POD HD FOR DUMMIES

An introduction into tone building (Version 1.6 Final Version)
by DutchGuitarist

Content

Introduction

This article is based upon my experiences with the POD HD PRO in a recording environment. All of the information here is 100% applicable to the HD 500(X) and HD PRO(X). Note that I do not have the variax, I only use standard guitars.

This guide is not a complete tone building tutorial, I just want to provide new POD users with a basic insight in the tone building process based on how I do this myself.

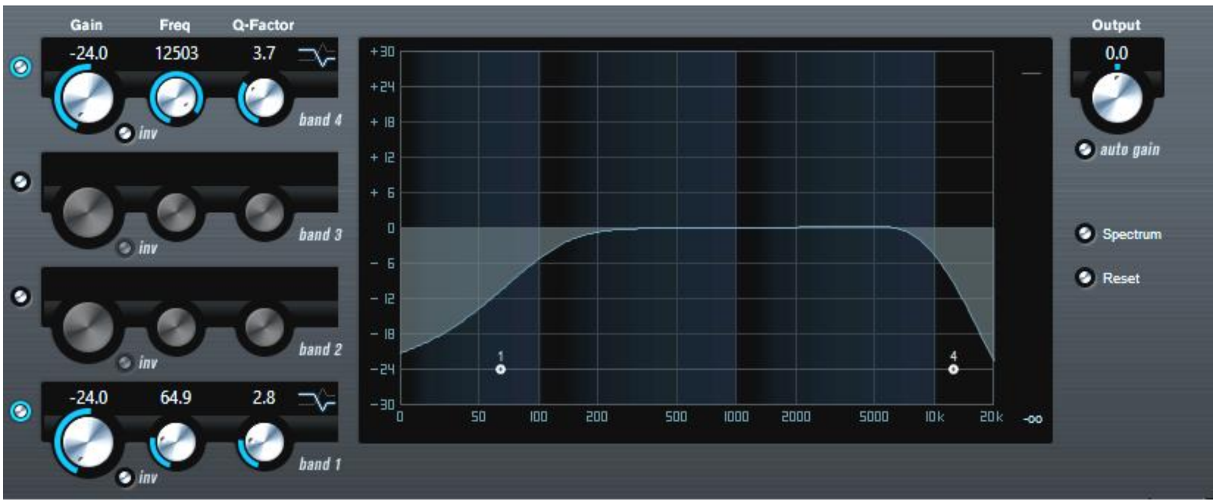
The learning curve

Understanding the POD HD is a quit complicated learning experience as the combinations of amp, cabs, mics and effects are almost endless. Sure there are examples on Line 6's Customtone, but I find these not very useful. First they are made by someone who may have a totally different idea about a good tone. Secondly these patches may not always sound right to you as they have been built with probably different input settings, guitars, pickups, monitors etc. Third and last issue: you don't learn anything about the tone building process. I must admit, when I bought the POD I also looked on Customtone, but at best I got an impression of how something could sound, nothing more. Building from scratch is a learning experience and the more patches you are building, the more you understand about the POD's behaviour and in the end you will be able to dial in your tones really fast.

Global EQ

The frequency range of an electric guitar goes from about 60Hz to about 1.2 KHz. From 1.2 KHz to 6 KHz there is the area where the harmonics live. Beyond 6 KHz there is nothing musical present. Given this it is a good idea to use Global EQ to cut all frequencies below 60 Hz and above 6 kHz. This way unwanted signals will not be recorded and it will reduce hiss and fizz. Do play around with the upper and lower settings, especially with the low end as this can help to reduce muddiness. Now you can do this individually for each patch with one of the available EQ's but to me it makes more sense to do this with global EQ (also saves an effect slot) and, if necessary, fine-tune the EQ curve after recording during the mixing stage.

Typically the Global EQ curve will look like this:



Input and output settings

As I use the POD HD for recording, the global output mode on the POD needs to be set to "studio direct". At the input side of the global settings we can choose the default input options. Here you can specify how the two input channels are fed. As a rule of thumb set input one to "guitar", because the built in tuner is tied to input one only. Set input two to either "same" or to "variax". You can also specify these parameters per preset in the HD edit program:

When you set input two to "variax" only input one is actually used. Remember that if you only use input one there is no signal on that second path (path B). So if you put an effect in this second pad it will not get a signal. To make it more complicated there are effects that are stereo preserving and other that "sum to mono". After such an effect the second path will get a signal after all, because a mono effect will split the incoming signal in two output signals (of half the strength). So if you don't use input 2 (set to variax) but start with a mono effect such as a noise gate both paths (A & B) will get a signal.

Later on I will explain the practical use of the input two.

Furthermore there is a parameter called "Z". This sets the input impedance of the guitar input. This affects tone and feel because your guitar pickups are being loaded as they would be by an effect pedal or a tube amplifier. I prefer to leave this on "auto", because I can change it in a patch when needed. In "auto" mode the input "Z" is set automatically based upon the first effect used. In the advanced manual there is a table where you can see how the setting is affected by effects.

All of these settings can be overwritten is you choose "preset" rather than "global". I prefer the per preset options because I want the flexibility during tone building.

The output settings are global settings and therefore you need to set these on the device itself. These settings have to do with how yu are going to use the POD HD.

When to connect to a mixer or recording device set the output mode to 'Studio/Direct'. If you connect to the front of an external amp, set it to 'combo/stack front' (you will get additional EQ options) If the destination is a poweramp choose combo/stack pwr amp).

