

Harrison AVA Drum Flow

AAX/VST/VST3/AU

The screenshot displays the Harrison AVA Drum Flow software interface, which is divided into several functional sections:

- EXPANDER / GATE:** Features a GATE/EXPANDER toggle, Hold (218 ms), Ratio (1:1 to 6:1), Hysteresis (4.0 dB), Knee (3.0), Depth (23.0 dB), Attack (0.5 ms), and Release (200 ms) controls. Includes a SIDECHAIN FILTER with a 1.05 kHz frequency and 1.00 Q.
- DRUM CHARACTER:** Contains two frequency response graphs. The left graph shows a peak at 146 Hz and a dip at 2.30 kHz. The right graph shows a dip at 502 Hz and a peak at 1.17 kHz. Includes an ATTACK section with 29.3 ms, AUTO SOLO, and TAIL with 205.0 ms. A bottom graph shows a frequency spectrum from 50 to 500 Hz.
- SIGNAL GENERATOR:** Offers White Noise, KICK, and SNARE presets. Controls include Sine Freq (50 Hz), Gate Hold (30 ms), Gate Attack (1.5 ms), Gate Release (500 ms), and FILTERS IN (HP Freq, LP Freq). Includes a SIGNAL LISTEN button.
- INPUT:** Features a gain knob and a Solo/Listen Clear button.
- COMPRESSOR:** Includes Ratio (3.0:1), Attack (0.5 ms), Release (60 ms), and Makeup (8.4 dB) controls.
- FILTERS:** Includes LPF (12dB/oct, 20.00 kHz) and HPF (12dB/oct, 192 Hz) settings.
- 32C CHANNEL EQ:** Features a 3-band EQ with LOW (310 Hz), MID (605 Hz), and HI (7.28 kHz) frequency sliders, each with a BELL filter and gain control.
- ROUTING:** A signal flow path showing FILTERS, EXPANDER / GATE, COMPRESSOR, DRUM CHARACTER, SIGNAL GENERATOR, and EQUALIZER.
- OUTPUT:** Includes an Output Trim knob.

Introduction

AVA Drum Flow is a channel strip plugin designed for mixing drums. It contains 6 routable processing elements and a master section with input, output, and routing controls.

The elements included are:

- **Drum Character:** A unique equalizer that detects drum hit transients and allows for the attack and tail sections of the envelope to be EQed individually.
- **Expander/Gate:** A versatile dynamics processor that can be used as either an expander or gate to tighten your drum sounds and remove bleed from other nearby drum mics
- **Signal Generator:** Enhance your drum sounds with extra 'snap' or 'thump' by blending-in a gated noise or low-frequency tone.
- **Compressor:** Harrison's full-featured console compressor with Ratio, Attack, Release, and Makeup controls
- **32C Channel EQ:** An emulation of Harrison's renowned 32C channel strip EQ.
- **Filters:** High- and low- pass filters with selectable shapes.

Plugin Specifications

Supported formats: AAX, VST, VST3, AU

Supported channel configurations: Mono, Stereo

NOTE: *This plugin uses lookahead and should not be used for processing signals in real time.*

Module Controls

The processing elements are separated into modules which can each be individually bypassed, soloed, and reset using the following controls on each module's top bar.



The **Reset** button resets all of a module's parameters to their default states.

