AMP block parameters

From Axe-Fx II Wiki

Contents

About amp modeling

AMP block: supported by which Fractal Audio products?

- Axe-Fx II: yes.
- AX8: yes.
- FX8: no.

Jma's Amp & Cab Quick Reference Guide

- Forum member jma provides a handy reference guide (http://olongjohnson.xp3.biz/index.html) , covering AMP and CAB block parameters and descriptions, MIDI CCs, Drive block descriptions etc.

AMP block and X/Y switching

- The Amp block supports X/Y switching.

Audio gap when entering the AMP block's Edit mode

- If you press Edit to adjust Amp block parameters, there will be a short gap in the audio stream because of processing tasks. source (http://forum.fractalaudio.com/axe-fx-ii-wish-list/40519-wish-no-gap-audio-when-editing-amp-block.html#post546660)

Resetting AMP block parameters

- Double-clicking Bypass (Axe-Fx II) resets an Amp block completely. This resets ALL block parameters to their default values for the current amp model. Use this to start from scratch. The AX8 and FX8 have their own ways of resetting an effect block: more info (http://wiki.fractalaudio.com/axefx2/index.php?title=Buttons,_switches_and_knobs_on_the_AX8_and_FX8) .
- De-selecting the current amp type by selecting another type and selecting the previous one again (aka "up/downing amp type" or "reselecting amp type"), resets most parameters to their default values for that particular model. Here's the list of affected parameters (as measured on Jan 30 2016):
    1. *PRE: nothing except the BRIGHT knob*
    2. *PWR: everything except OUTPUT LEVEL*
    3. *SPKR: everything*
    4. *EQ: all sliders and EQ Type (see ADV)*
    5. *PWR DYN: everything except COMP TYPE*
    6. *PRE DYN: everything except OUTPUT LEVEL*
    7. *DYNEQ: everything except CHAR TYPE, CHAR Q*
    8. *ADV: all except INPUT SELECT*
    9. *TREM/MIX: TREM FREQ, OUTPUT LEVEL, BALANCE, BYP MODE.*
- The parameter SET AMP(S) TO DEFAULTS in Hardware menu: Utility (Axe-Fx II) sets a more limited set of parameters in the amp block(s) (X and Y) to their default values. Use this after a firmware upgrade when the release notes recommend doing this, or

with old presets, to make sure that essential amp model parameters are set to correct values, while maintaining the current values of basic controls.

Controlling amp gain

- Here's a list of things you can do to increase or decrease amp gain. Also read this Wicked Wiki thread. (http://forum.fractalaudio.com/axe-fx-ii-discussion/49193-wicked-wiki-4-increasing-amp-gain.html) Note: adjusting Input Level in I/O does not affect gain!
  - Each Amp has one or more Drive parameters controlling the amount of gain. You can assign an external controller to vary the gain.
  - Adjust the global Amp Gain parameter in Hardware menu: Global.
  - Use the Input Trim parameter in the Amp block to boost the input signal.
  - Increase Master Volume for more power amp distortion.
  - Increase Mstr Vol Trim on the Advanced parameters page.
  - Use Boost in the Amp block.
  - Engaging the Bright switch often increases gain.
  - Engage the Sat(uration) switch.
  - Increase Level on the IN/GTE page to boost the input signal.
  - Insert a Drive block before the amp and set it to FET Boost or Tape Dist for a clean boost and attach a pedal to its Drive parameter. Or set it to TS808 or Tube Drive block with Drive all the way down and Level maxed (tightens low end). Alternatively use a NullFilter block before the amp instead of a Drive block, with lots of dBs to boost the amp's input level.
  - Jay Mitchell: "Cascade one amp block into another. Turn off "Sag" in the first one. Now you've got an extra preamp feeding your amp, which opens up an incredible spectrum of gain staging. For example, think Twin Reverb preamp, with Plexi tonestack set to "post," feeding a Plexi 2 with default settings. The possibilities exceed anything one person could hope to explore in a lifetime. You can get it awfully close. You want to minimize the effect of the preamp in Amp 2. To do this, set the 2nd Amp's Bright to off, MV to a high value and find a neutral setting for the tone controls in the 2nd Amp. Then use Drive in the 2nd Amp for your MV. The amp types you choose for this arrangement will make a huge difference, as will quite a few parameter settings."
  - Jay Mitchell: "Start with Tape drive, set the clipping mode to "HV tube", Drive moderate, Level as appropriate for the amp block it's driving, and you'll have another tube gain stage, complete with EQ."
  - Cliff: "If you want less distortion on low notes there are several ways to achieve this: 1) Use the Low Cut, 2) Increase Definition, 3) Increase Xfrmr LF. The first two reduce low frequency content going into the preamp, the last one reduces it going into the power amp (there's a hidden low-cut between the preamp and power amp but the user doesn't have access to this). So it depends on where you are getting the distortion from. If it's mostly preamp distortion, use #1 or #2. Otherwise use #3. You can add bass back with the Depth knob or in the EQ page. The default settings are accurate for the amp being modeled. As amps become more modern, it seems people's taste agree with yours and many modern amps feature aggressive low-cut and then add bass back in the power amp. So what you are doing is "modernizing" your amp. For example, the HBE has a very high low-cut and then adds bass back with a fixed Depth circuit." source

(http://forum.fractalaudio.com/axe-fx-ii-discussion/59934-warmth-6-0-9-0-a.html#post747576)

- Cliff writes about all the different gain controls. (http://forum.fractalaudio.com/tech-notes/95018-understanding-all-different-gain-controls.html)

AMP controls behavior

- Firmware 6: "In general most knobs now behave exactly like the actual amp when possible. In a few instances there may be minor discrepancies between the knob position of the model and actual amp due to programming constraints and/or peculiarities of the actual amp (such as poor potentiometer tolerance). Due to variations in presence circuit topologies the taper of the Presence parameter, in particular, may vary between the model and the actual amp. In other words, a different setting on the model may be required to achieve the same response as the actual amp. In most cases however, the Drive, Treble, Mid, and Bass knobs will be accurate to within 10% of the actual amp."
- If the real amp has two gain controls, the one which is the closest to the 1/4" input is modeled as Input Drive in the model. The other one is Overdrive.
- If the real amp has two inputs (f.e. low and high), the model is based on the high input. Set Input Trim to 0.500 to get the equivalent of using the low input.
- Cliff's comments:
  - "The tapers of the controls isn't really MIMIC per se. It's just me doing the dog-work and measuring the tapers. So, in that regard, the tapers match my amps. However manufacturers are notorious for changing tapers, sometimes right in the middle of a production run due to part availability. Furthermore the tapers in the Axe-Fx assume "true" logarithmic pots. Consumer-grade log pots are not true logarithmic, they're a crude approximation. At noon on a pot you'll get a nearly perfect match assuming the pot has 0% tolerance. As you deviate from noon there may be some error due to the approximation in the actual amp. As you get to the ends of the travel the error will decrease to zero. At any point there shouldn't be more than 10% or so deviation between the Axe-Fx knob position and the real amp. Master Volume tapers are NOT matched. If they were the amp volumes would jump all over the place when you switched amp types. IIRC I use a Log10A for the MV." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/80497-axe-fx-ii-firmware-version-12-04-public-beta-12.html#post978667)
  - "In most cases the knobs do translate. Usually within 10%." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/58613-axe-sounding-terrible-through-guitar-cab.html#post732551)
  - "If an amp has just "Tone" then that's mapped to Treble. Leave bass and mid at noon." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/67260-question-proper-axe-eq-values-amps-dont-have-them.html#post828717)
  - "If the amp has no Master Volume, set the MV to 10 (the model will default to 10 when you select it). If the amp has no midrange control, set the MID to 5.00. If the amp only has a "Tone" control, set Bass and Mid to noon and the Treble control is your tone control." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/69693-axe-tone-controls-vs-real-amps-tone-controls-question.html#post855688) and source (http://forum.fractalaudio.com/axe-fx-ii-discussion/67260-question-proper-axe-eq-values-amps-dont-have-them.html)
  - "Master Volume tapers are NOT matched. If they were. the amp volumes would

jump all over the place when you switched amp types. IIRC I use a Log10A for the MV." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/80497-axe-fx-ii-firmware-version-12-04-public-beta-12.html#post978667)