If you are using the unbalanced outputs (1/4"cable) and send the signal to a guitar amp you need to set the physical switch to 'AMP' are you connecting a mixer or a recording device set it to 'LINE' (this is a typical --10dbV level)

DSP Limit reached

This is probably the most common error you will get when using the POD HD. It means that the signal chain you are working on exceeds the internal system's memory (DSP). This will happen earlier when you are using the non "X" versions of the POD HD PRO or 500). If you run into the DSP limit reached issue consider the following (depending on what you want to achieve):

Change the amp model. Some amps are more DSP heavy than others. For instance the ENGL Fireball (Angel-F) uses more DSP than the Mesa Dual Rectifier (Treadplate) and both are high gain amps. Other DSP heavy amps are the Gibson 185 (Gibtone), Fender Bassman (Tweed B-man) and ÷13 (Divided by 13). Look at reverb: Spring, '63 spring and Particle Verb are more DSP heavy than the other reverb models. Pitch shifters and filters such as Synthomatic, attack strings, synth string and growler are all DSP heavy.

Last but not least you might consider an external effects unit (in the FX loop)

Setting up patches

I use the ABCD mode on my POD. In "ABCD" Mode: The A**,** B**,** C & D footswitches are used to select the presets of the current bank. So in each bank I make four different kind of flavours for the same amp/cab combination. These four flavours are Clean(A), Crunch(B), Drive(C) and Lead(D). This is no official naming convention and anything will do as long as the naming makes sense to you. Now, when you switch presets within a bank there is, for a spit second no sound because the new patch has to be loaded. To me this is not really an issue but if it is, there is a solution I will discuss later in this document.

As for assigning effects: I always use the same footswitches(1-4) for the same functionality, this way I do not have to guess what footswitch to use for what effect. Footswitch 1 is always compression/distortion, Footswitch 2 is for modulation effects, 3 is for delay and 4 is for reverb.

As another rule of thumb I set the physical master volume knob for recording at around 50%. (This works for me and I never touch this knob again, so all patches are designed with this setting).

NOTE: The pedal mode setting are system settings and as such you need to access these settings directly on the device.

You can built your patches directly on the device itself or use the HD edit software (remember system setting can only be edited on the device itself). However do not mix these two options, use only one, as using both options at the same time is likely to cause synchronisation issues.

Using downloaded patches

Using downloaded patches (such as from Customtone) might not give the desired tone at all. As said in the chapter on the learning curve there are many parameters that might need adjusting. Another aspect to always look at are the input settings. Make sure these are set to your specific configuration otherwise you might not hear anything at all.

Another issue that might occur is that the downloaded patch is corrupted or not mend to be used with your version of the POD HD. SO: backup you patches before you start experimenting with downloaded patches.

Last but not least: The creator of the patch may have used different system settings, simply because he/she has a different physical configuration.

Stereo or mono?

You can use the POD HD in either stereo or mono mode (per preset). And as with most other topics in this manual there is no right or wrong. It you are using the POD in live circumstances through a PA you might want to use a mono setting because then you only need one channel of the PA system and in most cases the stereo effect will not be noticed by the audience any way.

In a recording environment I prefer stereo setting because you can hear the stereo effects much better and it will make the guitar track sound wider (provided you use stereo time based and/or modulation effects.

Given this, I have decided to built everything in stereo because I use the POD mostly in a studio environment and when I am in a live situation(with a PA) I use a DOD 606 (line mixer/distribution amp) to mix the stereo signal into mono if necessary.

Using the HD edit software

I use the software a lot, in fact I use editing on the device only to make minor adjustments, such as channel (patch)volume.

Although the software comes with a user manual there are some (undocumented) features that may come in handy.

As explained earlier I use the ABCD mode to select a presets in the current bank. As the four presets, in my case, always use the same effects, amp and cabinet (only the mic's tend to be different, as well as the amp's drive). So rather than entering and editing these parameters for all four presets I use the drag & drop feature of HD edit. Once the first initial preset is ready I control+click this preset and drag it to the position of the second preset (say from preset A to B). Then I rename this second preset and I only need to adjust a few parameters such as amp drive, volume etc. There are two way to use drag & drop within a set list: with the control-key pressed you make a COPY of the dragged preset, without the control-key you MOVE the preset,

You can use this drag & drop feature also to copy preset from one set list to another. Also as there is no 'delete' function available you can use this feature to copy and empty preset to the preset you would like to erase.

Another thing (that is documented by the way) is to set a knob (any knob) to the desired value. The user interface more or less invites you to turn the software knobs to set the values, but there are two other ways that provide more accuracy(or at least save time). Position your mouse pointer above the knob and rotate the mouse wheel to raise or lower the value. You can also double-click the box that shows the value and just type the desired value, however this method requires the subject(effect/amp) to be active (switched on).

Personally I use the knob the get close to the desired value and use the mouse wheel for the final adjustments.

If you find yourself using a typical preset setup such as Gate-Compressor-Distortion-Amp-Delay-Reverb, you could consider to turn this setting (without the amp) into a template. When you are satisfied with a preset, remove the amp and save the preset to your PC hard disk. When building a new patch either import this template first or drag & drop it simply on the desired slot. In this way you may save valuable time.

Tone building essentials

To be honest there aren't any, it is up to you. From a logical point of view however it is likely you start with choosing the amp model, cab and a microphone. Important is to start with the clean patch and finish with the lead patch. If you would work the other way around you might find in the end that the clean is not loud enough and you will have to start tweaking again. Don't use headphones but use full range monitors instead. What good sounds with headphones will not necessarily sound good on monitors.