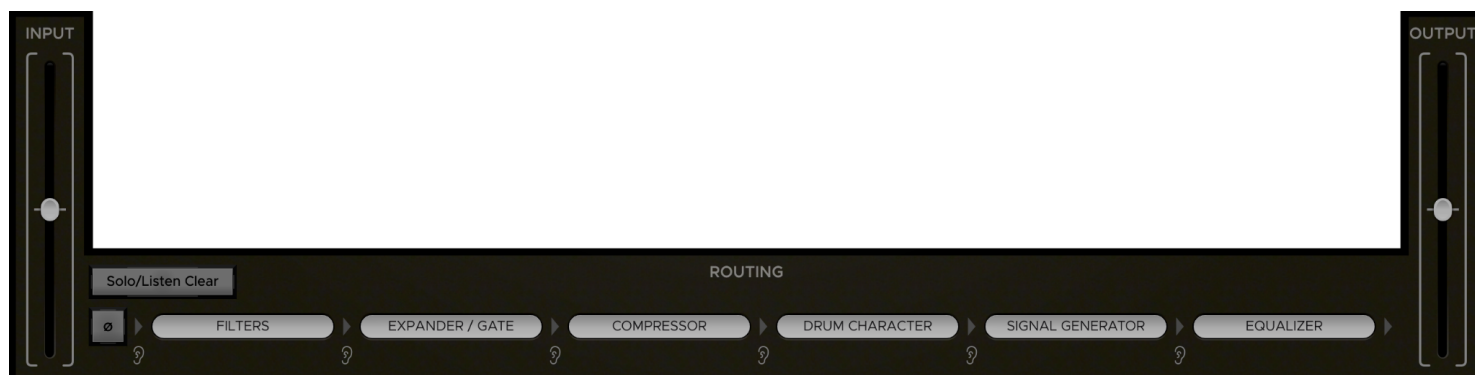


The **Solo** button bypasses all processing outside of the module being soloed except for master controls (polarity, input and output trim). This allows individual modules to be auditioned exclusively without hearing the effects of any of the other modules.



The **In** button toggles the enablement of each module. If disabled, a module's processing is bypassed and its effects will not be heard.

Master Controls (Input, Output & Routing)



The master controls consist of the Input, Output and Routing sections at the bottom of the plugin editor. The controls in these sections allow you to invert the input polarity, adjust the input & output trim, and set the routing order between processing elements.

The **Routing** section displays the order of signal flow between the elements from left to right. Each processing element is represented by a **routing item** which you can click and drag around to change the order of signal flow. Dragging a routing item to any of the arrows between positions will insert it at that position, shifting all other elements accordingly. If you drag one routing item over another routing item, it will swap the positions of the two items.

The “ear” icon buttons below the routing arrows indicate **Router Listen** points. These buttons allow you to listen at specific positions along the signal chain by automatically bypassing all modules past the selected point in the routing order.

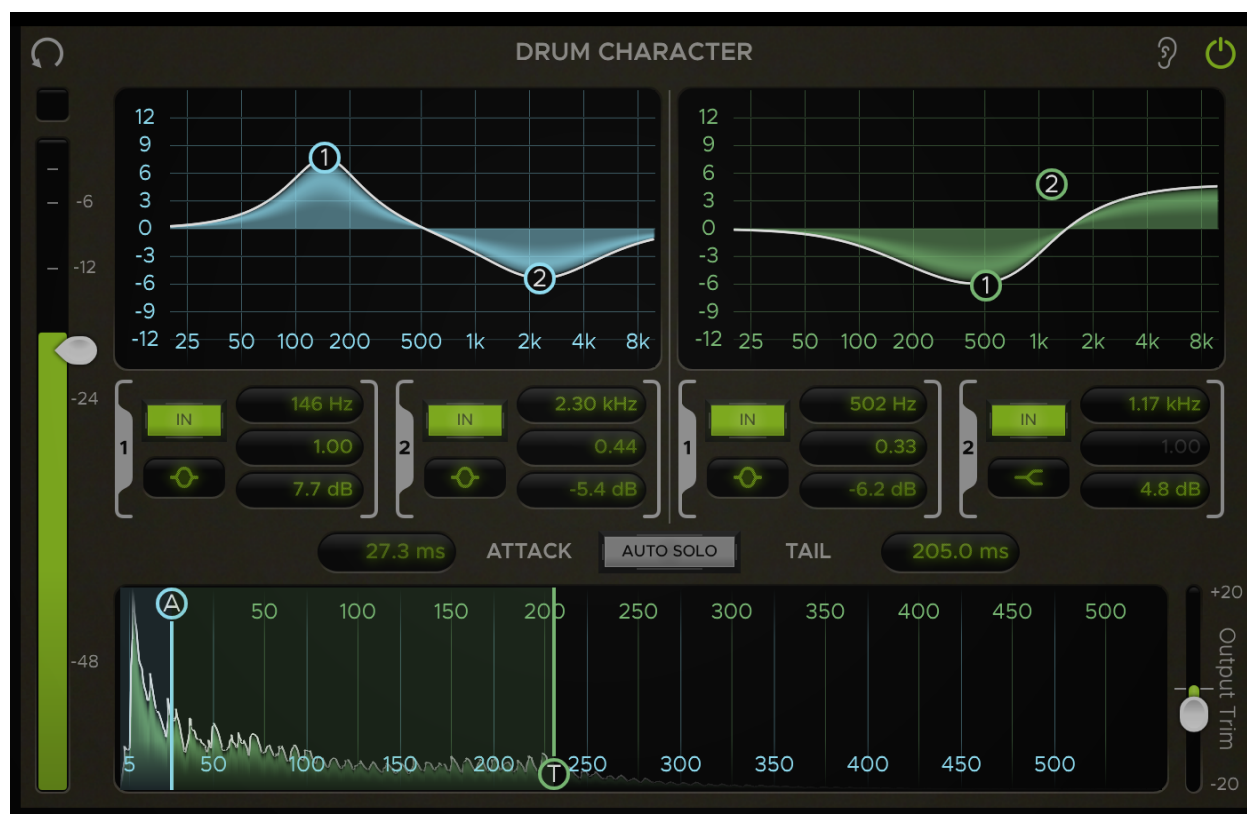
The **Polarity** button in the far left of the Routing section inverts the phase of the input signal when toggled on.