- "The only controls that do not necessarily match are the Presence and Master Volume. The Master Volume taper is the same for every amp. This was done to provide some consistency in finding the "sweet spot" of power amp break up and to prevent wildly fluctuating volume levels. The Presence control has a reverse log taper which gives the control a more reasonable behavior than that of most tube amps. A typical tube amp's Presence control does nothing over the first 80% of it's rotation which is stupid." source (http://forum.fractalaudio.com/amps-cabs/93693-real-vs-model-collection-thread.html#post1124915)
- Firmware 18: "Presence and Depth controls may not match the taper of the actual amp. On most amps the Presence control does nothing until you turn it almost all the way up. This seems a bit silly so we make the Presence behave more logically. Same goes for the Depth control. Drive, Bass, Mid and Treble will match the actual amp within the tolerance of the pots. Another caveat when comparing amps: many times the knobs aren't "centered". IOW if you put the Treble knob at noon it isn't actually at 50%. You can see this by turning the knob all the way down and all the way up. It may not be symmetrical. This happens when the pots don't have a flat spot and/or the pot is rotated within the mounting hole." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/96570-new-18-00-public-beta-5.html#post1158815)
- "IMO accuracy is paramount and that's why we've devoted so much in resources to that end. The purpose of a modeler is to model amps as accurately as possible. Now it's impossible to account for component tolerances and tone controls can vary as much as 20% or more between two same amps. We therefore model all amps assuming the tone controls are "perfect" ((IOW if the amp was designed with a 500K pot we use 500K even if our reference amp is off by some percentage). We also don't match the Presence and MV tapers for previously discussed reasons. In some cases the Depth taper does also not match although this is infrequent. However you can be assured that even if the tapers don't match the models will match at the extremes of the control range and therefore the model is accurate but there may not be a 1-to-1 correspondence between the amp's knob and Axe-Fx's knob, i.e. Presence on 7 on the amp may be 5 on the Axe-Fx but 0 on the amp will be 0 on the Axe-Fx." source (http://forum.fractalaudio.com/threads/axe-fx-vs-real-amp.107447/page-3#post-1286881)
- "Internet wisdom states that no two amps of the same type sound the same. That is true, but the reasons are far more simple than many would have you believe. Tales abound of esoteric effects such as wire dress, transformer orientation, phase of the moon, etc. And while these do have some effect, it is arguably inconsequential relative to the single biggest source of deviation: tone control tolerance. I've spent the last ten years modeling tube amps and the number one thing I see is that tone controls are very inconsistent devices. First of all the tolerance of the control is typically 20%. That's plus or minus 20% so 40% total. A 100K pot can be as low as 80K or as high as 120K. This is contrast to the tolerance of a typical passive component which is 5% or less (usually much less IME). Secondly the resistance at the midpoint can vary widely. A Log10A pot should be 10% of the resistance at midpoint. But, again, this can be off 20%. Let's take the case of a bass control which is typically wired as a rheostat. On one amp

the pot might be 10% high and the midpoint 10% high. Therefore with the control at noon (assuming, say, a 1M pot) the resistance will be 121K. Another amp off the assembly line might be 10% low. Therefore the pot will be 81K. That's a 40K difference between the two amps and that's not even worst-case. Now you can make the amps sound the same by simply turning down the control on one and/or turning it up on the other." source (http://forum.fractalaudio.com/threads/no-two-amps-sound-the-same-fact-or-fallacy.109537/)
    - "You'll never get the knobs to correspond exactly. Commercial quality potentiometers are terrible. They vary widely in both end-to-end resistance and resistance at midpoint. Variations of +/- 20% are common (that's 40% total!!!). The Axe-Fx always assumes an ideal potentiometer, i.e. a pot where the end-to-end and midpoint resistances are exactly the specified value. Furthermore commercial "audio taper" pots are not truly logarithmic. They use a crude piecewise approximation. The virtual pots in the Axe-Fx are true log. This is the #1 reason for the whole "no two amps sound the same". In fact they probably do sound the same but you need to adjust the pots on one (and possibly quite a bit) to make it sound like the other. For example if a tone control is at noon (5.00) on one amp you may need to set the other amp to anywhere from 3.00 to 7.00. This applies to any product that uses potentiometers, including drive pedals. This also means when matching any virtual amp/drive/etc. in the Axe-Fx to a real-world counterpart that you may need to deviate significantly to get the same sound. For example, to get our reference Dual Rectifier's orange channel to match the model I need to set the model's treble control to around 4.0 (with the amp's treble at noon). This is because the pot in the amp has a significant deviation from the intended resistance at the midpoint. It's a 250K linear taper pot but it reads around 100K/150K when set to the midpoint. This is quite typical of commercial quality pots." source (http://forum.fractalaudio.com/threads/so-about-that-new-si-diode-clipping-in-q5-xx-drive-block.120955/page-2#post-1439816)
- Conversion chart for real amp settings versus amp models. (http://forum.fractalaudio.com/threads/conversion-chart-for-real-amp-settings-vs-fas.106182/)
- The range of the gain taper in the Axe-Fx is 0-10. Volume on Fender amps go from 1 to 10. This translates to (source (http://forum.fractalaudio.com/axe-fx-ii-discussion/84090-thoughts-fw-14-ast-other-clean-ish-amps-4.html#post1024092)):
    - Fender 1 = Axe 0.00
    - Fender 2 = Axe 1.11
    - Fender 3 = Axe 2.22
    - Fender 4 = Axe 3.33
    - Fender 5 = Axe 4.44
    - Fender 6 = Axe 5.55
    - Fender 7 = Axe 6.66
    - Fender 8 = Axe 7.78
    - Fender 9 = Axe 8.89
    - Fender 10 = Axe 10.00

Fighting Fletcher-Munson

- Fletcher-Munson: how to avoid getting buried in the mix.

Power amp modeling

- Information about disabling power amp modeling.

More information about the Amp block

- About separating pre-amp and power amp.
- About amp model resolution: single or dual amp blocks (http://wiki.fractalaudio.com/axefx2/index.php?title=Amp_modeling#Amp_modeling:_single_and_dual_Amp_blocks_.28Axe-Fx_II.29) .
- About fizz.