Another thing to understand is that a tone that sound good at bedroom level will definitely sound different on stage volume and vice versa. So if you are satisfied with your patches at bedroom level, copy these to a typical set list for live performances and take the time to adjust the setting to the stage environment. Typically you will need to rework the settings of reverb (less) and the amp's tone controls: less bass and treble and raise the mids.

When you building your tones check them also "in the mix". In other words see how the tone stands out with a (useful) backing track. A word on "presence": personally I hardly use it because in general I am happy with the treble control as it is, but if I do use it, it is just a touch (<20%). Presence seems to boost the upper mid-range frequencies and to my ears it introduces a somewhat nasal sound that I don't like. The one exception to this is with the Marshall amp models, for some reason they sound a bit better to me with about 50% presence.

A last word of tone building essentials: There is a lot to choose from and, by all means, check all of these options out, but the end it is likely you will only be using a few amp models and cabinets (at least I do)

Signal Routing: Understanding the mixer block

A, for some, confusion concept is that the POD's signal chain includes the possibility of serial and/or parallel signal chain (using paths A & B).

If we take a look at the mixer block in POD it looks like this:

It has two controls for both path A and B: level and panning.

In the next example we see a serial processing example: the signal chain goes thru the effects in a serial way (one after the other)

If we move the two effects after the mixer block to patch A and B we will get a parallel signal chain, the effects in these paths are processed simultaneously.

The, often misunderstood, role of the mixer block in this case is that in the second (parallel) example people tend to pan A and B hard left and right in an attempt to get the maximum stereo spread.

However, in doing this you are merely using one side of both effect (left or right side). And in most cases using only one half of the effect's signal is not sounding very good. (although there are some exceptions I will discuss later in this document)

So, as a general rule of thumb leave the panning controls centred, unless you are pursuing something specific.

For instance if you are using a dual amp configuration you will most likely pan the paths hard left and right to get a really wide spread of your sound.

Another reason to set this panning hard left/right is described in the section “How to widening your sound” later in this document.

As for the level controls: use these to get the right balance in volume for path A & B or to raise/lower the overall volume.

In most cases I do not use this very much. The most common exception is when I need to raise the volume of a clean patch.

Choosing the amp, cabinet and microphone

I should not be writing this section because each and every option here very much depends what you are after, if you are after a metal tone you will certainly choose different component then if you were looking for a typical country tone (for example). And secondly, the type of guitar/pickups also have an impact on your choice of components. So, here I will stick to fundamentals only.

Most amps come in two shapes, the FULL version and the PRE version (the latter does not have the power amp section and is often , but not exclusively, used when you use and external power amp and cabinet). This also implies that the PRE models do not have the cabinet Deep Editing Parameters. But if you choose a pre-amp model you still get a cabinet & microphone. In this case you may want to set this to “no cab”. NOTE you cannot deselect the microphone, and when the cabinet is deselected (set to “no cab”) is microphone is still visible but it does not do anything auditable to the sound. For live patches, in which case I use a power amp (Marshall Valvestate) I mostly use the pre version of the amp model and I also use a cabinet and microphone for these models. I do this because this way some harshness is removed.

NOTE: when editing on the device itself you will normally see and asterisks in the upper left corner of the screen as an indicator you need to save the preset in order to make it definitive. When changing cabinets and or microphone this asterisk will not appear so make sure the press the save button twice if you intend to keep your changes. (This issue does not exist in the HD edit software, here you do see the asterisk as a sign that something has changed)

If you have acquired additional model packs you will notice that there are also models that do not have a corresponding PRE version, this is because these amps are migrated from older POD’s. Furthermore, in the additional amp models many new amps do not have a specific dedicated cabinet but are mapped to existing cabinets.

With the FULL models you get a default cabinet and microphone assigned and in most cases this will do but do experiment with the cabinets, because the effect on the tone is significant.

As for microphones: I didn’t realise this before I wrote this document but for high gain amps and distorted tones I almost always use the Neumann® U87 Condenser (87 condensor) or the Sennheiser MD 409 Dynamic (409 dynamic). For low gain amps and clean tones is use most of the time the Shure

SM57 Dynamic (57 on/off axis). Of course, the choice of microphone is personal and the fact that I use the above mentioned microphones mostly does not imply I don't use the other ones.

Using an external pre-amp

If you have, like I do, another pre-amp at your disposal, you can use this pre-amp in the signal chain as an additional extra soundsource. Depending on the pre-amp capabilities and your objectives you can either use the FXloop of the POD HD or use the FX send/returns of you pre amp (if available)

If you are using the POD HD as the 'leading' device, you will need to use the effect send-return effect loop (FXloop) and set the send, return and mix levels according to your(and the external app's) needs. NOTE: The FXloop of the POD is a bit noisy in my opinion.

For connecting my Peavey Tubefex, I use the above settings. The best advice here is to listen carefully. And don't forget: obviously you will need to make a physical connection with the external amp by connecting the FX send and return cables of from the POD to the external amp's input and output (or if you can connect to the amp's effect loop(whatever works for you)). Furthermore, at the back of the HD PRO (don't know if this also applies to other POD HD devices) there is a dedicated switch for the FX Loop: Set this switch to **LINE for** line level devices. The loop can also be used with stomp boxes by setting the switch to **STOMP**. As always, checkout what is best in your situation.

