setup process:

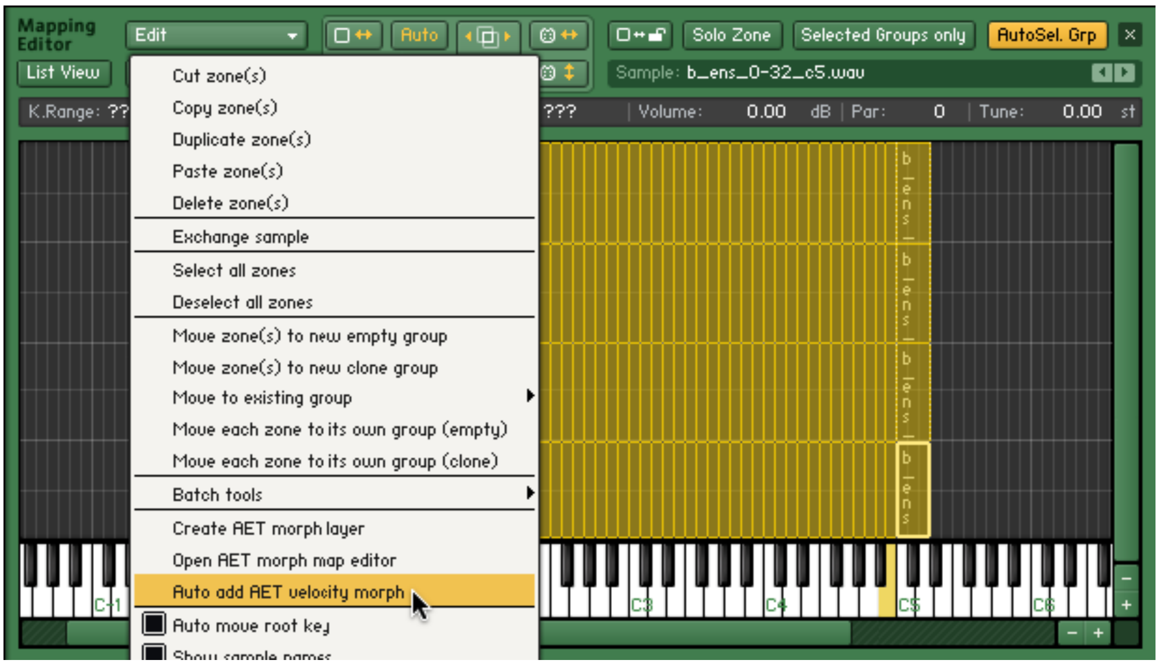
- A Morph Layer is a group of non-overlapping Zones whose samples are directly related in some way; these are often stacks of velocity layers or multi-sampled Zones that cover a certain key range, or in other words, blocks of adjacent Zones in the Mapping Editor.

- A Morph Map is a collection of one or more Morph Layers. This is what’s ultimately loaded into the AET Filter module, and its contents decide what the Morph knob on the module’s control panel will do. A Morph Map that contains only one Morph Layer is called a velocity map; in this case, the target frequency response will be derived from the Zone whose keyboard range matches the currently pressed key, and whose velocity range corresponds to the setting of the Morph knob. A Morph Map with multiple Morph Layers is the basis of an articulation morph; this is the more complex case, which allows you to set up morphs between two or more sets of Samples which are not necessarily related, such as different playing techniques or even entirely different instruments. Usually, the Morph Layers in such a map originate from different Groups in your Instrument.

- In short: If you want to use the expression filter for dynamic velocity morphing, a Morph Map with only one Morph Layer will do fine (the Auto add AET velocity morph function described below will even set this up for you); if you want to do more complex things, you’ll have to create a Morph Map that consists of multiple, manually defined Morph Layers. Now that you know what Authentic Expression Technology is and what it can do for you, let’s take a look at how to use it in real-world situations.

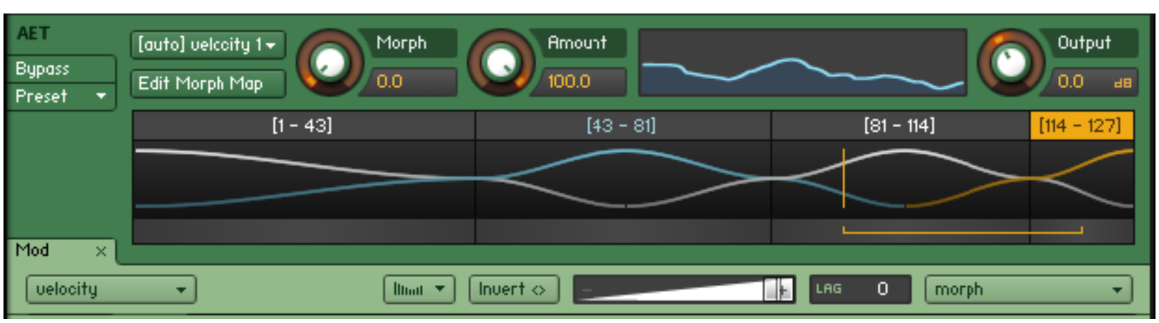
Creating a Velocity Morph

This is the basic use case, and since it is so common, KONTAKT offers you a function that automates it for you. Just load or create an Instrument with multiple velocity layers, select all of your Zones in the Mapping Editor (take care that you don't accidentally select multiple overlapping Zones; you might want to use the Selected Groups Only switch), then select the Auto add AET velocity morph command from the Edit menu.



You can find the "Auto add AET velocity morph" function both in the Edit menu and in the right-click context menu of the Mapping Editor.

When you open the Group Insert FX chain now, you'll notice that KONTAKT has added an AET Filter module for you. Take a look at its control panel and Modulation Router – it has already been set up with an auto-generated Morph Map, and the Morph knob is being modulated by the velocity.



The "Auto add AET velocity morph" function will add a ready-to-use AET Filter module to your Group Insert FX chain.

When you play some notes, KONTAKT will still only play the Sample that‘s assigned to the incoming velocity; the further the actual velocity is away from the middle velocity value of the played Zone, though, the more of the spectral characteristics of the Zone directly below or above will now be imparted on the sound. Once the velocity crosses the border to another Zone, the process will be reversed; now the next Zone will play, with the frequency response of the preceding one being used for filtering. This way, the timbral differences of your velocity layers are elegantly masked, resulting in a smooth dynamic response over the whole velocity range.

Of course, nothing stops you from assigning other modulation sources than the velocity to the Morph knob; you can just as well use a continuous MIDI controller or aftertouch for this purpose. Using a different modulation source for morphing makes it possible to sweep through the morph gradient while the sound is playing, which opens up very interesting performance possibilities. Keep in mind, though, that the filter will only act upon the Sample that was triggered when you pressed the key.

Creating an Articulation Morph

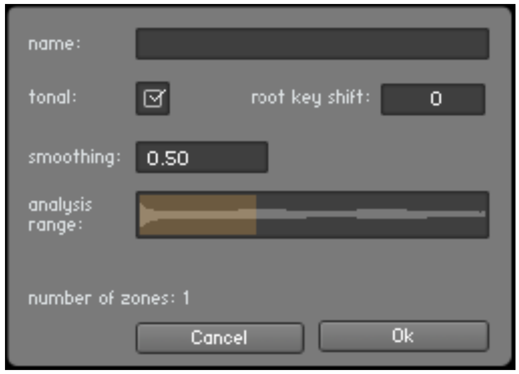
Setting up a simple velocity morph as described above is easy enough. There are times, though, when you need a more flexible way of configuring the morph filter. For example, you might want to set up a dynamically playable morph between two different, but musically related articulations of the same sound source, such as a choir singing “ahh” and “ooh”. This requires some more manual intervention. The procedure can be broken down into the following steps:

1. Create one Morph Layer for each of the Groups that the articulations are assigned to,

2. Combine two or more of these layers into a Morph Map, and

3. Load this map into an AET Filter module that you place in the Group Insert FX chain of the Group that should be processed.

In our example, we assume that the “ahh” and the “ooh” Zones are neatly split up into two different Groups, as would be the natural structure of a KONTAKT patch. In a first step, select all Zones from the “ahh” Group (using the Selected Groups only function if needed), then choose Create AET morph layer from the Edit menu. A dialog window with a few options will appear.