Regular gain and tone controls

AMP TYPES

- List of all modeled amps.
- Fractal Audio AMP models: forum member Yek's series of threads on the forum, discussing each amp model. (http://forum.fractalaudio.com/search/419348/?q=%22Fractal+Audio+AMP+models)
- When in the Type page of the Amp block, the ABCD Quick-Control knobs control Drive, Master Volume and Level, respectively.
- Selecting another Amp Type resets many parameters to the default value for the type. See AMP block parameters: resetting.

INPUT DRIVE, OVERDRIVE, MASTER VOLUME

Input Drive and Master Volume:

- Vintage amps don't have separate gain (drive) and master volume controls. Master Volume defaults to "10" in these models. Use Input Drive for volume and gain.
- Cliffs comments:
  - "For clean tones the Drive control should be set fairly low and the Master set very high. On a real "Blackface", for instance, the Master is essentially maxed since that amp has no master volume. A Blackface typically achieves full power at around 10-11 o'clock on the volume (Drive). It's also insanely loud. Beyond that everything starts to saturate and clip. If you set the Master low and the Drive high, for clean tones, the low end will tend to get muddy. Good cleans are obtained with little, if any, preamp distortion and a nice amount of power amp distortion. Power amp distortion has a much different character and tends to be glassy and bouncy. Preamp distortion is rougher and more compressed."
  - "The real key is to adjust the relative amounts of each. You want to balance preamp and power amp distortion for the best tone. What I do is start with the MV low and turn up the drive until I get the desired amount of gain and sustain. Then turn up the MV until I get the desired compression. Then fine-tune each." source

(http://forum.fractalaudio.com/axe-fx-ii-discussion/46055-pre-amp-vs-power-amp-distortion-axefxii.html#post601681)

Master Volume:

- Firmware 10: "Amp models now default to a starting Master Volume setting when selected. Also, the proper setting for non-MV amps is now a Master Volume setting of 10.0. Non-MV amps, therefore, will default to a value of 10.0 when selected. If more MV drive is desired for non-MV amps, the new MSTR VOL TRIM parameter in the Advanced GUI page can be used to increase (or decrease) the Master Volume. The starting MV value for non-MV amps is roughly the "sweet spot" for the amp. This is the point where the power amp starts to contribute to the tone and feel of the amp. Decreasing the MV will typically cause the amp to get brighter and less compressed and increasing the MV will cause the amp to get more midrange focus and more compressed. As always, your ears should be your guide."
- Firmware 15: "The amp modeling improvements have resulted in a significantly increased "sweet spot" for the Master Volume control. Previous advice to keep the Master Volume low for high-gain amp types no longer applies and, in fact, increasing the Master Volume can result in better tone (more bloom and swirl) and much better feel (due to power supply sag). Therefore most non-MV amps now default to a higher value than previously. This may result in louder preset volume which will necessitate reducing the Output Level to compensate."
- The MSTR VOL TRIM parameter in the Advanced GUI page can be used to increase (or decrease) the Master Volume. Cliff: "Just multiplies the MV by the amount. You only need to use it if you want more power amp drive and your MV is already at 10. IOW, if MV is 10 and you set MV Trim to 2.0 then the MV will be 20." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/67870-master-volume-trim.html#post834697)
- Cliff's comments:
  - "Master Volume tapers are NOT matched. If they were the amp volumes would jump all over the place when you switched amp types. IIRC I use a Log10A for the MV." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/80497-axe-fx-ii-firmware-version-12-04-public-beta-12.html#post978667)
  - "The Master Volume (MV) DOES affect the tone. It sets the level into the power amp modeling. The Level control has no affect on the tone. For MV amps, i.e a 5150, adjust the MV until the desired amount of power amp distortion is obtained. Most MV amps rely on preamp distortion and don't produce much power amp distortion. If you turn the MV up too high on them the tone will get muddy and flubby. Non-MV amps rely primarily on power amp distortion so you need the level into the power amp to be hot enough to push the power amp into distortion." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/55340-so-many-volumes-options.html#post698522)
  - "I just start low and bring it up until I get the desired compression. Then I chug the E string and if it's too buzzy or flubby I drop it down a bit. For tight, high-gain stuff you want to keep it low. For liquid, spongy tones you want to set it higher." source (http://www.thegearpage.net/board/showpost.php?p=13739825&postcount=27)
  - "MV is the most important Amp block control for tone. You have to find the sweet spot. Start at 3 and increase until desired compression is reached. Stock presets are set to sweet spots, subjectively (based on the guitar used and personal

opinion). Do not use MV for volume and don't turn it up too much (unless it's a non-MV amp). If an amp has Input Drive and Overdrive controls, use Input Drive for tone shaping and Overdrive as a flat gain control."

- "As you increase the MV you drive the virtual power amp harder. As you drive the virtual power amp harder the frequency response becomes more dynamic, just like the real amp. And, just like a real amp, there is a sweet spot where the compression, dynamic frequency response, distortion, etc. just feel "right". The particular settings for the sweet spot depend on the Input Drive, tone controls, etc. so there are no hard-and-fast rules. So, as always, use your ears. And don't be afraid to use your ears. You'll be a better player when you learn how the controls interact and how to find the sweet spot. The great players in guitar history new how to work their amps. They didn't rely on someone else to tell them where to set the knobs, they learned themselves." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/80497-axe-fx-ii-firmware-version-12-04-public-beta-19.html#post979668)
- "With Version 13.xx firmware the constraints on MV are lessened. Due to the improved power tube modeling you can increase the MV more without the tone getting flubby or harsh." source (http://forum.fractalaudio.com/cliffs-notes/77918-watch-your-master-volume-3.html#post994936)
- "The way I dial in the MV is to turn up the MV until the amp stops getting louder. This is the point at which the power amp is saturating heavily. Then I back it off until I get the right amount of preamp and power amp distortion. That's the sweet spot where you get the tone and the dynamics. Too little MV and it's all preamp distortion and there's not much dynamics. Too much MV and the power amp is clipping too much and it can get flubby and/or harsh. Just as with a real tube amp you have to get the power amp cooking to get the best tone and feel. Get that power amp working hard and the supply bouncing around and things get nice." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/96781-18beta-kick-balls-2.html#post1161823)
- "The taper of the MV on the Axe-Fx does not necessarily match the taper of the actual amp. We use the same taper on every model. IIRC it is a Log10 taper. Many amps use a taper that is more abrupt than that for marketing reasons. For example, a Blues Jr has a linear taper MV. This means the amp is near full volume when the MV is at 3. This gives the impression that an amp is "loud". When the unsuspecting customer is testing the amp and it gets really loud with the MV on 2 the customer instinctively goes "wow, this amp is loud, it must be good". Anyways we use a consistent Log10A taper on every model. In general this means you need to set the MV higher than you would on the real amp. For example if the real amp has a linear taper halfway on the model would be equivalent to 1 on the amp (assuming the amp is calibrated from 0 - 10). The taper of a logarithmic pot has the nomenclature LogXA where 'A' indicates audio and X is the percentage of the element resistance from wiper to CCW terminal with the pot at 50% rotation. So a 1 Megohm Log10A pot would have 100K between the wiper and the CCW terminal when the pot is at "noon"." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/96781-18beta-kick-balls-2.html#post1162192)
- "All Axe-Fx models have been "modded" to include a Master Volume. Setting the MV to 10 effectively removes it from the circuit." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/98368-non-master-volume-amps-10-vs-9-a.html#post1180180)