The **Solo/Listen Clear** button at the top-left corner clears any active Routing Listen point as well as any modules that are soloed. It will light up to indicate whenever a module solo or Router Listen is active.

The **Input** and **Output** sections to the left and right of the routing section contain the master Input and Output Trim controls which can be used to control the gain staging as necessary. The **Input Trim** adjusts the gain of the input signal before it gets processed by anything else while the **Output Trim** adjusts the gain of the output signal at the very end of the signal chain.

Processing Elements

Drum Character



The Drum Character is an EQ that detects the transients of drum hits and allows you to EQ their Attack and Tail ranges separately.

The **Threshold** slider attached to the input meter on the left sets the level at which the Drum Character EQ gets triggered. Whenever the input signal exceeds the Threshold, the Trigger Indicator light above the input meter will flash indicating that a new drum hit transient has been detected. Set the Threshold so that it only triggers on the desired drum hits.

Waveform Display

When a new drum hit is detected, its transient waveform will be shown on the **Waveform Display** at the bottom. The Waveform Display has controls for setting the **Attack** and **Tail** ranges of the transient. The **Attack Width** parameter sets the point in milliseconds in which the Attack range ends and the Tail range begins. The **Tail Width** parameter sets the amount of milliseconds in which the Tail range extends past the Attack Width. The Attack and Tail Widths can be adjusted on the Waveform Display using the sliders labeled "A" and "T" respectively or with the corresponding control labels above the display.

EQ Displays

Adjust the EQ settings for the Attack and Tail ranges using the EQ displays at the top of the module. The display on the left is the **Attack EQ** which is only applied to the section of audio within the Attack range. The display on the right is the **Tail EQ** which is only applied to the section of audio within the Tail range. The Attack and Tail EQs each have 2 bands which are labeled 1 and 2. The bands can be adjusted by clicking and dragging them on the graphs or by using the corresponding controls below the graphs.

The Attack and Tail EQ graphs provide a visual display of the EQ curves and allow for direct “hands-on” control of the bands. Each band is represented by a “bubble” on the graph which can be used to adjust its parameters. Click and drag a band’s control bubble to adjust its **Frequency** and **Gain**. You can also adjust the **Q** by right-clicking & dragging or by scrolling the mouse wheel over a band. Alt-clicking a band will reset it to its default state, while *double*-clicking will reset its Gain only. Special control modifiers such as these allow you to do a lot more with the band controls. For a complete list of control modifiers and shortcuts, see the [Control Behaviors & Shortcuts](#) section of the manual.

EQ Controls

The EQ control sections below the graph allow for more advanced control over the bands’ parameters. Each of the 4 band control sections contains controls for the corresponding band’s Enablement, Shape, Frequency, Q, and Gain. You can enable or disable a band by toggling the **IN** button at the top-left of its control section.

Directly below the IN button is the band **Shape** selector. Clicking this will open a menu where you can select the EQ shape for the corresponding band.