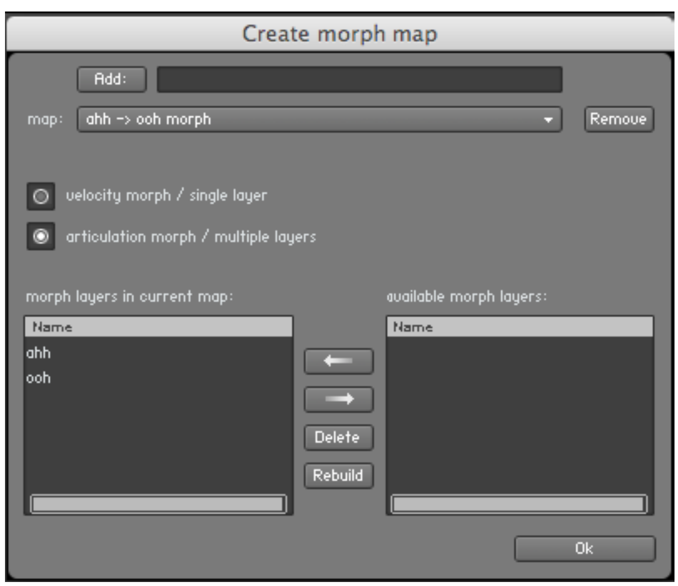


Before creating a new Morph Layer from the selected Zones, KONTAKT asks you to specify some details about the process.

Enter a descriptive name for your layer (such as “ahh”) and make sure that both the Tonal option is enabled and the Smoothing parameter is set to its default value of 0.5, then click OK. KONTAKT will now generate spectral fingerprints of all Zones and save them in a new Morph Layer. Once it has finished, repeat the process with the Zones in the other group. You have now created two new Morph Layers that contain spectral information which will be used in the resulting morph.

In order to tell KONTAKT which morph layers you’d like to use in your morph, you’ll have to create a new Morph Map and add those layers to it. To add a new Morph Map:

1. Open the AET Morph Map Editor from the Edit menu; a dialog window with an input line on top will appear.



2. Combine your Morph Layers into Morph Maps with the AET Morph Map editor.

3. Enter a descriptive name here (such as “ahh -> ooh morph”) and click the Add: button. The Morph Layers you just prepared will now turn up in the right one of two lists at the bottom of the dialog, meaning that they’re available for inclusion in your new Morph Map. Select them one-by-one in the right list and move them into your map by clicking the left arrow button. Note that the order in which you include them is important; in our example, we want the filter to morph the original “ahh” samples into the frequency response of the “ooh” samples, so the “ahh” layer has to be on top of the list.

4. Before you finish, make sure you switch the Morph Map type to articulation morph instead of velocity morph; otherwise, KONTAKT will display an error message, as velocity maps may only consist of a single Morph Layer. Click OK; your Morph Map is now ready for use.

Now would be a good time to stress the fact that in this scenario, the “ooh” samples will take no part whatsoever in the resulting sound; they just served as templates for setting up the morph filter. Even at the highest Morph setting, what you hear will actually be the “ahh” samples with the frequency response of the “ooh” samples superimposed. This also means that you should make sure that the “ooh” samples won’t be played when you press a key; the result would be a messy combination of filtered and unfiltered samples. The easiest way to do this is to turn the Volume knob of the “ooh” Group’s Amplifier Module all the way down.

To complete the process, switch over to the “ahh” Group, locate its Group Insert FX chain, and add an AET Filter module to an empty slot. Open the pull-down menu on the left side of its panel and select the Morph Map you just created. Now open the Modulation Router of the module and assign a controller, such as the modulation wheel, to the Morph parameter.

Setting the AET Filter module up in this way allows you to morph between your articulations in real-time using the modulation wheel.

That‘s it – when you play some notes now, you can seamlessly morph between “ahh” and “ooh” using the modulation wheel. Of course, you can also create morphs across more than two Morph Layers; by repeating the steps above, you could easily add another layer of the choir singing “mmh” to your patch.

Now that you know how to use AET in your own Instruments, we‘ll conclude this section with a description of the involved dialogs and their options.

Create AET Morph Layer Dialog

This dialog appears when you select the Create AET morph layer command from the Edit or context menu of the Mapping Editor. Its function is explained in section 22.4.3.

Name: The name of this Morph Layer, which will be used to identify it in the Morph Map Editor.

Smoothing: This value affects how much the analyzed frequency response will be smoothed out before it’s being used as a template for the filter. Sensible values range from 0.1 (no smoothing) to around 2.0 (strong smoothing).

Tonal: When this option is active, KONTAKT will analyze each sample harmonically in relation to the fundamental frequency of its respective root key value. You should only turn it off for material that’s entirely non-tonal, such as noise or percussion, or in case the root key value is wrong; in that case, KONTAKT will perform a plain frequency analysis with no respect to harmonic structures.

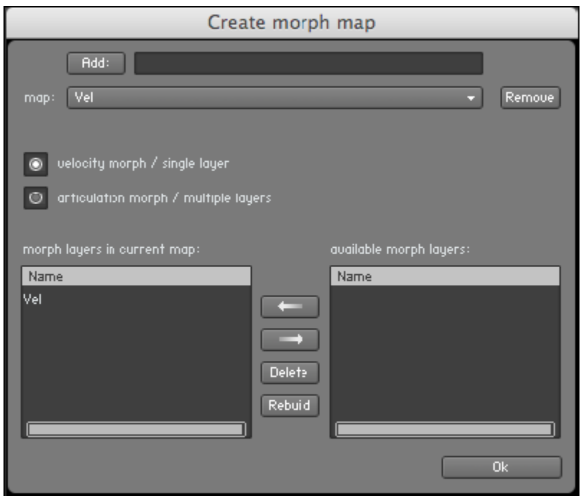
Root Key Shift: As described above, KONTAKT will use the root key value of a Zone to determine its Sample’s fundamental frequency when the Tonal option is enabled. There are cases, though, where the root key value deliberately differs from the true fundamental frequency of the Sample. For instance, you could be working on a patch that places multiple playing techniques of the same instrument in different octaves, so that C2 and C4 play the same note, but with different articulations. In that case, using the root key value without correction would mislead the analyzer into considering the wrong frequencies; with the Root Key Shift parameter, you can specify an offset between the actual note of the Sample and its root key value in semitones. The default value of 0 assumes a correct root key setting; a value of -12 corresponds to the actual fundamental tone being one octave below the root key.

Analysis Range: Click and drag the borders of the selection range here to limit the analysis process to a certain time frame of each sample. When analyzing samples that change their timbre over time, this allows you to tell KONTAKT at which point in a note the instrument will exhibit its most characteristic frequency spectrum, and exclude the rest from influencing the analysis result. For a piano or guitar, this would be the first seconds of each note; if you’d include the whole decay phase in the analysis, the average spectrum would turn out much darker.

Number of Zones: The number of Zones that are currently selected in the Mapping Editor and will be included in this Morph Layer when you click OK.

Morph Map Editor

This dialog appears when you select the Open AET morph map editor command from the Edit or context menu of the Mapping Editor. Its function is explained in section 22.4.3.



The AET morph map editor dialog.

Add: Enter a name in the input field next to this button and click on it to create a new, empty Morph Map.
Map: This pull-down menu contains all available Morph Maps of your Instrument. Select one to edit it.

Remove: Deletes the currently selected Morph Map from the Instrument, thereby bypassing any AET Filter modules that reference this map.

Velocity/Articulation: Specify the principal type of the Morph Map here. Velocity maps morph between the Zones of a single Morph Layer, while Articulation Maps morph between multiple Morph Layers. Refer to the previous subsection for a thorough discussion of the differences.

Morph Layers in Current Map: Contains the names of all Morph Layers that make up the current map. Selecting a layer and clicking the right arrow button removes it from the map.

Available Morph Layers: Contains the names of the Morph Layers available for inclusion in the current map. Selecting a layer from this list and clicking the left arrow button will add this layer to the Morph Map.

Delete: Selecting a Morph Layer from one of the lists and clicking this button will remove it from your Instrument.

Rebuild: This will re-open the Create Morph Layer dialog for the currently selected Layer, which allows you to readjust its analysis parameters.

AET Filter Module Controls

This is the processing module that performs the actual filtering of the source material. When using AET via the Auto add AET velocity morph command, KONTAKT will set this module up for you; in all other cases, you'll have to insert it into a Group Insert FX chain yourself.

The control panel of the AET Filter module. Shown here is an auto-generated velocity morph across six layers.

Morph Map: Use this drop-down menu to load a Morph Map into the module.

Edit Morph Map: Click on this button to open the Morph Map editor dialog explained in the previous section with the active Morph Map selected.