You can use your external amp as an exclusive source(without amp models form the POD HD) of in combination with the POD's amp models.

If you external amp is MIDI controllable you can control you amp channel switching with the POD. I will discuss this (and the limitations later in this document)

Where in the signal chain you will need to position the FX loop and/or POD amp model will be discussed in the next chapter.

Alternatively you can use your external pre-amp and the main source. Connect the amps effect send to the POD's input (remember that you may need to change the input setting of you POD if you using are the AUX input of the rear of your device instead to the guitar input) and connect the unbalanced outputs of the POD to your amp's effect return. To me this was a, sound wise, better option, because I have the option to bypass the tube section of the Peavey Tubefex and I can therefore still take full advantage of the POD's modelling capabilities. So check-out these two options and choose whatever works for you.

Amp Positioning

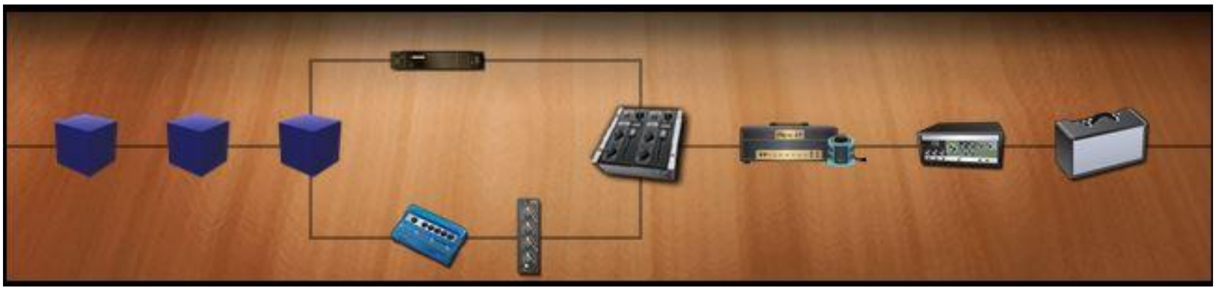
This topic might look a bit weird at first sight but here we have some options that will have serious impact on you final tone. In the POD HD there three option if it comes to where to position your amp model (or external preamp) in the signal chain (this applies only to single amp patches):

a) Before the A & B paths (so-called pre-position)
b) After the A & B paths (after the mixer) block (so-called post position)
c) On either the A or B path.

Looking option A:

In this example the signal goes to the amp first and then the two paths are processed in parallel and merged in the mixer block.

Looking at option B(with the same components):



This is basically the opposite first the signal is parallel processed in path A & B before it is merged in the mixer block and send to the amp.

Finally option C:

Again, we make use of parallel processing. In path A the signal is processed through the amp and in path B a clean signal is passed through the two effect modules. NOTE: make sure the bypass volume (BYP VOL) is maxed out (100%) if you intend to (de)activate the amp (in path A) with a footswitch and you still want a signal to pass through path A.

Whatever option you need depends on your specific requirements. I do suggest that you play with these option to notice the tonal differences of them.

Personally I mostly use option A (the pre-position amp position) and occasionally I used option C especially if I want to combine the slightly distorted sound with a clean sound and I do not want a second amp in order to save DSP.

IMPORTANT: Not all FX are stereo, some effects and all Amp models are mono. Therefore, if a stereo signal reaches a mono effect (or amp) the output will be in mono (the stereo signal is “mono-ized”)

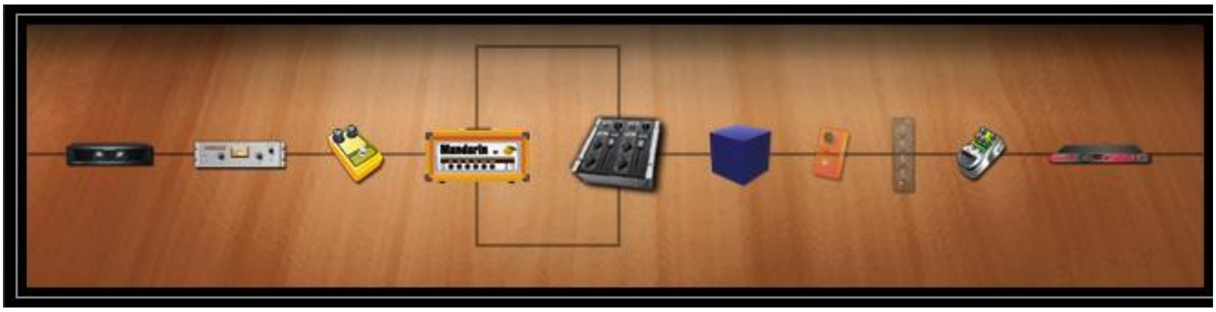
In the previous example Option B will therefore merge the two stereo effect blocks into a mono signal at the amp block. So be aware of this if stereo is important.

Signal chain: ordering effects

There are no strict rules, what works for you works for you, easy! So just a few tips to get started.

The first thing you need to be aware of is the fact that signal routing on the POD goes from left to right, pedal boards usually go from right to left. As for effects: I use dynamic and distortion effects before the amp and modelling, delay and reverb effects after the amp. EQ's anywhere I need them.

A logical sequence(but not necessarily) for effects is:



In this example I begin with a noise gate, then compressor and overdrive followed by the amp. After amp I use a phaser and StudioEQ (for volume compensation) and finally a delay and reverb.