- "Those amps (JCM, SLO) are all designed to get their character from power amp distortion. If you don't push the power amp all you are hearing is the preamp which is voiced to be trebly. The power amp then compresses the highs and the sound gets fatter." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/103152-1-biggest-user-error-3.html#post1235311)
- "I was helping a customer out yesterday. He was complaining about lack of feel and thinness. I asked him what his Master Volume was set at. He said 1.5. I asked "why so low?". He said "because that's where I set it on the real amp". I explained that the MV on the Axe-Fx has a much gentler taper than real amp and that 1.5 on the real amp is probably around 5 or more on the Axe-Fx. So he cranked the MV up and exclaimed "wow, that's what I'm looking for!". The MV taper on the Axe-Fx is a Log15A taper. This means the output is 15% of the input when the "pot" is at noon. Most amps use a higher taper than this, say 30A or even a linear taper. This is done as a marketing ploy. The unsuspecting customer sets the MV to 2.0 and goes "wow, this amp is loud". Thing is the amp doesn't get much louder. This also makes adjustment difficult because most of your volume range is constrained to a small fraction of the dial rotation. The Axe-Fx uses a gentler taper so that you can fine-tune the MV easier. So don't be afraid to crank that MV up. When the MV is turned up the virtual power amp works harder which causes the virtual power supply to sag which adds compression which adds feel. It also thickens up the tone when you play harder because the power amp is distorting. You'll get much better results if you learn to find the sweet spot. While playing, turn up the MV until the volume stops getting louder. At this point you are driving the power amp into heavy distortion. Now back off the MV until you get the desired tone and feel. With practice you'll learn to identify how much the power amp is being pushed and where the sweet spot is." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/103152-1-biggest-user-error.html#post1235049)
- "If you want the best tones out of the Axe-Fx you should stop copying photos of knob settings and learn how the knobs work. MV is one of the most important knobs as it controls how hard the virtual power amp is driven. Learning to find the sweet spot is an exercise that will pay handsome dividends. Take the JCM800 model. Set the MV very low, say 1.0. Play for a while. The sound will be harsh and scooped with a stiff feel. Turn the MV up to 5.0. Notice how there is more midrange and a softer high-end response, more compression and a better feel. Turn it up all the way and it will get fuzzy and indistinct. A real amp does the same thing. Learning to dial in the MV is among the most important abilities to harness the most from your Axe-Fx."
- Master Volume: The Master Volume (MV) controls how much signal level is sent to the power amp. Many vintage amps have no MV control and the power amp runs "wide open". Modern amps often get their distortion from the preamp and the Master Volume then allows the user to control the volume of the amp. The Master Volume in the Axe-Fx II, as well as on real amps, is probably the singular most powerful control in the amp block. As the Master Volume is increased the virtual power amp begins to distort. The virtual power amp also begins to sag and all sorts of beautiful magic occurs. The tone becomes more focused, the dynamic response changes, the note attack is accentuated, etc.”
- “MV is the one knob that everyone should master (pun not intended). It sets the amount of power amp drive which, in turn, controls how much current is drawn from the power supply which causes the supply to sag. The trick is to find the

sweet spot. Too much MV and it can get too compressed. To little and the amp will be stiff and too scooped. The MV works just like a real amp's MV except it won't get deafeningly loud if you turn it way up.” source (http://forum.fractalaudio.com/axe-fx-ii-discussion/94957-axe-fx-ii-firmware-17-04-released-5.html#post1138962)

- "With high-gain amps I start with the MV very low, say 2.0. Dial in the tone and gain and then bring the MV up until you hit the sweet spot. Back off the gain a bit if it's too gainy at that point.” source (http://forum.fractalaudio.com/axe-fx-ii-discussion/94957-axe-fx-ii-firmware-17-04-released-6.html#post1139089)
- "One common theme with new users is the "blanket over the sound" complaint. Often this is due to excessive Master Volume values. The Master Volume behaves just like the actual amps. However, unlike an actual amp, if you put it on 5.0 it won't cause your cat to hide for several days. This can cause the user to set the value too high as the physical feedback of painfully loud sound is not present. As you turn up the Master Volume many amps get darker and the bass gets mushy. The key is to find the sweet spot. Do this exercise: Take an amp like the HBE. Set the MV to around 2.5 and turn the Level to a comfortable volume. Turn the Presence up to around 8.0. Copy the settings to "Y" by double-clicking the "Y" button. Now you have the same amp model and values in X and Y. Turn up the MV in Y to, say, 6.0 and lower the Level until the volume is the same. Go back and forth between X and Y and notice how much darker Y is. This occurs because the virtual power amp is distorting, and quite heavily. Due to the impedance curve of the virtual speaker load this causes the bass and high treble frequencies to clip but not the midrange. The result is, naturally, compressed bass and high treble which can sound muddy and indistinct. Modern MV amps are not designed to overdrive the power amp considerably. They are designed to get most of their distortion from the preamp and then adjust the MV until the power amp just starts to clip which is the "sweet spot". Some amps, like the Recto Modern, will distort the power amp at very low MV values, around 2.0. In real life these amps are painfully loud at these settings but in our virtual world we are unaware of this because the Level control allows us to adjust the volume to any arbitrary level. Some modeling products intentionally limit how hard their virtual power amps can be overdriven. Even with the MV on 10 the virtual power amp is not being overdriven that much. Of course this is unrealistic. Our modeling is accurate and with the MV on 10 you will get the same amount of power amp distortion as the real amp when set to 10. With this great power comes great responsibility and that responsibility is understanding how the control works and how to set it properly." source (http://forum.fractalaudio.com/threads/setting-the-master-volume.119903/)
- "A little trick you can do to get the "bounce" of high MV without the muddy bass is to reduce the LF Res value on the Spkr tab. This will reduce the amount of bass clipping in the virtual power amp allowing you to turn the MV up. You can also reduce the HF Res to reduce the amount of treble clipping." source (http://forum.fractalaudio.com/threads/setting-the-master-volume.119903/page-2#post-1427734)
- "At low MV the source resistance into the PI is low which raises the highpass frequency due to the coupling cap and raises the lowpass frequency due to the Miller capacitance and snubber. As you increase the MV the source resistance increases which decreases both of these things. As you keep raising the MV the source resistance then starts to decrease as you get above 50% of the pot value."

source (http://forum.fractalaudio.com/threads/setting-the-master-volume.119903/page-2#post-1428935)