Each band has the following shape options:

- Low Shelf (12 dB/Octave)
- High Shelf (12 dB/Octave)
- Bell
- Notch (infinite “cut” at the selected frequency, with adjustable Q width)

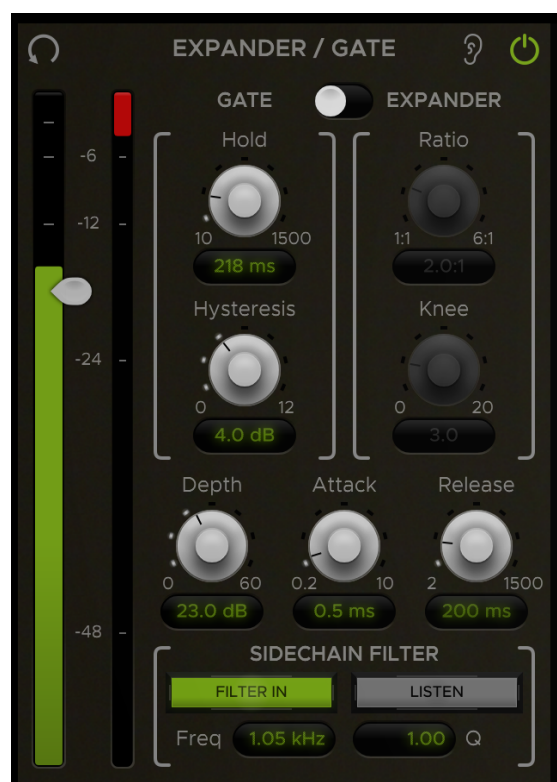
The control sections also provide **Frequency**, **Q**, and **Gain** controls which can be adjusted by clicking and dragging their value labels or by typing new values directly into them. For ease of use, these will accept bare numbers such as “500” or numbers with units such as “500Hz”, “1k”, “1kHz”, “20dB”, etc.

Auto/Quick Solo

The **Auto Solo** function allows you to isolate the individual ranges while you adjust their controls. When the Auto Solo button is toggled on, adjusting the Attack EQ bands or the Attack Width will automatically solo the Attack portion of the transient and adjusting the Tail EQ bands or the Tail Width will automatically solo the Tail portion. The solo will only remain active for as long as the control is being adjusted.

Another way to solo the Attack or Tail range is to use the **Quick Solo** modifier key. When adjusting any of the EQ or Width controls, you can hold down the **Control** key on your keyboard to temporarily solo the corresponding range until the key is let go or until the control is no longer being adjusted.

Expander/Gate



The Expander/Gate is a dynamics processor that attenuates signals below a set threshold. This is useful for removing unwanted parts of a signal that were picked up due to microphone bleed.

The slider attached to the **Input Meter** on the left controls the **Threshold**. The Threshold sets the dB level in which the Expander/Gate opens. Signals above the threshold will be allowed to pass through unaffected while signals below the Threshold will be attenuated.

Directly to the right of the Input Meter is the **Reduction Meter** which displays the level of gain reduction in dB.

The **Depth** sets the amount of gain reduction that is applied when the Expander/Gate is fully closed.

The **Attack** sets the amount of milliseconds it takes for the Expander/Gate to open fully once the Threshold has been exceeded.

The **Release** sets the amount of milliseconds it takes for the Expander/Gate to close all the way once the input signal falls below the Threshold.

The **Mode** switch at the top toggles between **Gate** and **Expander** mode. The availability of certain controls depends on the selected mode. The **Hold** and **Hysteresis** are only available in Gate mode and the **Ratio** and **Knee** are only available in Expander mode.

Gate Controls

The **Hold** sets the amount of milliseconds the gate will stay open for once the input signal has dropped below the Threshold.

The **Hysteresis** sets the amount of dB below the Threshold in which the Gate will close. This allows the signal to fall below the Threshold by a certain amount of dB before the Gate starts closing. For example, if the Threshold is set to -12dB and the Hysteresis is at 4dB, the Gate will open at -12dB and close at -16dB (4dB below the Threshold).

Expander Controls

The **Ratio** determines the amount of gain reduction applied for every decibel the signal falls below the Threshold. For example, a ratio of 2:1 means that an input signal that is 1dB below the Threshold will be reduced so that the output is 2dB below the Threshold.

The **Knee** controls the “smoothness” of the Ratio angle. With a *low* Knee value (Hard Knee), the Expander will react more abruptly as the signal crosses the Threshold. With a *high* Knee value (Soft Knee), the Expander will react more gradually.

Sidechain Filter

The Expander/Gate has a **Sidechain Filter** that can be used to filter the key signal that triggers the Expander or Gate. It is a band-pass filter that attenuates all frequencies outside of the target frequency range so that the Expander/Gate will be more sensitive to signals within that range. The Sidechain Filter controls can be found in the Sidechain Filter section at the bottom. The **FILTER IN** button toggles the filter in or out. When the Filter is toggled in, the other Filter controls will become enabled. The **Freq** control sets the center frequency of the Filter and the **Q** control sets the width of the band-pass range around it. The **LISTEN** button allows you to hear the filtered sidechain signal when toggled on.