Sometimes I use a compressor after the amp which also works fine:

This is a clean patch with Mesa Dual Rectifier(treadplate) were I have the Noise Gate and StudioEQ (to lower the gain) before the amp and after the amp smart harmony followed by a tube compressor and finally delay and reverb. Note that the smart harmony id in path A, so I can mix it with the unaffected signal from path B (see the chapters on “understanding the mixer block” and “amp positioning” for more details on parallel and serial processing)

Deep editing parameters

This is a complicated but powerful issue. There are two types of deep editing parameters (DEP's), one set for the amp and one set for the cab. Let look at the amp DEP's first:

Master: This controls the amount of power amp distortion and it also influences how strong the effect of other parameters will be (with more master volume the effect of the other DEP's is greater. Personally I like to keep this master DEP at 50 to 75%. Too high values may introduce fizz.

Sag: in most cases I set this parameter to zero as it makes the tone more responsive and percussive. However this will also make the tone slightly thinner and you need to compensate for this with the bass and mid parameters.

Hum: It does what it says this parameter adds hum to the signal, I do not use this often, but a little hum may make the tone a little darker sounding, do experiment!

Bias: This controls the bias of the power tubes. I tend to set this to the max for more tube compression and because low settings may introduce fizz.

BiasX: sets the voicing of the power tubes. Set this high for more tube compression. This will however also introduce fizz. I set parameter mostly low for a more percussive feel.

Now for the cab DEP's:

Low cut: I use this one to remove the boomy lows from the signal (usually around 10 o'clock. This may vary from amp/cab/mic combination).

NOTE: the following controls will only be audible when the POD is in Studio/Direct output mode.

Resonance level: I set this to zero by default as to my ears it gives the best tone. However if set to zero the thump and decay parameters have no use anymore, because these depends of the resonance setting. So if you need some extra bottom in your tone you might want to raise the resonance value and use the thump parameter

Thump: use this parameter to add some extra bottom. Start at zero and then increase the amount of thump until your satisfied. This parameter only works when the resonance parameter is not set to zero. The higher the resonance setting the more auditable the working of this parameter will be.

Decay: at a low level it gives a "tighter" speaker cone, at high level a "looser" speaker cone. This parameter only works when the resonance parameter is not set to zero. The higher the resonance setting the more auditable the working of this parameter will be.

How to tame a hi-gain amp model

As said earlier, I always aim for four flavours (patches) for each amp/cab combination. So If you are trying to do this with the high gain models you might run into the challenge of creating a clean patch with these models. Typical high gain amps are Bogner Überschall(Bomber Uber), Bogner Shiva (Mahadiva), ENGL Fireball(Angel F-ball) , Peavey 5150 (PV Panama), Mesa Dual Rectifier(Treadplate) and all of the Line 6 models. How can we make these amps producing an undistorted clean tone? The key here is not to overdrive the pre-amp. For starters: use only input 1 and set the second input to variax, this will lower the input signal with roughly 6dB's. This will however not be enough and as a second step we will need to set the pre-amp drive as low as possible (0-1%) If the signal is still too hot and you still here a (somewhat) distorted tone add a StudioEQ before the amp and use the gain to lower the input signal until the tone is clean (enough). Then use the other volume controls (preamp & poweramp) to set the proper volume for the clean patch. You might even have to raise the volumes in the mixer section to achieve the desired overall volume or add an second StudioEQ after the mixer to raise the volume, but remember this solution will cost you an effect slot. As an alternative you could also use the "PAD" switch on the device, in PAD mode the input signal will be lower, but this is a global setting and you probably not want this for all your patches.

Alternatively, you can put the amp AFTER the mixer block and use the mixer block level controls to lower the input signal. I found that with a setting of -20 dB I was able to get a pure clean tone from the Mesa Dual Rectifier (Treadplate):

This way you are also saving a effect block as you don't need the StudioEQ to lower the input signal.

How crank up a low-gain amp model

This is about the opposite of the previous topic. Low gain amps (such as the Fender and Vox models) are good for clean patches but they need some additional help in producing convincing lead sounds.

First of all set input 2 to "same" this will boost the input signal with 6dB's. But if the drive is all the way up and the tone is not satisfactory you will need to add a distortion effect. I always use a compressor before the distortion. When you use a distortion effect you can set the drive on the amp relatively low as the pedal does the work. My personal favourite distortion effects are Overdrive, Tube drive and Screamer. As for the compressor, I tend to use the tube compressor almost exclusively.

Compression and Distortion

As said earlier my favourite distortion effects are Overdrive, Tube Drive and Screamer an as for compressing I normally go for the Tube Compressor. Now if you are using high gain models you don't have to use the distortion of the amp itself. It may be a very viable option to lower the drive of the amp and add a compressor and distortion effect instead. You might be surprised how good this will sound. And as always, don't overdo it on the drive setting. Most patches I have heard have way to much distortion, resulting in this infamous "can of bees" sound.

Clean tones: handling dynamics

Clean tones have, by definition, more dynamics than distorted ones. So you might want to tame those dynamics. There are several ways to do this. The most logical option is to add a compressor to your signal chain in order to smooth out your clean tone. Normally a compressor is used BEFORE the amp model, but I suggest trying to put in AFTER the amp model because this way the compressor works more like a limiter and does not impact the natural string attack so much. The compressors that are most suitable are the Tube Comp and the Vetta Juice.