- "The Master Volume behaves just like the actual amps. However, unlike an actual amp, if you put it on 5.0 it won't cause your cat to hide for several days. This can cause the user to set the value too high as the physical feedback of painfully loud sound is not present. As you turn up the Master Volume many amps get darker and the bass gets mushy. The key is to find the sweet spot. Do this exercise: Take an amp like the HBE. Set the MV to around 2.5 and turn the Level to a comfortable volume. Turn the Presence up to around 8.0. Copy the settings to "Y" by double-clicking the "Y" button. Now you have the same amp model and values in X and Y. Turn up the MV in Y to, say, 6.0 and lower the Level until the volume is the same. Go back and forth between X and Y and notice how much darker Y is. This occurs because the virtual power amp is distorting, and quite heavily. Due to the impedance curve of the virtual speaker load this causes the bass and high treble frequencies to clip but not the midrange. The result is, naturally, compressed bass and high treble which can sound muddy and indistinct. Modern MV amps are not designed to overdrive the power amp considerably. They are designed to get most of their distortion from the preamp and then adjust the MV until the power amp just starts to clip which is the "sweet spot". Some amps, like the Recto Modern, will distort the power amp at very low MV values, around 2.0. In real life these amps are painfully loud at these settings but in our virtual world we are unaware of this because the Level control allows us to adjust the volume to any arbitrary level. Some modeling products intentionally limit how hard their virtual power amps can be overdriven. Even with the MV on 10 the virtual power amp is not being overdriven that much. Of course this is unrealistic. Our modeling is accurate and with the MV on 10 you will get the same amount of power amp distortion as the real amp when set to 10. With this great power comes great responsibility and that responsibility is understanding how the control works and how to set it properly." source (http://forum.fractalaudio.com/threads/setting-the-master-volume.119903)

Input Drive and Overdrive:

- Some models have Input Drive and Overdrive controls. Input Drive is the same thing as Drive in earlier firmware versions. If the real amp has two gain controls, the one which is the closest to the 1/4" input is modeled as Input Drive in the model. The other one is Overdrive.
- Firmware 10: The Amp block now differentiates amps that have both Input Drive and Overdrive controls, i.e. Mesa Mark series, Dumble, etc. When a model is selected for amps of this type, the menu shows both controls. For other types the menu shows only the Input Drive control (which was formerly called simply “Drive”). The Overdrive control defaults to noon when amps with this control are selected. As such, any presets based on these amps may need to be updated as this control was not present previously and the amount of drive may differ now. Note that these two controls are applied to the appropriate point in the circuit for the amp being modeled, i.e. for Dumble-style amps the Overdrive is prior to the last triode stage. In Mesa Mark amps the Overdrive is applied prior to the third triode."
- Cliff's comments:
  - "Input Drive increases the gain amount as you rotate the knob clockwise. As the gain increases the tone is shifted from a treble and upper mid emphasis, which

produces an up front sparkling tone, to a lower mid and bass emphasis, which produces a thick meaty tone. Overdrive increases the gain amount as you rotate the knob clockwise but with no alteration of the tonal balance. Different combinations of Input Drive and Overdrive settings will have a dramatic effect on the response of the amplifier and the personality of your instrument. It is easy to get familiar with the action of these controls and you'll be amazed with your ability to make any guitar sound mellow, fat, soulful or aggressive." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/71449-understanding-input-drive-overdrive.html#post875745)

  - "Input Drive is the modeled amp's gain, drive, volume, etc. control. It adjusts the attenuation at the input to the amplifier gain stages after the input buffer. On a Marshall Plexi, for example it is the "Loudness" control. On a typical Fender amp it is the "Volume" control. On many high-gain amps it is called either "Gain" or "Drive". On a real amp this is implemented using a variable resistor called a potentiometer. Many amps include a "bright cap" on the drive control which is a small value capacitor placed across the terminals of the pot that bleeds treble frequencies through as the gain is reduced. Sometimes this bright cap is switchable via a switch on the amp. Sometimes it is fixed."
  - "In a typical amp Input Drive is called various names (Drive, Volume, Gain, etc). It is the knob closest to the input jack. In many cases this potentiometer has a bright cap on it so the frequency response will be dependent on the knob position. In some amps there is also a second drive control. This is your Overdrive knob. It does not have a bright cap so it only affects the gain." source (http://forum.fractalaudio.com/axe-fx-ii-reviews/71497-new-10-10-firmware-thoughts-2.html#post877347)
  - "Some amps possess an attenuation control between the later gain stages. Examples of the are the Mesa/Boogie Mark series, Dumble ODS and others. This control allows the user to vary the gain staging. The Input Drive can be turned up and the Overdrive turned down so that the earlier stages distort more and the later stages distort less and vice-versa."
- The range of the gain taper in the Axe-Fx is 0-10. Volume on Fender amps go from 1 to 10. This translates to (source (http://forum.fractalaudio.com/axe-fx-ii-discussion/84090-thoughts-fw-14-ast-other-clean-ish-amps-4.html#post1024092)):
  - Fender 1 = Axe 0.00
  - Fender 2 = Axe 1.11
  - Fender 3 = Axe 2.22
  - Fender 4 = Axe 3.33
  - Fender 5 = Axe 4.44
  - Fender 6 = Axe 5.55
  - Fender 7 = Axe 6.66
  - Fender 8 = Axe 7.78
  - Fender 9 = Axe 8.89
  - Fender 10 = Axe 10.00
- You can attach a Scene Controller to Input Drive, and then use scenes to vary the amount of amp gain. Note that with some amps models this will temporarily increase CPU load a lot. Cliff: "Depending upon the amp model it can take a lot of CPU to calculate the Input Drive network. Some amps have simple networks that are rapidly solved. Others, like the Hook Lead and Rhythm models have complex networks that require more math. If you attach a modifier to the Input Drive it is constantly

recalculating the network which increases CPU usage."

INPUT TRIM

- This parameter lets you adjust the range of gain of the amp. It's the same thing as the Amp Gain parameter in the Global menu but Input Trim operates per preset.
  - Cliff: "They (Amp Gain and Input Trim) are basically the same thing. The global amp gain has a smaller range as it's designed to be for fine-tuning between guitars whereas the local trim allows you to radically alter the response of the model. The local trim is equivalent to -20 to +20 dB." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/40418-input-trim-global-amp-gain.html#post545862)
- If the real amp has two inputs (f.e. low and high) the model is based on the high input. Set Input Trim to 0.500 (= -6 dB) to get the equivalent of using the low input. source (http://forum.fractalaudio.com/axe-fx-ii-discussion/62915-fender-twin-reverb-drive-low-low-low-but-still-breaking-up-any-advices-4.html#post780229)
- Input Trim at 0.500 = -6 dB.
- Enabling Boost (12dB) is equivalent to setting Input Trim at 4.
- Input Trim can be attached to a controller in the Modifier menu for a variable boost.
- Search this page using the keyword "trim" for information regarding specific amp types.
- Fractal: "You might want to convert the Input Trim parameter to dB if you're used to thinking of it that way. As a rule of thumb, every 2x multiplier = +6dB boost. In other words, Input Trim = 4.0 produces a +12dB boost." Here's a handy calculator (http://www.sengpielaudio.com/calculator-db.htm) " (source (http://forum.fractalaudio.com/axe-fx-ii-wish-list/57127-amp-block-boost-switch-level-control.html#post716617)).
- Cliff's comments:
  - "Input Trim is something you shouldn't play with normally unless you want to deviate from the actual amp. Input Trim allows you to reduce or increase the gain of the virtual amps input buffer. This is analogous to changing the type of tube for V1 in an actual amp. Some people like less gain for V1 so will replace a 12AX7 with a 12AT7. Some people want a little more gain." source (http://forum.fractalaudio.com/axe-fx-ii-reviews/71497-new-10-10-firmware-thoughts-2.html#post877347)
  - "The Axe-Fx II contains a parameter known as "Input Trim". This is just a straight gain control at the very front of the amp block. It has a range of 0.1 to 10.0 (-20 to +20 dB). So what use is a straight gain control at the front? Doesn't the Input Drive do the same thing? The short answer is "no". The long answer is "probably not". On many amps the Drive knob, which may also be called Gain or Volume, has what is known as a "bright cap" across the physical potentiometer. This capacitor shunts high frequencies around the pot so that the Drive control is not a straight gain. It has an associated frequency response. As the Drive is turned down more high frequencies are shunted around the pot which results in a net treble boost. If the Drive is turned all the way down the treble boost is maximum, if it is turned all the way up the treble boost is zero. The roots of the bright cap are due to manufacturers trying to compensate for different types of guitars. Guitars with single coil pickups tend to brighter but with less output. The user would then turn the Drive knob high on the amps. Conversely a guitar with humbuckers has more