Signal Generator



The Signal Generator offers a convenient way to enhance drum hits with various types of synthesized signals, such as white noise on a snare or a low-frequency sine wave on a kick. The generated signal is fed through a gate which is keyed off of the main input allowing the signal to be triggered on drum hits. The generated signal can then be blended with the main input signal in parallel using the **Wet** and **Dry** faders.

Signal Type

Choose the type of signal you want to generate using the **Signal Type** menu at the top. You can select between **Sine Wave**, **White Noise**, **Int. Input**, or **Ext. Input**.

In **Sine Wave** mode, the **Sine Freq** slider will become enabled allowing you to set the frequency of the sine wave.

In **Int. Input** (Internal Input) mode, instead of generating a signal, it sends a duplicate of the direct input through the gate to allow for parallel gating.

Ext. Input (External Input) mode takes whatever signal is being sent to the plugin's external key (sidechain) input and uses it as the generated signal. This allows you to use signals generated from external sources outside the plugin.

Gate Controls

The Signal Generator has Threshold, Hold, Attack and Release controls for setting the signal gate.

The slider attached to the **Input Meter** on the left controls the gate **Threshold**. Use this to set the level in which you want the gate to open for the generated signal to pass through. The Trigger Indicator light above the input meter will light up whenever the gate opens.

The **Gate Hold** sets the amount of milliseconds the gate will stay open for once the input signal has dropped below the Threshold.

The **Gate Attack** sets the amount of milliseconds it takes for the gate to open fully once the Threshold has been reached.

The **Gate Release** sets the amount of milliseconds it takes for the gate to close all the way once the input signal is no longer above the Threshold.

Filters

The Signal Generator also has a pair of high and low pass Filters which can be used to filter the generated signal as needed and are especially useful for shaping white noise. The Filter controls can be found in the Filter section at the bottom of the module. The **FILTERS IN** button toggles the Filters in or out. Note that the Filters will be disabled in Sine Wave mode.

Presets

The **Kick** and **Snare** preset buttons can be used to quickly dial-in settings based on the type of signal being processed.

Signal Listen

The **SIGNAL LISTEN** button at the bottom-right of the module allows you to listen to the generated signal only.

Wet & Dry Level

The **Wet** and **Dry** faders can be used to balance the levels of the Wet (generated) signal and the Dry (input) signal.

Compressor



The Compressor is based on the Harrison Mixbus channel strip compressor with the addition of Attack and Release controls.

The slider attached to the **Input Meter** displayed horizontally across the top controls the compressor **Threshold**. The Threshold sets the dB level in which gain reduction starts to occur.

Directly below the Input Meter is the **Reduction Meter** which shows the level of gain reduction in dB.

The **Ratio** determines the amount of gain reduction applied for every decibel the input signal exceeds the Threshold. For example, a ratio of 2:1 means that an input signal that's 2dB over the Threshold will result in an output that's only 1dB over the Threshold.

The **Attack** time is the amount of milliseconds it takes for the gain reduction to reach full attenuation once the input signal passes the Threshold.

The **Release** time is the amount of milliseconds it takes for the signal to return to its original, uncompressed state once the input signal falls below the Threshold.

The **Makeup** control is used to boost the gain of the output signal to compensate for any gain that was lost during compression.

32C Channel EQ



The 32C Channel EQ is modeled after the analog EQ found on the Harrison 32C channel strip. It has four bands with Frequency and Gain controls, optional high & low shelf filters, and an Output Trim control.

The four bands consist of a **HI** band, **LOW** band, and two **MID** bands. Each band section has **Frequency** controls on top and **Gain** controls on the bottom. You can adjust these parameters with the knobs or by typing values directly into the corresponding control labels.

By default, the HI and LOW bands are *shelf* filters and the MID bands are *bell* filters with a proportional Q. The HI and LOW bands can be toggled between **Bell** and **Shelf** mode using the corresponding **BELL** buttons in each section. The bands are in Bell mode when the button is toggled on and Shelf mode when it's toggled off.