Alternatively you can experiment with the tube microphone pre amp (Vintage Pre) before the amp model. This also smooth out the dynamics.

Delay and Reverb

These are (when used) always the two last effects in my signal chain. For delays I tend to use either the ping-pong delay or the stereo delay and occasionally I use the ducking delay especially on lead patches. I set the mix parameter for all delays between 25 and 30%. Delay is always mapped to footswitch 3.

For reverb I mostly use the plate reverb with the mix between 16 and 25%. You might want to switch off the reverb while recording and add reverb during the mixer stage. In that case assign the reverb to one of the footswitches. I use footswitch 4 for reverb and by default it is on. I have been playing around with the ducking delay and that's an interesting option as well because the reverb is than only heard when stop playing, so the sound is tighter works great for Rhythm patches.

Some other reverb option to look at the early reflection parameter in the AMP section of the POD (only available when a speaker cabinet has been selected). This standard available option can add a room sound to the amp's tone (and does not cost you any extra system resources)

Early reflections are sounds that arrive at the listener after being reflected from parts of the listening space, such as walls, ceilings and floor. They arrive later than the direct sound, often in a range from 5 to 100ms.

Effects and volume

One of the known issues with effects on the POD HD is that most of them make the volume quite a bit louder when engaged. This volume change can be up to 6dB (80A Flanger). If I need to compensate for this change in volume I add the StudioEQ, and only use the gain control to compensate for the volume difference. Make sure the StudioEQ is assigned to the same footswitch as the effect, so they are turned on/off simultaneously.

How to boost a signal for soloing

I might be an idea to include a boost option in your patch that you can engage for soloing. I must admit, I don't use this option as: a) I use the POD mainly in a studio environment and b) If I do need a boost, I change to another patch (in my case always preset 'D').

There are multiple options for this: add an (extra) distortion effect with the drive at (or close to) 0, add a Studio EQ and increase the gain only (typical clean boost), add a compressor or add a volume pedal.

If you do not use the effect loop of your POD to connect to external effects/amps, you can connect the physical send and receive connectors with a patch cable, send the "send" at 0db and the "receive" to the required boost level. Set the mix at 100%.

In all cases, beware of digital clipping as you are increasing the general level of the signal, and if you experience this you will need to make significant changes to you patch. So as a rule of thumb: if you intend to use a boost option for you patch start with the boost setting first!

Widening your sound

If you are looking for a way to widen the sound of your patch there are some interesting options to explore. The easiest way is to use a stereo effect in your chain (keep in mind that you should not use a mono effect later in the chain). Other than this you can also try some of the following options:

Put the "frequency shift" one path (after the spit) set the mode to stereo, the frequency to 1Hz and the mix between 50 and 100%. In the mixer settings Keep PAN A and B both to centre.

Alternatively you can use a digital delay instead: Sync off, Time 20ms to 40ms, feedback 0-3%, mix at 75%, set the PAN controls in the mixer to hard left and right. Again to not use mono effects afterwards. Do experience with different settings!

A third option is to add a GraphicEQ with neutral settings in one path nothing in the other path, then set the PAN controls in the mixer to hard left and right and listen to what has happened. The POD adds a little bit of latency for each engaged effect. By introducing latency on one side only you create a comb filter. A comb filter adds a delayed version of the signal to itself, causing interference. With this option I usually raise the 440Hz with 1-3dB for some more bottom and I reduce the 1.1kHz with -3dB. The last option for "a wall of sound" is using two amps/cabs. You can also combine dual amps with the tips given in this section.

NOTE: As said earlier in this document, not all effects are stereo preserving. If a stereo signal reaches a mono effect or amp block, the output will be mono.

Using Dual Amps

This is a great feature of the POD HD. The options are almost endless. However you may run into DSP issues if you do not have the “X” version of the POD HD(Pro or 500) So the you are a heavy user of effects you will probably run into the “DSP limit reached” issues. One trick with dual amps is to use the same amp in both paths but vary the tone controls, cab and microphone. This way you can use one amp for the bottom end and the other for the highs. Choose the right cab for the job (such as the hiway with the 57 on axis for the highs and the XXL V-30 with 409 Dynamic microphone for lows) Another option is to use a clean and a overdriven amp (Such as the Mesa Dual Rectifier(Treadplate) for overdrive and the Dr. Z Route 66(PhD Motorway) for the cleans, use the mixer to blend the two paths.

As switching presets within a bank causes a short period of silence, using dual amps is an option to overcome this:

In this example I use two amps: Mesa Dual Rectifier(Treadplate) in path A, Soldano(SOLO-100) in path B. These two amps assigned to the same footswitch. This way the footswitch acts as a toggle switch between the two amps: one if on, the other is off. Make sure that the bypass volume of both amps is set to zero. You can access this parameter in the amp section when the amp is deactivated. In the picture below you can see the bypass volume parameter of Amp B (BYP VOL) is set to zero.