output but sounds darker. To compensate the user would typically turn the Drive down. This will result in a treble boost compensating for the darker response. The Input Trim control allows one to fine-tune the amount of treble boost first and then adjust the amount of distortion. So it is probably more correct to think of the Drive control as a combination Drive/Treble control. With this in mind experiment with the Drive control combined with the Input Trim. Indeed some manufacturers have actually implemented separate Drive and Trim controls on their amplifiers. For example the Fryette (VHT) Deliverance has two controls: a Gain knob and a Cut knob. The Gain knob has a bright cap across it while the Cut knob is just a straight volume adjustment. The purpose of these two knobs is exactly as described above." source (http://forum.fractalaudio.com/cliffs-notes/81116-input-trim.html#post985500)

- "The Input Trim control allows you to adjust the input attenuation without changing the frequency response. If you turn down the Input Drive and the model has a bright cap the amp will get brighter. Now you may like the brighter tone but wish there were more gain. Input Trim allows you to increase the gain without changing the tone. Conversely you may like the darker tone with Input Drive set high but wish there were less gain. In this case you can lower Input Trim. Most real amps do not possess an Input Trim control. Instead they usually have a switch or two input jacks that select between a high-gain and low-gain input. Almost invariably the difference between these two jacks is 6 dB. All the Axe-Fx amps are modeled using the high-gain input or switch position (if any). To simulate the low-gain input set the Input Trim to 0.5 which is 6 dB less". source (http://forum.fractalaudio.com/tech-notes/95018-understanding-all-different-gain-controls.html)
- "It controls the voltage divider into V1. Many amps, for example, have two 68K resistors feeding V1. If you plug into the High input the resistors are in parallel and the gain is 1.0. If you plug into the Low input the resistors are in series and the gain is 1/2. Input Trim is a more flexible way of accomplishing the same thing but without being constrained to only two gain values."
- "All the models assume the "Hi" input on the amp was used if there are multiple inputs. If the amp has a Lo input this is typically half the sensitivity so you would set Input Trim to 0.5 to replicate. The beauty of Input Trim is you can set it to any value you like rather than being stuck with a switch with only two values." source (http://forum.fractalaudio.com/threads/ax8-clean-tones-input-trim-is-the-key.120261/#post-1431111)

BASS, MID, TREBLE

- These are the usual tone controls: Bass, Middle and Treble. The exact behavior (range) depends on the amp model.
- "In most cases the knobs do translate. Usually within 10%." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/58613-axe-sounding-terrible-through-guitar-cab.html#post732551) More information. (http://wiki.fractalaudio.com/axefx2/index.php?title=AMP_block_parameters#AMP_controls_behavior)
- If you don't like the effect of high gain on your low notes, turn down Bass, or increase Low Cut Freq, and then increase the low frequency level in the Amp's GEQ.
- Cliff's comments:
  - "If an amp has just "Tone" then that's mapped to Treble. Leave bass and mid at

noon." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/67260-question-proper-axe-eq-values-amps-dont-have-them.html)
  - "The tapers match my amps. However manufacturers are notorious for changing tapers, sometimes right in the middle of a production run due to part availability. Furthermore the tapers in the Axe-Fx assume "true" logarithmic pots. Consumer-grade log pots are not true logarithmic, they're a crude approximation. At noon on a pot you'll get a nearly perfect match assuming the pot has 0% tolerance. As you deviate from noon there may be some error due to the approximation in the actual amp. As you get to the ends of the travel the error will decrease to zero. At any point there shouldn't be more than 10% or so deviation between the Axe-Fx knob position and the real amp." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/80497-axe-fx-ii-firmware-version-12-04-public-beta-12.html#post978667)
  - (firmware 18) "Drive, Bass, Mid and Treble will match the actual amp within the tolerance of the pots. Another caveat when comparing amps: many times the knobs aren't "centered". IOW if you put the Treble knob at noon it isn't actually at 50%. You can see this by turning the knob all the way down and all the way up. It may not be symmetrical. This happens when the pots don't have a flat spot and/or the pot is rotated within the mounting hole." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/96570-new-18-00-public-beta-5.html#post1158815)

BOOST

- The Boost switch increases signal at the input of the Amp block with 12 dB.
- Sometimes enabling the Boost switch works better than turning up preamp gain. It's a clean boost so it'll increase the gain across the entire frequency spectrum. BOOST is modifiable so it can be activated remotely.
- Enabling Boost is the same as setting Input Trim at 4.
- Boost can be assigned to an external controller.
- Want an adjustable Boost: use a NullFilter block instead.

BRIGHT switch

- Bright switch, located under the TREB knob.
- The Bright *switch* is not to be confused with the BRIGHT *knob* which is shelving filter control between the preamp and power amp (see below).
- The bright cap in real amps is used to compensate for guitars with weak pickup loads. Treble increases as gain is decreased. It changes the mids which affects gain too.
- Turning up the amp's Drive or Master may decrease the impact of the Bright switch, depending on the amp type.
- It's possible to change the effect of the Bright switch on the tone by adjusting the Bright Cap value.
- In some amp types, such as Plexi, the Bright switch is set at a very high value and has a large impact on the amount of gain.
- Cliff's comments:
  - "If an amp doesn't have a bright switch, the operation of of the models' bright switch is undefined. I chose what I considered a reasonable value for the bright cap but if that doesn't satisfy the user then they are free to change it to a different

value." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/70382-vh4-amps-sat-switch.html#post865222)
    - "The Bright switch always controls the bright cap on the input volume". source (http://forum.fractalaudio.com/axe-fx-ii-discussion/74295-any-way-disable-bright-cap-dumble.html#post909499)
    - "The Bright switch switches in/out one or more capacitors on the Input Drive network." source (http://forum.fractalaudio.com/threads/always-look-on-the-bright-side-of-life.110008/#post-1317029)
    - "The Bright Switch models the "bright cap" across the drive/gain/volume /comodjulator pot. Some amps have no bright cap, some amps have a hardwired bright cap and some amps give you a switch to turn it on/off. A bright cap increases treble as the pot is turned down. The original impetus for this was that guitars with weak pickups tend to be brighter and guitars with hot pickups tend to be darker. So you're likely to turn the gain up for the guitar with weak pickups which reduces treble response. On the guitar with hot pickups you're likely to turn the gain down which increases treble response to counteract the darker tone. If the amp has no bright cap the Bright Switch defaults to off. If the amp has a hardwired cap the switch defaults to on. If the amp has a switch it defaults to whatever we felt sounds the best. Our particular JCM800 reference amp has no bright cap because someone removed it. However the model has the switch on because that's the way the amp would've come from the factory. There are actually 11 different (IIRC) bright cap models in the Axe-Fx/AX-8. The earliest bright cap circuits were just a cap from the input terminal to the wiper. Over time designers have developed more complex circuits with resistors in series with the cap, another cap from the wiper to ground, treble peakers before/after the pot, etc. The user doesn't have the ability to select the model though, it is hard-coded into each model."source (http://forum.fractalaudio.com/threads/fractal-vs-helix-let-the-comparisons-begin-video-series.119390/page-2#post-1423491)