Morph: This is the most important parameter of the module; it should usually be modulated with an external source, such as the note velocity or a MIDI controller. Based on the contents of the selected Morph Map, KONTAKT will build a continuous "morph gradient" that combines and connects the filter responses required to achieve the various timbres of the included Morph Layers with smooth transitions. Using the Morph parameter, you can set the filter to any point in this gradient. At its lowest value, the signal is changed towards the respective sample in the first Morph Layer of the map (if this happens to be identical with the playing sample, the filter will be flat); at its highest value, the sample is changed towards the respective sample in the last Morph Layer of the map. You can follow this behavior graphically in the morph curve view described below.

Amount: The amount to which the filter influences the resulting signal. At the lowest setting, the filter has no effect.

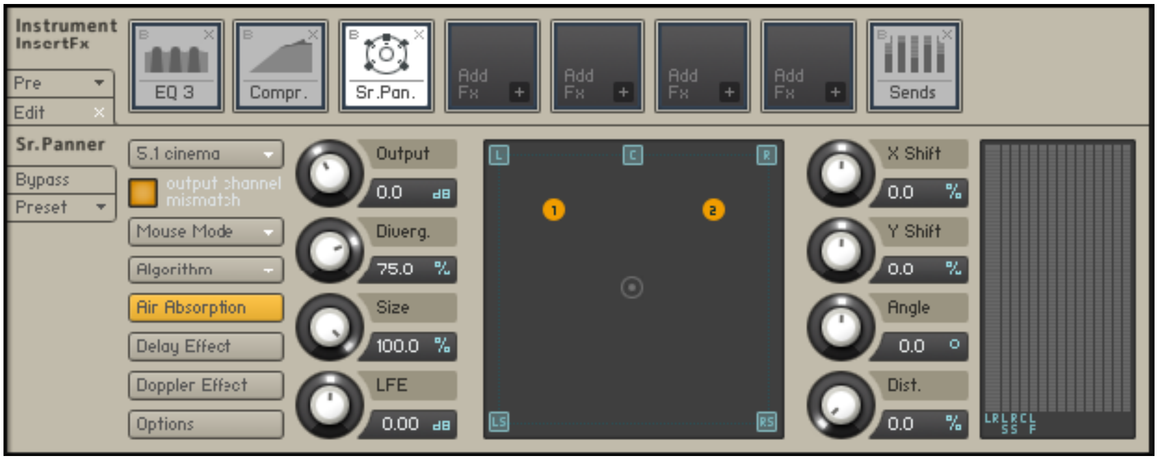
Filter Curve: This graph represents the actual, currently active filter response, which is the difference between the analyzed frequency response of the source (i.e. the currently playing
Sample) and the target (i.e. the Morph Layer that appears in the Morph Map at the point that's set with the Morph knob).

Morph Curve View: This is a graphical representation of the selected Morph Map and the resulting filter gradient. It's divided into multiple color-coded sections and curves, with the sections depicting the Morph Layers that make up the map, and the curves representing the amount to which the frequency response of the respective layer will influence the final filter response at any point. On top of each section, the name of the respective Morph Layer is being shown for reference. At the peak of each curve, the currently played sample will be morphed into the frequency response of the respective layer to the maximum amount. The section and curve of the active sample (i.e. the source) are marked yellow; these correspond to a flat filter response. The sections and curves of the other layers are alternatingly colored white and blue. When playing a note and changing the Morph parameter, a horizontal, angled bracket below the graph marks the source and target points of the gradient that KONTAKT uses to determine the final filter response.

Output: The output level of the module in dB.

Surround Panner

This module provides extensive and powerful surround mixing and automation capabilities to Group signals. It works with a multitude of input and output channel configurations, ranging from mono up to 16-channel surround sound, and allows you to place input signals as sound sources on a spatial plane, move them either manually or via automation, and simulate a range of natural dampening and doppler effects which occur when sound sources are being moved around the listener.



Menus and Buttons

Surround Format:
This drop-down menu selects the output format of the module. Options range from simple speaker-subwoofer splits (1.1) up to 16-channel surround formats, and also include a wide range of common cinema and music surround formats like 5.1, 7.1 and 10.2. Changing this setting does not affect the positions of your sound sources on the plane. The following table lists all available surround formats along with their channel assignments. Refer to these channel maps if you're unsure which channel will carry which speaker signal at the module's output.

Audiochannel #	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
1.1 Mono + LFE	C	Lf														
2.0 Stereo	L	R														
2.0 Stereo Vide	L	R														
2.1 Stereo + LFE	L	R	Lf													
2.1 Stereo Vide + LFE	L	R	Lf													
3.0 Surround (LRS)	L	R	S													
3.0 Front (LCR)	L	R	C													
3.1 Surround (LRS) + LFE	L	R	S	Lf												
3.1 Front (LCR) + LFE	L	R	C	Lf												
4.0 Surround (LRCS)	L	R	C	S												
4.0 Quadraphonic	L	R	Ls	Rs												
4.1 Surround (LRCS + LFE)	L	R	C	S	Lf											
4.1 Quadraphonic (+ LFE)	L	R	Ls	Rs	Lf											
5.0 Cinema	L	R	Ls	Rs	C											
5.0 Music	L	R	Ls	Rs	C											
5.0 Pentaphonic	L	R	Ls	Rs	C											
5.1 Cinema + LFE	L	R	Ls	Rs	C	Lf										
5.1 Music + LFE	L	R	Ls	Rs	C	Lf										
5.1 Pentaphonic + LFE	L	R	Ls	Rs	C	Lf										
6.0 Cinema EX	L	R	Ls	Rs	C	Cs										
6.0 Music EX	L	R	Ls	Rs	C	Cs										
6.0 Hezaphonic	L	R	Ls	Rs	C	Cs										
6.1 Cinema EX + LFE	L	R	Ls	Rs	C	Cs	Lf									
6.1 Music EX + LFE	L	R	Ls	Rs	C	Cs	Lf									
6.1 Hezaphonic + LFE	L	R	Ls	Rs	C	Cs	Lf									
7.0 Cinema	L	R	Ls	Rs	Lc	Rc	C									
7.0 Music	L	R	Ls	Rs	L2	R2	C									
7.0 Heptaphonic	L	R	Ls	Rs	L2	R2	C									
7.1 Cinema + LFE	L	R	Ls	Rs	Lc	Rc	C	Lf								
7.1 Music + LFE	L	R	Ls	Rs	L2	R2	C	Lf								
7.1 Heptaphonic + LFE	L	R	Ls	Rs	L2	R2	C	Lf								
8.0 Octaphonic Circ	L	R	Ls	Rs	L2	R2	C	Cs								
8.0 Octaphonic Rect	L	R	Ls	Rs	L2	R2	L3	R3								
8.1 Octaphonic Circ + LFE	L	R	Ls	Rs	L2	R2	C	Cs	Lf							
8.1 Octaphonic Rect + LFE	L	R	Ls	Rs	L2	R2	L3	R3	Lf							
10.2 Surround TH + 2 LFE	L	R	Ls	Rs	L2	R2	C	Cs	Lh	Rh	Lf	Lf				
12.0 Cinema Eztended	L	R	Ls	Rs	Lc	Rc	C	Cs	L2	R2	L3	R3				
12.1 Cinema Eztended + LF	L	R	Ls	Rs	Lc	Rc	C	Cs	L2	R2	L3	R3	Lf			
13.0 Cinema Plus	L	R	Ls	Rs	Lc	Rc	C	Cs	Lh	Rh	L3	R3	CC			
13.1 Cinema Plus + LFE	L	R	Ls	Rs	Lc	Rc	C	Cs	Lh	Rh	L3	R3	CC	Lf		
16.0 Cinema Surround	L	R	Ls	Rs	Lc	Rc	C	Cs	Lh	Rh	L3	R3	L2	R2	C2	C3
16.0 Theater Surround	L	R	Ls	Rs	Lc	Rc	C	Cs	Lh	Rh	L3	R3	L2	R2	C2	C3

Table of available surround formats, along with their channel assignments.

Explanation of channel abbreviations:

Left	L
Right	R
Left 2	L2
Right 2	R2
Left 3	L3
Right 3	R3
Center	C
Left Center	Lc
Right Center	Rc
Center Surround	Cs
Center Center	CC
Center 2	C2
Center 3	C3
Surround	S
Left Surround	Ls
Right Surround	Rs
Low Frequency Effects	Lf
Left High	Lh
Right High	Rh

If you need to re-assign channels to different outputs, you can do this on the Channel Routing page of the Amplifier Module (refer to chapter 20 of this manual for an in-depth explanation).