NOTE: *The proportional-Q design was used in the original Harrison consoles to allow an engineer to use the EQ as either a subtle tone shaping tool, or a steep filter problem-solver, with only 2 knobs. As the gain is progressively increased (or cut), the bandwidth becomes progressively narrower (the Q increases). A typical technique is to increase the gain to maximum, sweep the frequency to find an offending sound, and then turn the gain to minimum to “notch” out the unwanted frequencies.*

Filters



The Filter module contains a **Low-Pass Filter** (LPF) and a **High-Pass Filter** (HPF) with variable slopes. Set the corner frequencies of the Filters by adjusting their **Frequency** knobs or by typing values directly into the corresponding Frequency control labels.

The **Slope** selectors above the Frequency control labels allow you to select the Slope for each of the Filters. Clicking the Slope selector will open a menu where you can select from 5 different Slope settings: 6 dB/Octave, 12 dB/Octave, 24 dB/Octave, 36 dB/Octave, and 48 dB/Octave. You can also switch through the Slope options more quickly without opening the menu by scrolling the mouse wheel over the Slope selector.

Control Behaviors & Shortcuts

Various mouse & keyboard shortcuts can be used to perform certain actions on the following controls:

Band Controls

<u>Action</u>	<u>Shortcut</u>
Reset Gain to default	Double-click
Reset Frequency, Gain and Q to default	Alt-click
Adjust Q	Right-click & drag Shift-click & drag Scroll mouse wheel
Quick Solo (temporarily solos the corresponding range while the control is being adjusted)	Hold Control key while adjusting a band
Fine adjust Frequency and Gain	Command-click & drag (OSX Only)

Combo Boxes (Drop-down menus)

<u>Action</u>	<u>Shortcut</u>
Reset to default	Alt-click
Scroll through menu items without opening menu	Scroll mouse wheel
Select next/previous item (menu must already be open)	Keyboard arrows
Close menu (if open)	ESC

Labels

<u>Action</u>	<u>Shortcut</u>
Reset to default	Alt-click
Fine adjust	Shift-click & drag Control-click & drag (Windows only) Command-click & drag (OSX only)
Adjust value	Scroll mouse wheel
Close editor (discarding changes)	ESC
Close editor (submitting changes)	Enter
Quick Solo (Drum Character only)	Hold Control key while adjusting

Sliders/Knobs

<u>Action</u>	<u>Shortcut</u>
Reset to default	Double-click Alt-click
Fine adjust	Shift-click & drag Control-click & drag Command-click & drag (OSX only)
Adjust value	Scroll mouse wheel

Additional Features

At the top-right corner of the plugin editor are the **Momentary Bypass** and **Settings** buttons.

Momentary Bypass



The **Momentary Bypass** button bypasses the plugin for as long as it's held. This is useful for auditioning A/B comparisons quickly.

Settings



The **Settings** button reveals a pop-up menu where the window size and theme can be adjusted. It also contains a link to the plugin manual and version info at the bottom.

Installation & Licensing

The AVA Plugin “*Pain-Free Demo*”

When you install Harrison’s AVA plugins, you are provided a “*pain-free demo*” of unlimited length. You may continue to use the demo for as long as necessary to determine if the plugins meet your expectations.

We **do not**:

- Annoy you with a popup authorization message, every time you start your DAW.
- Install anything to your system outside of the plugin (AU, AAX, VST) folder.
- Prevent the plugin from saving or loading its state.