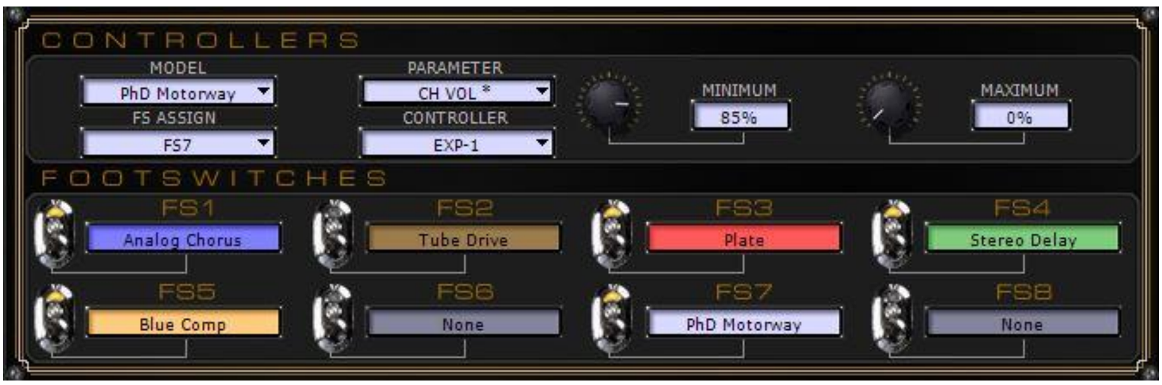
The last interesting option in a dual amp configuration is what I call morphing as I will explain in next chapter.

Morphing technique

The morphing technique also applies to the dual amp configuration. Here we use the expression pedal to seamlessly morph from one amp to the other by assigning the expression pedal to the channel volume. In next image you see that both channel volume knob are controlled by expression pedal 1.

This setting is done in the controllers section of HD edit. Note that for the first amp the minimum value is set to zero. (this means that in the heel position there no sound coming from this amp.

For the second amp we see the opposite the maximum value is now set to zero, which implies no sound is coming from the amp with the pedal in toe position.



Using the POD HD as a midi controller

The POD HD can be used as a midi controller (with limited functionality). This way you can synchronize patches with external devices, for instance an external midi controllable amp. When you switch patches on the POD HD it will send a 'program change' message to this external amp as well.

In this MIDI controller tutorial I will use a midi controllable pre-amp (Peavey Tubefex) and use the POD HD to change the presets on the Peavey. (basically this should work for all midi controllable devices). In other words: How to implement 'program change' messages.

Let's start with the physical audio connection:

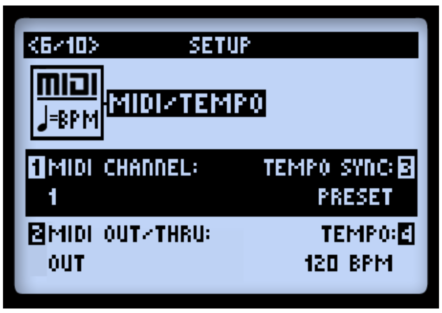
The Pre-amp is connected to the POD using the POD's FX loop (send & return) You can set the desired send and return levels in the FX block. Note that the use of the return is optional if you are able to send the output if the external amp directly to a mixer or poweramp.

Note that you can also set the mix. In this case I have set it at 100% but if you want to blend it with amps models from the POD you can do this here.

You might need to set the same parameters in the pre-amp is the signal is too noisy.

For controlling the pre-amp you will need a standard MIDI cable to connect the devices (which is a standard 5-Pin MIDI cable). To send MIDI communication to another device, connect the MIDI cable from the POD's **MIDI OUT/THRU** to the **MIDI IN** of the external device.

Then you need to setup the global MIDI settings



As you can see I use **MIDI channel 1** for exchanging information (you can choose any of the available 16 channels) And secondly I have set the MIDI out/thru to **OUT. Note:** Make sure the receiving device is *listening* to the same MIDI channel!

By default, the POD will send a 'program change' message for each preset in a setlist (bank & preset per bank). Remember my POD is set to ABCD mode (so 4 presets per bank). If I understand the discussion on the line 6 forum correctly you need to set it to ABCD mode and NOT to FS 1-8)

So what happens is that, for instance, preset 12A will send exactly the same program change message as preset 12A in another set.

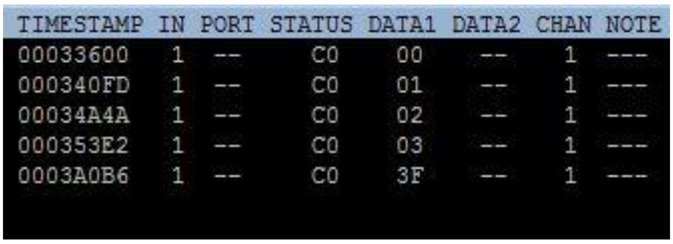
Within any setlist the following program change commands are given(by default)

Preset	Program Change Value
1A	0
1B	1
1C	2
1D	3
2A	4
2B	5

and so on....

So in a POD setlist you have 64 presents (from 1A to 16D) and range of program change values goes from 0 to 63.

With free monitor programs such as MIDI-OX you can actually see what messages are send by the POD. In the image below(from Midi-OX) you see the program change message for presets 1A thru 1D and 16D. The last gives a hexadecimal value of 3F which corresponds with 63 (decimal)



TIMESTAMP	IN	PORT	STATUS	DATA1	DATA2	CHAN	NOTE
00033600	1	--	C0	00	--	1	---
000340FD	1	--	C0	01	--	1	---
00034A4A	1	--	C0	02	--	1	---
000353E2	1	--	C0	03	--	1	---
0003A0B6	1	--	C0	3F	--	1	---

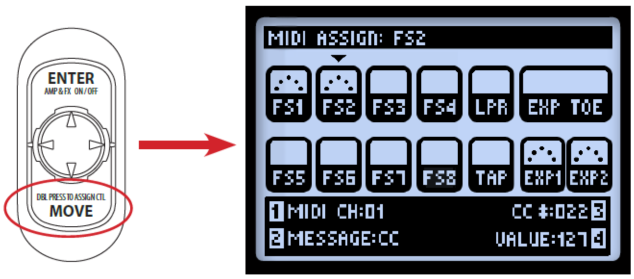
In some cases this maybe fine and if that is the case you're almost done. Look at what preset on the external device is triggered by the program change message, set this preset on the external device to the desired values, and do this for all four (ABCD) patches in any bank that uses the external device.