Mouse Mode: This drop-down menu allows you to choose from a list of algorithms which affect how your input sources will be positioned on the plane and respond to mouse movements.

- Mono Mix: All sound sources will be set to the same position.

- Sync: When you drag a sound source, all other sound sources will move along in the same direction.

- Center Mirror: Sound source positions will be mirrored at the center point of the plane.

- X Mirror: Sound source positions will be mirrored along the X axis.

- Y Mirror: Sound source positions will be mirrored along the Y axis.

- XY Mirror: Sound source positions will be mirrored along the X and Y axis.

- Individual: Each source can be positioned individually with the mouse.

Algorithm: Determines how the level of a sound source will be affected by its distance from the center. The drop-down menu offers three algorithms:

- Constant Power – this panning algorithm will adjust the relative speaker levels of a sound source in a way that preserves the source's apparent volume, regardless of its panning position. Placing a source icon right above a channel icon will isolate the source signal on the respective channel. Moving it around the plane will distribute its signal among the respective speakers in relation to their distance from the source, keeping the overall apparent volume (or, more precisely, the power) constant. This behavior will be affected by the Divergence setting.

Sinusoid – this algorithm uses a sine function to adjust the volume of a source in relation to its distance from each speaker. Setting the Divergence parameter to lower values will result in a more directional image. If you move a source far away from speakers and use a high Divergence setting, its level can drop to silence.

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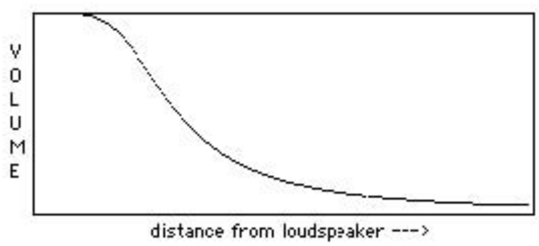
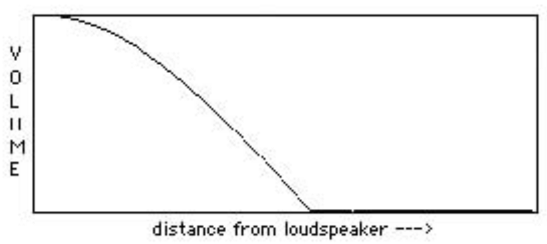
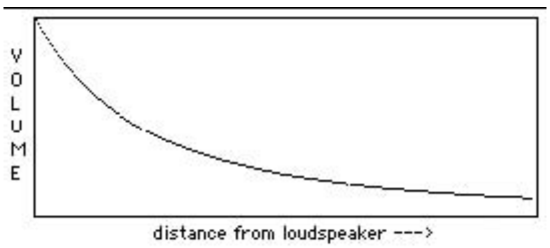


- Constant Power – this panning algorithm will adjust the relative speaker levels of a sound source in a way that preserves the source's apparent volume, regardless of its panning position. Placing a source icon right above a channel icon will isolate the source signal on the respective channel. Moving it around the plane will distribute its signal among the respective speakers in relation to their distance from the source, keeping the overall apparent volume (or, more precisely, the power) constant. This behavior will be affected by the Divergence setting.

- Sinusoid – this algorithm uses a sine function to adjust the volume of a source in relation to its distance from each speaker. Setting the Divergence parameter to lower values will result in a more directional image. If you move a source far away from speakers and use a high Divergence setting, its level can drop to silence.



- Logarithmic – this panning algorithm uses a logarithmic function to change the level of a source in relation to its distance from each speaker.

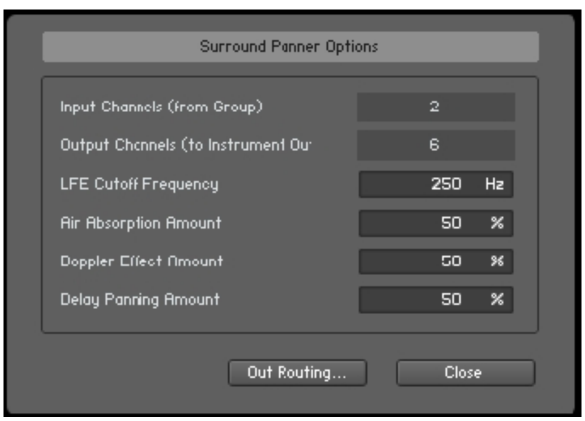


None of the above algorithms or modes are tied to a specific channel format. We recommend that you experiment with the algorithms and parameters in order to find the most suitable setting for your surround production.

Air Absorption: If a sound source moves away from a listener in the real world, the sound that reaches the listener will gradually lose its high frequencies. When this button is activated, the Surround Panner will simulate this absorption effect. If you'd like to increase the impression of distance further, even after you've dragged a sound source all the way to the plane border, increasing the size of the sound field with the Size control will gain you some more space.

Delay: As sound waves take some time to propagate through the air, sounds from sources which are further away from a listener will be delayed in relation to sounds in the vicinity. When this button is activated, the Surround Panner will replicate this effect with a delay line. Enabling this feature can dramatically improve localization, but it also uses a lot of CPU power. If you want to simulate positioning entirely with delays (and keep levels always constant), set the divergence control to 0%.

Doppler Effect: In the real world, this effect – usually associated with ambulances or racing cars going past – is a direct consequence of the delay and intrinsically tied to it; in the microcosm of KONTAKT, though, both effects can be separately controlled. When this button is activated, the Surround Panner simulates the pitch change when a sound source moves quickly toward or away from a listener. To hear this effect in action, move the sound source rapidly from one corner into the opposite one. It's more pronounced when the travel distance is longer, so you might have to zoom out with the Size control in order to hear it.



The Surround Panner's Options Dialog lets you adjust various parameters of the Doppler, Air Absorption, and Delay effects.

Options: This button will open a pop-up dialog which allows you to change the underlying parameters of the Doppler, Air Absorption, and Delay effects; furthermore, you can adjust the crossover frequency for the LFE channel. The dialog will also display the number of input and output channels that are currently in use. The button labeled Out Routing… will take you directly to the Channel Routing section of the Amplifier Module, where you can change channel routings and create mix-up or mix-down configurations to convert between channel layouts.

Controls

Output: Adjust the module’s output level.

Divergence: Adjusts the amount of distance-dependent level changes and, consequently, the directional focus of sound sources on the surround plane. Setting the knob to 0% will result in constant levels regardless of positioning.

Size: Adjusts the size of the surround plane. When set to 100%, the area surrounded by the speakers will fill out the plane window, so it won’t be possible to move a sound source beyond the limits of the speakers. Turning the knob counter-clockwise will “zoom out” and allow you to place sources outside the area of the speaker arrangement.

LFE: If the currently selected output configuration includes a low frequency effects (LFE) channel, this control adjusts its output level. Note that the LFE channel signal is derived from a sum of all input signals via a frequency crossover; the frequency at which the split occurs can be set in the options dialog.

X Shift: Imparts a constant offset on the X positions of all sound sources.

Y Shift: Imparts a constant offset on the Y positions of all sound sources.

Angle: Rotates all sound sources around the center point.

Distance: Adds a constant offset to the distances of all sound sources from the center point.

Meter: The output levels of all channels in the currently selected Surround Format are displayed here.

Surround Panner Automation

Automating the surround panner is particularly interesting for creating sounds that move around the room. There are several ways to accomplish this. If you need complete control over your motion patterns, you can use your host automation or external MIDI controllers to control the positioning parameters from outside KONTAKT. If you want to create automatic motion, using KONTAKT’s Modulation Router opens up a range of very interesting and creative possibilities.

Host/MIDI Automation

If you want to modulate the Surround Panner via your host or external MIDI controllers, you can assign external host automation data or MIDI controllers to the X Shift, Y Shift, Angle, and Distance parameters by dragging the respective automation sources from the Auto tab of the Browser onto the knobs you'd like to automate. For more information about external automation, refer to section 12.7 of this manual.

Internal Modulation

Using internal modulators on the Surround Panner offers a multitude of interesting possibilities, ranging from sounds that move around the room in a circular fashion to unpredictable and organic patterns of random motion. To create a modulation assignment, right-click on a knob and choose a modulation source from the drop-down menu, then adjust the assignment parameters and the source's controls (if any). After you have created an assignment, observe the Surround Panner's plane window when playing a note; you'll notice that for every static source icon, there's a darker icon moving along the plane. The bright icons display the sources' original positions (which can still be changed by dragging the icons around), while the darker icons depict the actual positions after all modulations have been applied.