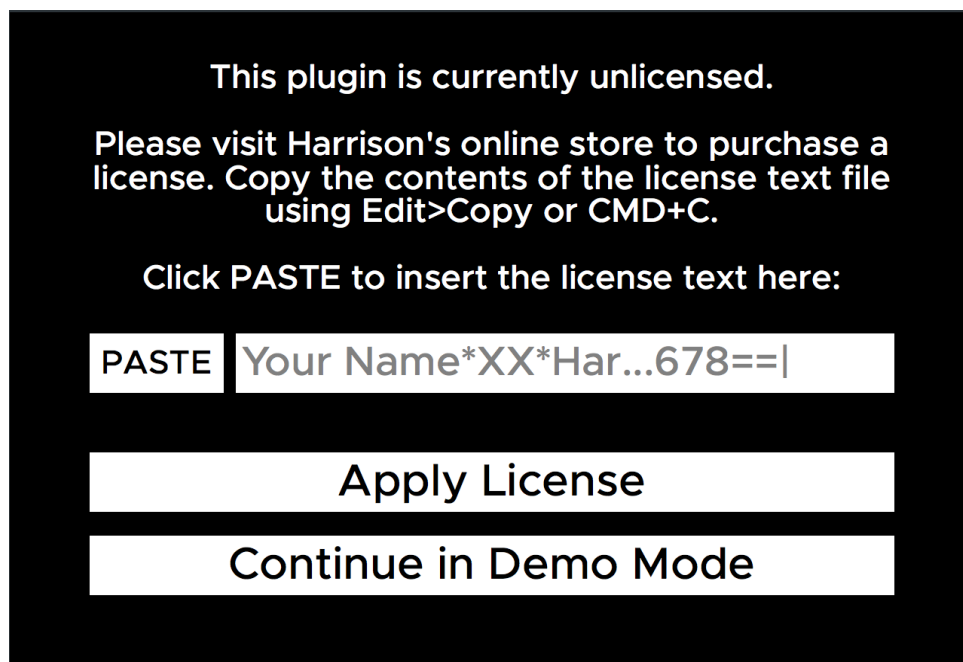
In demo mode, the control panel will become disabled after a short period of operation. You can re-initialize the plugin’s control panel by clicking “Continue in Demo Mode”. This is the only limitation. The plugin’s effect will continue normally, allowing you to evaluate the sound of the plugin on your session. You may still adjust the plugin’s controls in your DAW’s “Generic plugin UI”, from a control surface, or via automation.

If you send a session to a collaborator, they are not required to purchase an AVA license to hear the plugin’s effect. They may simply download the demo to install the plugins, and the plugins will operate normally. This allows you to share your sessions with other engineers, and have confidence that the mix will sound as intended. If they want to make changes to the plugin settings then they should buy a license for their personal use.

This product includes software developed by the OpenSSL Project for use in the OpenSSL Toolkit (<http://www.openssl.org/>)

Purchasing and Installing your Plugin License

In DEMO mode, after using the plugin for a certain amount of time, it will display a message:



This message will come up periodically unless a license is provided. You can purchase a license directly from Harrison's [web store](#) or from an authorized AVA plugin [reseller](#). Once you've purchased a license, install it by entering your **license key** into the designated field on this screen. Continue to the next section for detailed instructions on how to obtain your license key.

Obtaining your License Key

After you purchase a license, you will receive a **license key** that will look something like this:

{your name}*AVA-DF*DC-138971-151520-1|gGLbnvCZJe386ULohzfQN6clsoZ71n7c+AUg==|

This license key is required to install your license and activate the full version of the plugin. The way to obtain your license key varies depending on where you purchased the license from.

If you purchased the license from Harrison's [web store](#), you will receive an email with a link to download a text file containing your license key.

If you purchased the license from a reseller, you will receive an email containing the license key text.

Licensing your Plugin

When prompted by the plugin, copy and paste your license key into the designated text input field. This will automatically install your license and register it with the plugin.

Once installed, Harrison's license files are simple text (.txt) files which are stored in the user's HOME folder. The HOME folder is the folder that contains your Desktop, Downloads and other folders. The location of your HOME folder is different for each operating system:

- **Mac OSX:** /Users/{your username}
- **Windows:** C:/Users/{your username}
- **Linux:** /home/{your username}

You may keep a copy of this file for your own use. The license is a per-user license and you may install the license file on **all** of your computers. To install the file on a new computer, simply save the file to a USB key or other transferable disk; then transfer the license file to the new computer's HOME folder. The license file has your name encrypted in it, so you should not share it with other people. Your name will appear on the top of the plugin's control panel to indicate that it is licensed to you.

The plugin will automatically detect your license key as long as the license file is stored in the HOME folder of the computer you are using. After transferring the license file to a new computer, it may be necessary to close and re-open your DAW in order for the plugin to detect the license.

If you don't want to keep your license in your HOME folder for whatever reason, you may also store it one of these alternative locations:

- **Mac OS X:** ~/Library/Application Support/harrisonconsoles
- **Windows:** ~/AppData/Local/harrisonconsoles
- **Linux:** ~/.config/harrisonconsoles

Licensing from a USB Drive

The plugin can also detect a license if it is stored on a USB drive. This can be useful if you're using someone else's computer and you want to license an AVA plugin without installing your license onto the computer.

In order to do this, you must name your USB drive as “HARRISON” and place your license file in its root directory. This is the only way the plugin can detect your license on a USB drive.