If you want more control you need to set the program change values for all the patches that need different program change values.

You can do this on the device itself or the HD Edit.

On the device:

Hold the MOVE button to access the MIDI assign screen



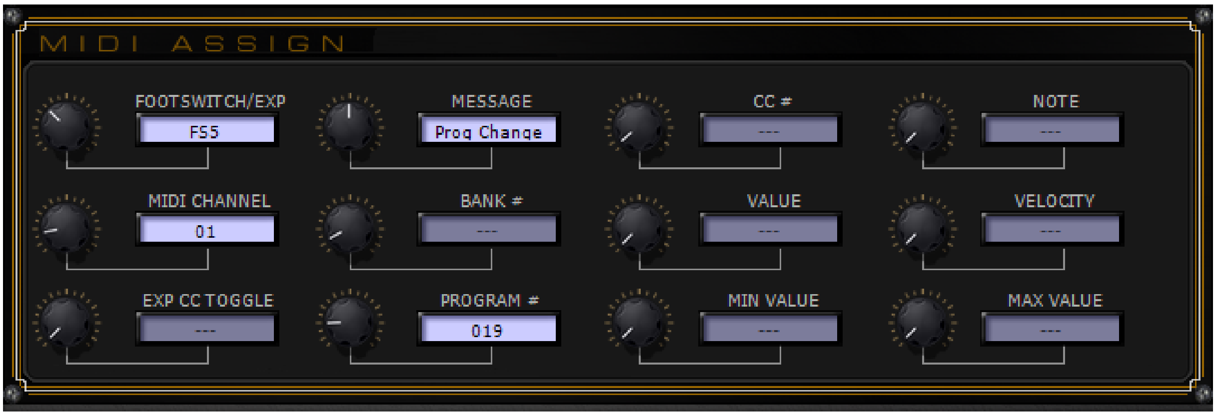
Choose which Footswitch you want to use. Note that in ABCD mode FS5 thru FS8 are used for the ABCD patches.

So if you want to change the ‘B’ patch, then select FS6

Set the midi channel (knob 1)
Set the message to program change 9 (knob 2)
Set the desired program value (knob 4)
Save

Using HD Edit

Go to the controller window pane



The operation is basically the same, in this example I have set the program change value for patch A (which is FS5) to program 19 on the external preamp.

Remember the above mini tutorial is based upon the assumption you want to send program change message to an external device when changing presets on the POD. You can however assign these program change message to other footswitches as well, for instance to activate an external MIDI controllable effect.

As customizing program change message is per preset it could become time consuming. You could make templates of certain settings and load these when you start building a new patch. Or if you always use the same external device settings duplicate the settings on the external device (if you want preset 2B to trigger the same program change message as 1B, copy 1B to the location of 2B on the external device. Alternatively you might be able to do **input mapping** on the external device like I can do with the Peavey tubefex. This means I can re-assign the incoming program change message to the desired preset on the external amp.

A word on the limitations:

The ‘program change option’ described above was enough for me at this point in time. It looks like you can only assign one MIDI message to a footswitch. This means that if you need to change a bank as well you will need to press two footswitches, one to trigger the bank change and one for the program change within the selected bank. Now, I am not a tap dancer nor want I to become one so I stick to the program change only for now. I guess if you do need multiple MIDI messages you’d better look for a separate midi controller.

Eliminating noise: hard gate versus noise gate

Especially with high gain patches you may want to eliminate noise. There are two options noise gate of hard gate. The noise gate is easier to operate, just adjust the threshold parameter and you are about ready. Using the hard gate gives however more control as you can set the open and close values yourself as well as the hold time (the time the gate waits before closing). As starting point for dialling in the hard gate I set the open threshold initially to 0dB and then make the softest where I want the gate to open(this can be a single note of even just rubbing a string) Usually this open setting is around -70dB. For the close threshold I start at 0dB as well, play a single note of chord and listen how long it is auditable I adjust the close threshold until the notes/chords are cut off in a natural way (usually I end up around -80dB) I leave the Hold time at 0ms. The decay is used to help when the cut off is unnatural. I set is somewhere between 0 and 200 ms, depending on the type of patch. If it is used for staccato playing keep the value low, set it higher for more smooth playing.

Levelling patches

The presets in an bank (ABCD) should be consistent in volume in respect to one another. You might think about monitoring the volume of a preset with a dB meter and this will function to some extent. However there is something called perceived volume, in other words how do you hear it. As a rule of thumb I start with the clean patch and use the DB meter from my Digital Audio Workstation (DAW) to set the initial volume, which should not exceed a recording level of -12dB because we will need the headroom in the later stage. The other 3 patches are primarily done by ear but should not exceed the -10dB point preferably.

Levelling patches across banks/patches is a more difficult issue as amp models do not have the same initial volume. So if you are planning to use various amp models it is wise start with the one that is the quietest in volume and adjust the louder ones afterwards, because the other way around will simple not work (at least not that easy)