It's hardly possible to describe the whole range of applications of modulating the Surround Panner's parameters, so we'll look at a few common scenarios instead. Re-creating these should get you a feel for what will be possible with some further experimentation.

- Circular motion. In order to rotate your sources around a pivot point, thereby creating a circular motion path, modulate the Angle parameter of the Surround Panner via a Sawtooth LFO. The distance of each source from the pivot point can be changed either by dragging the original positions, or via adjusting the surround panner's Distance parameter. To change the direction of the movement, enable the Invert button of the assignment in the Modulation Router. You can also synchronize the motion to the host or Master Editor tempo by switching the unit of the LFO's Freq. parameter to a note value.

- Random motion. This is a basic method to make your sources move around in an unpredictable way. Just assign two random modulators to the X Shift and Y Shift parameters. You can adjust how far the sources will stray from their original positions by adjusting the modulation intensities of these assignments.

Fly-by paths. Assigning envelope modulators to various parameters, most notably X Shift and Y Shift, allows you to define accurate and reproducible motion paths. It can get a while to understand how various envelope shapes relate to the resulting motion patterns, but once you've gotten the grip on it, you'll be able to create virtually all kinds of pre-defined motion paths – especially when you're using flexible envelopes.

Saturation

This module is basically an amplifier with a non-linear characteristic. It allows you to recreate the effect of tape saturation, which causes an increase of high-level energy in your signal.

Saturation: Adjusts the transfer curve. A negative setting results in a characteristic that will expand the signal – lower sample values will be attenuated, higher values will be amplified. Positive settings do the opposite and thus simulate the compression-like saturation of an analogue circuit. At a value of 0.0, the signal will pass the module unprocessed.

Output: Adjusts the module's output level.

Lo-Fi

This module adds various digital artifacts, like quantization noise or aliasing, to a clean signal. It's great for roughing up sounds that would otherwise be too plain and featureless.

ControlsBits: Re-quantizes the signal to an adjustable bit depth. Fractional bit levels (such as 12.4 bits) are possible and can add considerable "grit". Audio CDs have a quantization depth of 16 bits, old samplers frequently used 8 or 12 bits, and 4 bits evoke memories of countless irritating children's toys.

S.Rate (Sample Rate): Re-samples the signal to an adjustable sample rate. The re-sampling is done without any kind of (usually mandatory) low-pass filtering, which causes all kinds of wonderful aliasing artifacts. The sample rate goes all the way down to 50 Hz, which will not leave much of the original signal.

Noise: Adds hiss to the audio signal.

N.Color: Adjusts the frequency characteristic of the noise and acts as a low-pass filter.

Output: Adjusts the module's output level.

Stereo Enhancer

This module allows you to control the width of your signal's stereo base, change the panning, and create a pseudo-stereo signal from mono sources.

Pseudo Stereo: When activated, the module uses a pseudo-stereo algorithm to create a stereo signal from a mono source. This feature should only be used with mono signals and tends to create mono-incompatible sounds, which can disappear from a mix when it's being played back in mono.

Spread: Collapses (counter-clockwise) or expands (clockwise) your signal's stereo base. At the far left position, stereo signals will be summed to mono. Positive values will result in an artificial widening of stereo sources that has a tendency to apparently expand beyond the speakers, but watch out – just like the Pseudo Stereo feature, this tends to cause mono incompatibilities in your mix.

Pan: This control allows you to place your signal within the stereo field. It works exactly like the Pan control of the Amplifier module.

Output: Adjusts the module's output level.

Distortion

This module causes distortion by clipping or rounding off high sample values. It thereby simulates the behavior of overloaded transistor or tube circuits, adding artificial harmonics to a sound.

Mode button: Selects between either Tube or Transistor characteristics. Tube distortion creates a smooth saturation, which emphasizes even harmonics, while the Transistor setting generates odd harmonics that create a harsher-sounding clipping effect.

Drive: Adjusts the amount of distortion.

Damping: Turning this knob clockwise attenuates high frequencies in the output signal, thereby counteracting the brightness caused by the artificial harmonics.

Output: Adjusts the module's output level. Since distortion boosts the gain considerably, it's often necessary to attenuate the signal at the output stage.

Send Levels

This utility module can be added to the Group Insert Effects and Instrument Insert Effects chains and allows you to send a signal from within the insert chain to any existing send effect at adjustable levels.

Levels: On the left side of the Send Levels module, you'll see a level control for each send effect that's currently in one of your Instrument Send Effects slots – if you didn't add any send effects yet, the panel will be empty. The knobs allow you to adjust the level at which the signal will be sent to the respective effect.

Level meters: These LED-style peak meters provide visual feedback on the send levels.

Skreamer

This module offers an alternate overdrive algorithm that sounds warmer and smoother than the Distortion effect.

Tone: Controls the brightness of the sound. Turning this knob clockwise will result in a more pronounced top-end, which works great on bright, screaming leads and biting rhythms. Turning it counter-clockwise results in a mellower, darker sound.

Drive: Adjusts the amount of distortion.

Bass: Adjusts the low frequency gain.

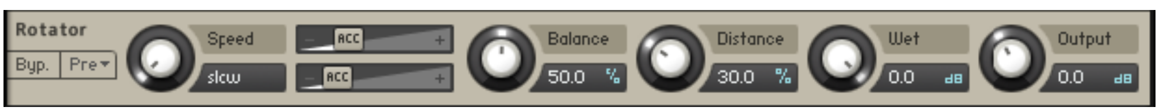
Bright: Adjusts the high frequency gain.

Wet: Adjusts the ratio between clean and distorted tone. At the maximum setting, only the wet signal will be heard.

Output: Adjusts the module’s output level.

Rotator

The Rotator effect realistically simulates the sound of rotating speaker cabinets, which are commonly associated with drawbar organs that became popular in rock music of the 60s and 70s. Although the effect is almost intrinsically tied to “the” prototypical drawbar organ sound, it works equally well on guitars, synth pads, and a wide range of other sounds.



Speed: Although this parameter appears as a knob in order to facilitate automating, it really only has 2 positions – Slow and Fast. A change of this setting realistically simulates the acceleration or braking of the rotor.

Acceleration and Brake Speed (horizontal faders next to the Speed control): These adjust how quickly the rotors of the treble (upper fader) and bass (lower fader) parts of the cabinet will react to speed changes. At the rightmost position, the respective speaker will change its speed instantly, while it will take a long time to reach its designated speed with the fader at the leftmost position.

Balance: Controls the relative levels of the cabinet’s treble and bass parts.

Distance: Controls the simulated distance between the cabinet and the pickup microphones. A closer distance results in a wider stereo panorama.

Twang

The Twang effect simulates the rich tube sound of classic guitar amps from decades ago. It’s ideal for screaming leads and crunchy rhythm guitar sounds, as well as clean sounds with personality.

Bright: A tonal option which increases the high frequency content of the signal.

Polyphonic: If this button is inactive, the Twang module will work as a mono effect, which causes stereo signals to be summed to mono at its input; when active, the effect operates on each input channel separately.

Volume: Controls the input level. In contrast to the Output knob, which merely adjusts the overall level of the module, this knob works like the gain control of a guitar amp and affects the amount of distortion.

Treble, Mid and Bass: These controls adjust the respective levels of the signal’s high, midrange, and low frequency components.

Output: Adjusts the module’s output level.

Cabinet

This module simulates the sound of a guitar cabinet recorded through a microphone. By following a distortion effect (like the Skreamer) with this module in your insert chain, you can simulate a complete guitar amp.

Cabinet Type (small pane at the left side of the module): Allows you to choose the simulated cabinet model via the up and down buttons.

Size: Adjusts the size of the simulated cabinet. Larger cabinets tend to have a more pronounced bass response, while smaller cabinets can sound thin and tinny.

Air: Controls the level of early reflections in the room response, adding a sense of space to the sound.

Treble (upper horizontal fader): Boosts or cuts the level of the higher frequencies.

Bass (lower horizontal fader): Boosts or cuts the level of the lower frequencies.

Output: Adjusts the module's output level.

Phaser

This effect continually changes the phase relationships in your signal with an all-pass filter. This results in a comb filtering effect, which attenuates some frequencies while boosting others. The sound is similar to that of a flanger, but in a more subtle manner.

Depth: The amount of LFO modulation. Higher values cause the phaser effect to sweep over a wider frequency range.