BRIGHT knob

- This high treble control is a shelving filter between the preamp and power amp and may be used to darken or brighten the output of the preamp. This control replicates the “Presence” control found in the Mesa Triaxis preamp when set to negative values (the Presence control in the Triaxis is actually a high frequency cut shelving filter).
- This is not to be confused with the Bright switch (see above), which engages/disengages a capacitor across the drive pot.
- Cliff's comments:
    - "The Bright Knob is an active fourth tone control at high frequencies. Think of it as "High Treble". You can use it to add a little zing to a preset or remove harsh high frequencies. You can also use it to simulate the behavior of the Presence control on a Triaxis (which is really just a high cut). Turn it down to simulate "Presence" settings less than 10." source (http://forum.fractalaudio.com/threads /always-look-on-the-bright-side-of-life.110008/#post-1317029)

CUT

- Located under the Bass knob.

- This reduces the amount of low frequencies into the amp simulation. This can be used to achieve a “tighter” tone or to reduce low-end “flub”.

Cut is similar to increasing Low Cut Frequency but still retains some low end so it doesn’t get thin.

- Cut is a first-order shelving filter (high-pass) at 120 Hz. Fractal: "You can use a Filter block before the Amp set to Shelving if you want to add more flexibility to what the Cut switch is doing in the Amp block." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/72470-new-cut-parameter.html#post887770)
- Cliff's comments:
  - "120 Hz is where most amp designers put it. A typical cathode bypass has the pole at approx. 85 Hz. Assuming 6 dB gain reduction that puts the center frequency at 120 Hz." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/72470-new-cut-parameter.html#post887765)
  - "The bass cut switch is before the distortion so it will change the feel and breakup characteristics. The bass cut is basically intended to give you that Tube Screamer with Drive on 0 sound without having to use a dedicated Drive block." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/74445-hi-cut-cab-bass-cut-amp-%3D-overkill.html#post910785)
  - "Cut engages a lowshelf filter at the input. This would be analogous to partially bypassing the input buffer cathode on a tube amp." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/96327-controlling-lows.html#post1154831)

FAT

- Located under the Mid knob.
- When engaged it shifts the center frequency of the tone stack, “fattening” the tone. It's similar to the FAT switch on a Mesa Boogie or Bogner amp.
- Cliff's comments:
  - "The Fat switch simply alters the tone stack treble capacitor. So the effect depends on the location of the tone stack." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/51361-fat-switch-fiasco-2.html#post660652)
  - "The Fat Switch multiplies the tone stack treble cap by four. Depending upon the type of tone stack, tone control settings, position, etc., etc. the effect can be more or less noticeable." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/76310-fat-switch.html#post932073)

DEPTH

- Depth is a low frequency tone control for the power amp section.
- Cliff's comments:
  - "Depth = Resonance = "Whomp" = whatever colloquialism the manufacturer can think of. Depth differs from Bass in that it is applied in the power amp as opposed to the preamp. It is done by modifying the feedback network. Less lows are fed back thereby increasing bass response in the power amp. It is analogous to Presence except it affect bass instead of treble. If you look at the Axe-Fx II's menu you'll see the tab says "PWR AMP" thereby indicating it is a power amp control."

    source (http://forum.fractalaudio.com/axe-fx-ii-discussion/79186-amp-5150-depth-control-what.html#post963090)
  - "Many amps have no depth circuit, e.g. Fenders, most Marshalls, and generally most older designs. In these cases the Depth knob will default to zero indicating the amp has no depth circuit. Some amps have a fixed depth circuit, e.g. 5153, Freidman BE/HBE, Dirty Shirley, TripTik, Tucana, et. al. In these cases the Depth knob will default to a value that corresponds to the fixed circuit. Finally some amps have a variable depth circuit, e.g. 5150, Diezel, et. al. In these cases the Depth knob defaults to a non-zero value that I think sounds good but that's just my taste. The choice of IR and the monitoring system can greatly influence the amount of perceived bass. The desired amount of bass is a preference. If you are into "classic" tones then a Depth of zero would be a logical choice." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/88747-lets-talk-about-depth-control.html#post1071598)
  - "Depth does not work at a Damping of 0 since Depth modifies the feedback and there is no feedback." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/59333-damping.html#post740665)
  - "The effect of the Presence and Depth is consistent with the behavior of the real amp and depends on the amount of negative feedback. As you decrease negative feedback the presence and depth controls have less effect (as in a real amp). Also, as you increase Master Volume the presence and depth may appear to be less effective (key word is "appear") as the power amp distorts more and this masks the effect of the controls." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/87420-may-stupid-questions-fw15.html#post1058088)
  - "I look at the Depth knob as a starting point. Just because the 5153 has a fixed depth circuit doesn't mean you can't adjust the knob. The only reason that it is fixed on that amp is there wasn't enough room to put in separate knobs for each channel. The 50W version has an adjustable Depth (it's called resonance and the knob is on the back). The original 5150 had adjustable Depth. The designers choice isn't necessarily best. It depends on the cabinet (or IR) and your personal preferences. Short answer: use your ears. I think too many people are scared to use their ears. But if you constantly rely on the ears of others you'll never learn how to create your own signature sound." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/88747-lets-talk-about-depth-control.html#post1071786)
  - "Presence and Depth controls may not match the taper of the actual amp. On most amps the Presence control does nothing until you turn it almost all the way up. This seems a bit silly so we make the Presence behave more logically. Same goes for the Depth control." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/96570-new-18-00-public-beta-5.html#post1158815)
- MIMIC whitepaper (http://www.fractalaudio.com/downloads/manuals/axe-fx-2/Fractal-Audio-Systems-MIMIC-(tm)-Technology.pdf) : "Note however that the taper of the presence (and depth) control can deviate from the actual amp. In our tests we found that the presence control on many amps did nothing for the first 80% of its rotation and all the action occurred in the last 20%. We feel that this design anomaly is undesirable and therefore did not model that aspect."
- Firmware 15: "Improved Amp block feedback network accuracy especially for those amps that have depth networks. This causes the Presence and Depth controls to interact (as they would on a real amp) but yields greater realism."