For additional questions about the license, please contact our plugin support department via email: plugins@harrisonconsoles.com

Finding the AVA plugin in your DAW

DAW operations vary tremendously. Generally speaking, to use the AVA plugin you must follow these steps:

- 1) Install the plugins, using the provided plugin installer application.
- 2) Follow your DAW’s instructions to “scan” for new plugins.

Please refer to your DAW manual to see which plugin formats (AAX, AU, VST, or VST3) are supported by your workstation.

Most plugin formats require that plugins are installed to a specific location on the system. In these cases, the AVA installer will place the plugins in the default locations.

For **VST** format, on **Windows**, it is normally expected that the user defines the location of VST plugins, and tells the workstation where to find the plugins. The default location for VST plugins is *C:\Program Files\Steinberg\VST*. The AVA plugin installer will save the VST (.dll) files to that location if a custom VST path is not provided by the OS. If you are having trouble, you should verify that the path is included in your DAW VST scan paths.

System Requirements

Your system must meet the following requirements in order to run Harrison plugins:

- Windows 7 or higher
- Mac OS 10.9 or higher

- Linux GCC 5 compatible systems
- OpenGL 3.2 or higher
- libcurl-gnutls on Linux

Note for Pro Tools users: Pro Tools 11 or higher is required on Mac, 10.3.5 or higher is required on Windows.

Additional requirements on Windows

Harrison plugins require the “Windows C++ Universal Runtime” to be installed on your computer. Universal runtime is part of the system’s core libraries on Windows 10. On Windows versions Vista to 8.1, it will be automatically installed via Windows Update. It is also possible to install the runtime manually, but that is outside the scope of this manual.

Note for Linux Users

If you’re experiencing issues with the plugin’s graphics or the GUI isn’t showing up correctly on Linux, make sure your system’s graphics driver supports the minimum required OpenGL version of 3.2 or higher. To check this, run `glxinfo | grep Max` in a terminal window and make sure that the value listed for the `OpenGL core profile version string` is at least 3.2 or greater.

If your core profile version meets this requirement but you are still experiencing graphics issues like the ones mentioned above, it is likely that your system is reporting an OpenGL version lower than 3.2 to ensure backwards compatibility with deprecated OpenGL functions. To work around this, you can force the system to report an OpenGL version of 3.2 or higher by setting this environment variable: `MESA_GL_VERSION_OVERRIDE=3.2`

You can use this as a prefix when opening your DAW from a terminal command like so:

```
MESA_GL_VERSION_OVERRIDE=3.2 {command_to_launch_program}
```

This will set the environment variable for that particular terminal session only, so it will be reset once the session is ended. If you don’t want to run this command every time you launch your DAW, you can set the environment variable in your profile script so that it automatically gets set every time you login.

Thank you for purchasing this Harrison AVA plugin!

Harrison has been making products for the audio engineering industry since 1975. We continue to make these products, with intense dedication, from our Nashville, Tennessee factory.

As the audio industry has changed through the decades, Harrison has too. Originally launched to provide mixing consoles for music recording studios, the Harrison brand has extended to provide products for live broadcast, live sound, theater installations, post-production, film dubbing stages, and bedroom studios. We've lived through the launch of the commercial music industry, the maturation of analog, the digital revolution, the establishment of the internet, and now the acceptance of computers as the de-facto platform for the audio industry. In our pursuit of audio excellence, we've developed our own analog circuits, digital I/O solutions, digital DSP engines, control surfaces, audio workstations, and nearly every aspect of pro-audio technology.

"If you asked a hundred engineers, mixers and producers to define the "perfect console" you would probably get a hundred answers. The answers, however, would all contain the same salient points."

"The "perfect console" would have totally transparent sound, capturing the true musical quality of all performances. It would have unlimited features, facilities, and functions. It would perform all required tasks with a minimum of operator effort. It would be totally reliable and require no preventive or corrective maintenance...

You can spend the rest of your life looking for the "perfect console". You'll never find it. You will find, however, that we have what you need." - quote attributed to Dave Harrison; Harrison MR20 brochure; circa 1982

True-er words were never spoken; indeed every year there are many products and technologies that seem tantalizingly necessary for audio excellence. Our lasting experience has proven that simple, straightforward, well-engineered, and great-sounding tools are the only necessity.

The AVA plugins represent the pinnacle of Harrison's engineering practices for digital audio sound & workflow. We sincerely hope that you enjoy using these tools as much as we enjoyed creating them. Your continued support is greatly appreciated!