Speed: The LFO modulation speed. To synchronize the speed to your host or Master Editor tempo, click on the Speed control's unit display and choose a note length value from the drop-down list.

Phase (0 to 90 degrees): Imparts an LFO phase difference between the left and the right stereo channel. This can considerably increase the width of the output signal's stereo base.

Feedback: This control adjusts the emphasis of the peaks and notches that the comb filter effect imparts on the signal.

Return (visible when used as Send Effect): Adjusts the return level of the module's output signal.

Dry and Wet Sliders (visible when used as an Instrument Insert Effect): Adjusts the respective levels of the original and processed signals. Note that the typical phasing effect is created by the combination of both signals, so setting these to the same levels results in the most pronounced effect.

Flanger

This module splits the audio signal up and delays one version in relation to the original signal. By modulating the delay time, as well as feeding an adjustable amount of the output signal back into the input, the Flanger creates a characteristic “whooshing” sound. Just like the Phaser module, the Flanger uses a separate LFO for each stereo channel, with the phase relationship between both LFOs being adjustable.

Depth: The amount of LFO modulation. Higher values cause the flanging effect to sweep over a wider range.

Speed: The LFO speed. To synchronize the speed to your host or Master Editor tempo, click on the Speed parameter’s unit display and choose a note length value from the drop-down list.

Phase (0 to 90 degrees): Imparts an LFO phase difference between the left and the right stereo channel. This can considerably increase the width of the output signal’s stereo base.

Colour: Adjusts the delay line’s range of operation and, consequently, the color of the flanging effect. Small values result in short modulated delay times, making the Flanger sound more like a phaser.

Feedback: Feeds a certain amount of the delayed signal back into the module’s input, thereby creating a more pronounced effect.

Return (visible when used as a send effect): Adjusts the module’s return level.

Dry and Wet sliders (visible when used as an Instrument insert effect): Adjusts the respective levels of the original and processed signals. Note that the typical flanging effect is created by the combination of both signals, so setting these to the same levels results in the most pronounced effect.

Chorus

The Chorus module “thickens” the audio signal by splitting it up and detuning one version in relation to the original. Separate LFOs with an adjustable phase relationship detune each stereo channel independently for creating wide-panorama effects.

ControlsDepth: Adjusts the range of modulated detuning. Higher values give a more pronounced chorusing effect.

Speed: Adjusts the LFO speed. To synchronize the speed to your host or Master Editor tempo, click on the Speed parameter's unit display and choose a note length value from the drop-down list.

Phase (0 to 90 degrees): Imparts an LFO phase difference between the left and the right stereo channel. This can considerably increase the width of the output signal's stereo base.

Return (visible when used as a send effect): Adjusts the module's return level.

Dry and Wet sliders (visible when used as an Instrument insert effect): Adjusts the respective levels of the original and processed signals. Note that the typical chorus effect is created by the combination of both signals, so setting these to the same levels results in the most pronounced effect.

Reverb

This module simulates the natural reverberation that occurs when a sound source is placed in an acoustic environment, thus adding a feeling of spaciousness to the sound.

Controls

Pre-Dly.: Introduces a short delay between the direct signal and the reverb trail build-up. This corresponds to the natural reverberation behavior of large rooms, where a short time elapses before the first reflection of a sound wave returns from a wall.

Size: Adjusts the size of the simulated room. This affects the duration of the reverb trail.

Colour: This control allows you to adjust the construction material of the simulated room and, consequently, the color of the reverb trail. Low values simulate softer surfaces like wood, while high values simulate the reflection behavior of hard surfaces like concrete.

Damping: Sets the amount of simulated absorption that takes place in rooms due to furnishings, people, or acoustic treatments affecting the reflection behavior.

Stereo: Higher values increase the stereo base width of the output signal. Lower values simulate a closer distance to the sound source.

Return (visible when used as a send effect): Adjusts the module's return level.

Dry and Wet sliders (visible when used as an Instrument insert effect): Adjusts the respective levels of the original and processed signals. In common scenarios, the reverb signal is mixed in at a lower level than the direct signal.

Delay

This module offers a delay line that can optionally be synced to the tempo and provides an adjustable feedback level, a low-pass filter, and a pan control for ping-pong echo effects. If you don't use the tempo syncing feature, the available delay range is 5 to 2900 ms. Delay times lower than 20 ms are not discernible as delays, but can produce interesting comb filtering effects.

ControlsTime: The delay time in milliseconds. To synchronize the time to your host or Master Editor tempo, click on the Speed parameter's unit display and choose a note length value from the drop-down list.

Damping: Attenuates high frequencies in the delayed signal. Turning this control clockwise will increase the damping effect. If you have set a feedback level, the signal will gradually lose more high frequency content with each repetition.

Pan: Setting a value higher than 0 creates a panning effect, which alternates echos between the left and the right side of the stereo panorama – this is affectionally called a ping-pong delay. Higher values will result in wider panning; at 100, signals alternate between the far left and far right channel.

Feedback: Controls the amount of the output signal that's being fed back into the input of the delay line, thereby creating a series of echos that gradually fade into silence.

Return (visible when used as a send effect): Adjusts the module's return level.

Dry and Wet sliders (visible when used as an Instrument insert effect): Adjusts the respective levels of the original and processed signals. In common scenarios, the delayed signal is mixed in at a lower level than the direct signal.

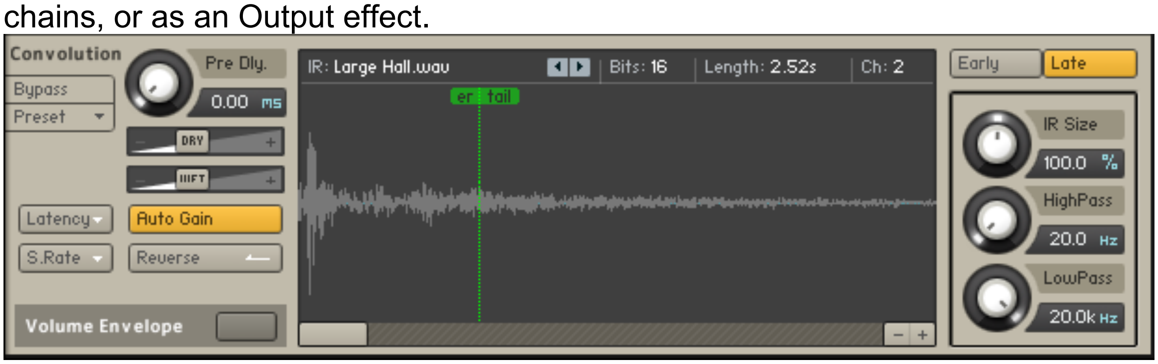
Convolution

For tips and tricks involving the convolution engine, please visit http://www.sonicreality.com/DrumMasters2

Convolution is a sophisticated mathematical process that, technically speaking, allows you to replicate the acoustical behavior of a linear system – such as a room, a speaker or a hardware reverb unit – for use with your own signals. To accomplish this, a short audio recording of a wide-band signal played through the system is fed into the convolution processor. This recording is usually a normal audio file, called an impulse response.

Convolution is best known among users as a method for achieving highly realistic reverbs; it works just as well for simulating the characteristic resonances of speaker cabinets and other loudspeakers, though.

KONTAKT's Convolution processor is somewhat unique in that it fully supports a multi-channel signal flow, allowing you to use surround impulse responses. It can be used within the Instrument Insert Effects and the Instrument Send Effects chains, or as an Output effect.



KONTAKT includes an extensive library of impulse responses, which range from recordings of real rooms and speaker cabinets to synthetic impulse responses that are well-suited for special effects, but you can just as easily use third-party impulses in WAV format.

Impulse Window: This window provides a display of the currently loaded impulse response and, if active, the Volume Envelope. You can drag impulse responses from the library into this window to load them – this will keep your other settings intact. The row at the top displays the filename of the loaded impulse response, as well as its bit depth, sample rate, and number of channels.

Just like in the Mapping Editor and Wave Editor, hovering your mouse pointer over the filename will show the full path to the file.

Preset Menu: In addition to the usual list of available presets, this drop-down menu offers an Open IR File command, which allows you to load a sample in WAV or AIFF format to be used as an impulse response.

Pre-Dly.: Just like the Reverb module's parameter of the same name, this control introduces a short amount of delay between the direct signal and the convolution output. This is useful when used with reverb responses to simulate the reverberation of big rooms, where a short delay occurs between the direct sound and the first reflections from distant walls.

Return (visible when used as a send effect): Adjusts the module's return level.

Dry and Wet sliders (visible when used as an Instrument insert effect): Adjusts the respective levels of the original and processed signals. In common scenarios, the reverb signal is mixed in at a lower level than the direct signal.

Latency: Adjusts the module's latency setting in five steps (1.5, 2.9, 5.8, 11.6, and 23.2 ms). If you hear crackles or other artifacts, you can try to increase this value, which will result in KONTAKT's overall latency being increased (and thus all signals being delayed). If you don't want this, you can disable the latency compensation by activating the last entry in this menu. This way, the overall latency won't be increased, but the wet signal of the Convolution processor will be delayed in relation to the dry signal (which actually can work just fine with reverbs).

Sample Rate: Allows you to divide the sample rate in nine steps (1/1, 1/1.5, 1/2, 1/2.5, 1/3, 1/4, 1/6, 1/8, and Auto). If the Preserve Length button is not enabled, changing the sample merely changes the playback speed of the impulse response, resulting in longer reverb trails and changed frequency characteristics. Activating Preserve Length will keep the reverb trail intact, but lower the sampling rate at which the convolution process takes place, thereby reducing the CPU usage along with the quality.

Reverse Button: Reverses the impulse response for special effects.

Auto Gain: If this button is active, the processor will keep the overall level constant when parameters are adjusted that would otherwise affect the level. If you turn this off, be sure to monitor at low levels as you make adjustments, as volume changes can be drastic – take care of your ears!

Volume Envelope: This feature allows you to change the volume characteristic of the impulse response to you needs. When activated, a graphically editable 8-

segment envelope will be drawn on top of the waveform display in the impulse response window.

Early / Late: These mode buttons switch the 3 knobs below between affecting the early reflections and the trail part of the impulse response.

IR Size: Artificially compresses or stretches the impulse response in time.

HighPass: Adjusts the cutoff frequency below which the signal’s frequency content will be attenuated.

LowPass: Adjusts the cutoff frequency above which the signal’s frequency content will be attenuated.

Gainer

This module can be used both within the Instrument Insert Effects chain and as a send effect. Depending on where you choose to place it, it serves two different purposes.As an Instrument insert effect, you can boost or attenuate the signal level between the previous stage’s output and the next stage’s input. In other words, it works like an additional amplifier stage.

The idea behind using the Gainer as a send effect needs some more explanation. As every send effect can optionally route its own output signal to one of the Aux Channels in the Outputs Section (instead of feeding it back into the Instrument), the Gainer can act as a transparent bridge between the send slots and the Aux Channels. Therefore, it allows you to send signals at adjustable levels to the Aux Channels on a per-Group basis, which greatly increases routing flexibility and can even save a lot of CPU resources – just move effects that you would otherwise have created as send effects in multiple Instruments to an Aux Channel instead, and use the Gainer within the Instruments‘ send effects slots in order to forward signals to this Aux channel. By changing the Aux channel‘s physical output assignment, you can even use external effects, whether they are plug-ins in your host program or outboard devices, from within Groups.

After you have inserted the Gainer module into a send slot, you‘ll notice that in addition to the Gain control, it also provides the Return control which is common to all send effects on the right side of its panel. Next to the numerical readout of this control, you‘ll notice a small „I“ icon. Clicking on it opens a drop-down menu that allows you to select one of the Aux Channels as the routing target for the module‘s output. This way, whatever you‘ll send to this slot via the Send Levels module will end up on the specified Aux Channel.

ControlsGain: The amplification or attenuation factor that will be applied to the signal in dB.

Filters

A filter is a signal processor which changes the frequency content of a signal that passes through it. This means that in contrast to effects like distortion, reverb, or chorus, it only changes the amplitude and phase of frequency components which are already present in your signal, without creating new frequency content.

Traditional filter designs generally exhibit one of four characteristics:

- **Lowpass filters** attenuate all frequencies above their cutoff point, leaving frequencies below unaffected.

- **Highpass filters** attenuate all frequencies below their cutoff point, leaving frequencies above unaffected.

- **Bandpass filters** attenuate all frequencies below and above a frequency range, which itself is being left unaffected.

- **Band reject filters** attenuate all frequencies within a range. Frequencies outside this range are being left unaffected.Filters of these characteristics are further distinguished by the steepness of their attenuation curve, which is usually expressed in dB per octave. As the cutoff frequency is defined as the frequency where an attenuation of 3 dB occurs, a filter slope of 12 dB/octave with a cutoff frequency of 440 Hz attenuates frequency content at 880 Hz (an octave above the cutoff frequency) by 15 dB, frequencies at 1760 Hz by 27 dB, and so on. In digital filter design, specifying a filter slope as a number of "poles" has become the norm, with each pole corresponding to 6 dB of attenuation per octave, such that a 1-pole filter will exhibit a gentle slope of -6 dB/octave, while a 6-pole filter with its slope of -36 dB/octave is more akin to a sonic razor blade. The pole notation is being used for KONTAKT's collection of sampler filters. If you find the concept of filter slopes confusing, don't worry – just remember that lower-order filters (like 1-pole and 2-pole forms) are generally better suited for gentle, unnoticeable tonal corrections, while higher-order filters (like 4-pole and 6-pole forms) tend to change your signal characteristics by a significant amount, and are thus better suited for broad treatments or effects.

KONTAKT's collection of filter modules is divided into 4 categories:

- Sampler Filters have no discernible sonic fingerprint, and are thus well-suited for neutral tone shaping.

- Synth Filters emulate the characteristics of classic synthesizer filters. They have a distinct character, often making them the first choice for synthetic sounds.

- Effect Filters are special filter modules that don't fit any of the traditional filter characteristics of either lowpass, highpass, bandpass or band rejection. An example of these would be vowel filters, which emulate the resonances of the human vocal tract.

- EQs are the kind of frequency tools that you can find on mixing consoles. They offer some ways of tonal alteration which aren't available with traditional filters, such as attenuating or boosting a specific frequency range by an adjustable amount.

The most convenient way to access KONTAKT's filter collection is by browsing the Filters page on the Modules tab of the Browser. Here you'll find a list of all available filters in each of the four categories, with icons depicting their frequency response, and descriptions of what they do and how you can use them. If you have found a filter that you'd like to try, just drag it into one of your Instrument's signal processor slots.

In the following subsections, we'll briefly introduce the available filter modules in each category and describe their parameters.

Sampler Filters

This category contains utility filters which allow you to change the frequency content of your signals in a variety of highly adjustable ways, without imposing any prominent "signature sound" on them.

Pole Lowpass

Cutoff: Adjusts the frequency above which signals will be attenuated with a filter slope of 6 dB/octave. This fairly gentle roll-off characteristic is typical of guitar tone controls.

1 Pole Highpass

Cutoff: Adjusts the frequency below which signals will be attenuated with a filter slope of 6 dB/octave.

2 Pole Bandpass

Cutoff: Adjusts the center frequency. Frequencies above and below this frequency will be attenuated with a filter slope of 12 dB/octave.

2 Pole Lowpass

Cutoff: Adjusts the frequency above which signals will be attenuated with a filter slope of 12 dB/octave. This roll-off characteristic is noticeably steeper than that of a 1-pole filter, but it’s still well-suited for subtle changes.

Reso. (Resonance): With a value greater than 0, this control will boost a small frequency range around the cutoff frequency. This emphasis is often associated with synthesizer sounds, especially when combined with filter modulation.

2 Pole Highpass

Cutoff: Adjusts the frequency below which signals will be attenuated with a filter slope of 12 dB/octave.

Reso. (Resonance): With a value greater than 0, this control will boost a small frequency range around the cutoff frequency.

4 Pole Lowpass

Cutoff: Adjusts the frequency above which signals will be attenuated with a filter slope of 24 dB/octave. This rather steep roll-off characteristic has been used in countless synthesizer filter designs, both in vintage and modern systems.
Reso. (Resonance): With a value greater than 0, this control will boost a small frequency range around the cutoff frequency.

4 Pole Highpass

Cutoff: Adjusts the frequency below which signals will be attenuated with a filter slope of 24 dB/octave.

Reso. (Resonance): With a value greater than 0, this control will boost a small frequency range around the cutoff frequency.

4 Pole Bandpass

Cutoff: Adjusts the center frequency. Frequencies above and below this frequency will be attenuated with a filter slope of 24 dB/octave.

Reso. (Resonance): With a value greater than 0, the narrow frequency range around the center frequency will be boosted, while at the same time, the slope at which the filter attenuates frequencies to both sides of the center frequency will become steeper, thus narrowing the frequency range of the output signal.

4 Pole BR (Band Reject)

Cutoff: Adjusts the center frequency of the “notch” at which frequency content will be attenuated.

Reso. (Resonance): With values greater than 0, this control will boost frequencies around the center frequency.

6 Pole Lowpass

Cutoff: Adjusts the frequency above which signals will be attenuated with a filter slope of 36 dB/octave. This is a unusually steep filter cutoff that's best suited for special effects.

Reso. (Resonance): With a value greater than 0, this control will boost a small frequency range around the cutoff frequency.

Synth Filters

These filters all exhibit a discernible character; their sound is similar to that of filters found in various vintage synthesizers, so they're a good choice for synthetic sounds.

PRO-53

This is the same filter section that is provided by Native Instruments' PRO-53 software synthesizer. It's similar in nature to the 4-pole lowpass filter, but has a different and more distinctive signature sound.

Cutoff: Adjusts the frequency above which signals will be attenuated with a filter slope of 24 dB/octave.

Reso. (Resonance): With a value greater than 0, this control will boost a small frequency range around the cutoff frequency. The resulting sound is often associated with synthesizer sounds, especially if combined with filter modulation.

4-Stage Ladder

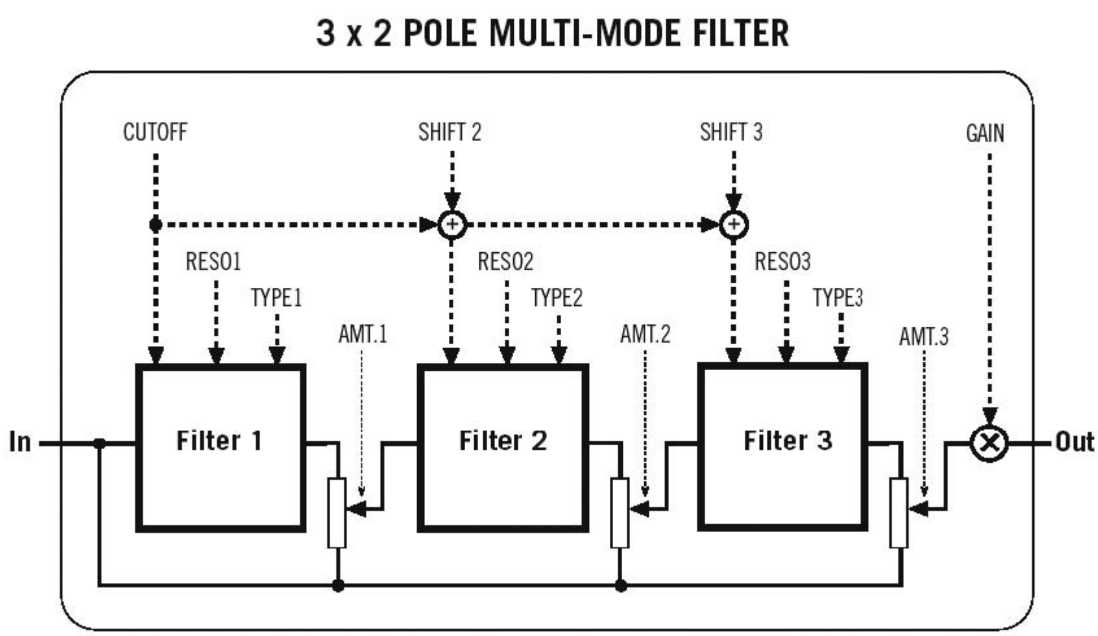
This is another 4-Pole filter type, but based on a filter structure that exhibits different resonance characteristics. Increasing the resonance parameter adds a very pronounced peak, and frequencies below the cutoff point will be attenuated at higher settings.

Cutoff: Adjusts the frequency above which signals will be attenuated with a filter slope of 24 dB/octave.

Reso. (Resonance): With a value greater than 0, this control will boost a small frequency range around the cutoff frequency.

3x2 Multimode Filter

The 3x2 Multimode Filter provides three separate filter bands, each of which can be continuously "morphed" between 3 characteristics (lowpass, bandpass, or highpass). Each filter band has a slope of 12 dB/octave. By combining these bands at various amounts, you can create almost every imaginable filter configuration. What's more, the resonance controls of each filter band exhibit a behavior known from very high-quality analog filters: at high settings, the filter will begin to oscillate and produce sound, even if there's no signal present at the input. This effect is known as self-oscillation.



The basic internal signal flow structure of the 3x2 Pole Multi-Mode Filter.

The 3x2 Multimode FIter requires more CPU power than other KONTAKT filters, so use it only when you need this level of sophistication, or a self-oscillating filter.

Cutoff: Adjusts the cutoff frequencies of the 3 filter bands in unison. The displayed value is only absolute for the first (topmost) filter band, the other two bands have cutoff frequencies relative to this one (see below).

Shift 2: Adjusts the second filter band's cutoff frequency as an offset in relation to the first filter. With a value of 0, both filters will have identical cutoff frequencies, while increasing the value will set the second cutoff frequency higher than the first.

Shift 3: Adjusts the third filter band's cutoff frequency as an offset in relation to the second filter.

Reso. 1 to Reso. 3 (Resonance): Adjusts the resonance (boost at the cutoff frequency) for each filter band. Values of 98% or higher will result in self-oscillation.

Type 1 to Type 3: Adjusts the characteristic of each filter band, allowing you to morph continuously between a lowpass (0.0), bandpass (0.5), or highpass (1.0) characteristic.

Amt. 1 to Amt. 3 (Amount): Adjusts the amount to which each filter band will affect the overall result. At a value of 0, the respective filter will be inactive.

Gain: As high resonance settings can significantly increase the signal level, KONTAKT will automatically reduce the level the output level in such cases. You can compensate this with the Gain control, but be careful – it's easy to get excessive volume levels from this filter.

Effect Filters

Filters in this category don't match any of the traditional filter characteristics, and are thus better suited for special effects.

Phaser

This module creates a distinct comb filter effect by using an allpass filter design that radically alters the phase relations in your signal. Note that there's also a Phaser module in the standard effects section, which has a built-in modulation mechanism. While that module and the phaser filter share the underlying principle, the filter is better suited for timbral changes, while the Phaser effect module is recommended for creating the classic effect of the same name that can be found in countless effects processors and guitar stomp boxes.

Cutoff: Adjusts the center working frequency of the phaser's comb filter effect. Changing this parameter will alter the tonality of your sound in a distinct and not always easily predictable way.

Reso: Adjusts depth and narrowness of the notches that the phaser imposes on the frequency spectrum, and thereby the intensity of the effect.

Vowel A

This filter simulates the resonant frequencies of the human vocal tract; when forming a vowel, the throat and mouth cavities will change their shape in order to create a complex, natural filter which emphasizes certain frequencies in the sound created by the vocal chords. These characteristic frequencies, called formants, allow the human hearing to discern between different vowels, and are being replicated by this filter.

Cutoff: Adjusts the center frequency of the filter. Various distinct frequencies across the spectrum will produce different vowels.

Reso. (Resonance): With values greater than 0, this control will emphasize the frequencies around the center frequency in order to create a sharper sound and enhance the effect.

Vowel B

This filter works similarly to the Vowel A module, but has a slightly different sonic character.

EQs

KONTAKT’s fully parametric peak equalizers allow for a wide range of tonal alterations and corrections. Using up to three EQ bands per module, you can boost or cut any frequency range throughout the entire spectrum by up to 18 dB, with an adjustable Bandwidth parameter allowing you to choose between gentle corrections or very steep “surgical” edits.

The EQ module is available in 1-band, 2-band, and 3-band flavors. You can switch freely between these configurations without losing your settings. All bands offer 3 identical controls, namely:

Freq. (Frequency): Adjusts the center frequency at which the boost or cut will occur.

Bandw. (Bandwidth): Adjusts the width of the frequency range that will be affected in octaves.

Gain: Adjusts the amount of boost (positive values) or cut (negative values) at the center frequency.

Support

For more information on Drum Masters 2 or Infinite Player compatible products please visit www.SonicReality.com.

For additional technical information regarding Sonic Reality or your Infinite Player please contact Support@sonicreality.com.

For additional Infinite Player compatible products please visit www.Downloadablesoundz.com or your local music software retailer.

For tips, tricks and patch updates please visit
http://www.SonicReality.com/Support

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