PRESENCE / HI CUT

- Presence is a high frequency tone control for the power amp section.
- If Negative Feedback (Damping) is zero, Presence turns into a HiCut control.
- If Power amp modeling is off, Presence turns into a shelving filter where "5" is neutral.
- The range (min/max values) of the virtual Presence knob in the model is the same as the range of the real Presence knob on the modeled amp. But the visual position of both knobs does not necessarily match, because the taper is not the same.
- Cliff's comments:
  - "The Axe-Fx II presence operates just like a real amp and modifies the virtual power amp feedback. This actually does create a sort of "magic" since it changes the shape of the distortion vs. frequency. That's the advantage to the nonlinear feedback network that the Axe-Fx uses. Negative feedback makes the distortion transfer function "harder". The presence control reduces negative feedback at high frequencies. This increases the treble but also softens the transfer function so you get more highs but the softer distortion reduces the amount of harsh overtones." source (http://forum.fractalaudio.com/axe-fx-ii-recordings/85267-slo-100-models-axefx-amplitube.html#post1033870)
  - "The effect of Presence and Depth is consistent with the behavior of the real amp and depends on the amount of negative feedback. As you decrease negative feedback the presence and depth controls have less effect (as in a real amp). Also, as you increase Master Volume the presence and depth may appear to be less effective (key word is "appear") as the power amp distorts more and this masks the effect of the controls." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/87420-may-stupid-questions-fw15.html#post1058088)
  - "A tube amp's presence control is basically a type of treble control. It affects a higher range of frequencies and operates on a different principle but the net effect is an increase in high frequencies. There is also a slight increase in distortion in the higher frequencies since the power amp becomes less linear for those frequencies". source (http://forum.fractalaudio.com/axe-fx-ii-wish-list/40491-global-power-amp-presence-value-independent-amp-sim.html#post546376)
  - "HiCut is dependent upon Damping, just like a real amp. Hi Cut is modeling the Miller capacitance at the input to the Phase Inverter. The more negative feedback, the less the Miller capacitance." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/47260-adv-hi-cut.html#post615856)
  - (firmware 18): "Presence and Depth controls may not match the taper of the actual amp. On most amps the Presence control does nothing until you turn it almost all the way up. This seems a bit silly so we make the Presence behave more logically. Same goes for the Depth control." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/96570-new-18-00-public-beta-5.html#post1158815)
- The Presence control in the Amp block behaves like the actual amp rather than an idealized version. The Presence Frequency parameter is now a frequency multiplier rather than an absolute frequency as the frequency of the presence circuit depends on the Presence control position. The Presence Frequency parameter works by scaling the value of the virtual presence circuit’s capacitor value. Setting the Pres/Depth Type parameter to Active or Active Pres will override the authentic modeling and implement an ideal presence circuit with fixed center frequency.
- Presence control is set to a default value when an amp model is selected. This is done because many amps, i.e. Double Verb, Deluxe Verb, et. al., have no presence control

and the value should be set to zero for best accuracy. On the other hand some amps, i.e. Jr. Blues, 65 Bassman, et. al, have fixed presence networks. The Presence control will default to the appropriate value for these amps. For amps that do have a presence control the Presence parameter will default to a value that is deemed typical for the model.

- MIMIC whitepaper (http://www.fractalaudio.com/downloads/manuals/axe-fx-2/Fractal-Audio-Systems-MIMIC-(tm)-Technology.pdf) : "Note however that the taper of the presence (and depth) control can deviate from the actual amp. In our tests we found that the presence control on many amps did nothing for the first 80% of its rotation and all the action occurred in the last 20%. We feel that this design anomaly is undesirable and therefore did not model that aspect."
- When switching off power amp modeling, check the Presence setting. Cliff: "If you turn off power amp modeling always check the presence control. It changes from a "classic" control to a shelving type where 5.00 is neutral. I just spent an hour trying to figure out why this preamp model I am working on wasn't matching. Forgot to set the presence control to 5.00." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/45574-one-thing-always-check.html#post595595)
- Firmware 15: "Improved Amp block feedback network accuracy especially for those amps that have depth networks. This causes the Presence and Depth controls to interact (as they would on a real amp) but yields greater realism."
- With some USA amp models, the Presence knob turns into a "Presence Shift" button.

Other parameters (alphabetic)

AC LINE FREQUENCY

- See PWR SUPPLY TYPE.

AC VOLTAGE (VARIAC)

- This parameter sets the relative AC line voltage into the amp simulation implementing a virtual “Variac”. Note that normally the volume would vary with the Variac setting in a real amp. The simulation compensates for the volume change by applying the inverse. Some additional level correction may be required.
- A Variac is said to be required to get the Brown sound (EVH). Try it around 75%.
- Cliff's comments:
  - "The Spongy/Bold switch (on a Mesa) is basically a Variac." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/102553-rectifier-solidstate-vs-tube.html#post1228691)
  - "It's impossible to compensate exactly. Some manual adjustment of the volume may still be required." source (http://forum.fractalaudio.com/threads/volume-changes-with-variac-setting.109951/#post-1316615)

B+ TIME CONSTANT

- Firmware 10: "High values of Sag along with low B+ Time Constant values can cause “ghost notes” when the supply type is AC (as in a real amp). Lower B+ Time Constant

values will make the amp feel “faster” but too low can cause ghost notes."
- Cliff's comments:
  - "B+ Time Constant is the time constant associated with the Supply Sag parameter. The power tubes draw current from the supply. The supply has a finite resistance. As the power tubes draw more current the supply voltage droops. The rate of change of the droop and recovery is dictated by the supply capacitance. The product of the resistance and capacitance is the time constant. It's typically around 10 ms. You can vary this using the B+ Time Constant parameter. It is not a simply compression though. As the supply sags, the headroom is reduced but many other things happen. One thing that happens is that the screen voltage droops. The screen voltage is derived from the B+. However the screen has it's own dynamic response, which is often 2nd-order since there is often a filter choke. If you listen carefully to the models with a filter choke you can hear the screen voltage "bounce" when you hit a power chord. The damping of the screen filter is not exposed to the user. When the screen voltage droops, the power tube gain decreases. It effectively shifts the bias point. There is quiescent draw from the supply as well. As you increase the bias (Power Tube Bias) the quiescent draw increases which decreases available headroom. The Axe-Fx II does not model all this stuff with compressors, like other products do. It actually uses a differential equation for the supply and the current from the power tubes. It then solves the equation at each sample instant to find the supply voltage and screen voltage." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/46615-b-time-constant.html)
  - "The effect of lower B+ is equivalent to increasing Transformer Match. A lower B+ means the plates clip sooner which is the same as increasing the turns ratio on the transformer. This is assuming that you rebias since typically lower the B+ affects the bias." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/60172-there-parameter-manipulate-b-voltage-video-inside.html#post751161)
  - "The attack comes first as things compress. Then the supply bounces due to a much slower time constant. Compression time constants are in the neighborhood of a few ms, B+ time constants are typically 10-20 ms. You can adjust these on the Advanced page. Many amp designers like a "fast" power supply but if you lower the B+ time constant too much you get ghost notes." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/103735-axe-fx-ii-quantum-rev-1-00-public-beta-39.html#post1245408)
- "The higher the value the stiffer the power supply. This will result in less compression from the power amp. "Most" guitar players prefer a value around 10ms as it it accentuates the attack without excessive ghosting. It seems as though you prefer less attack so raising the value will accomplish that. Another option is to change the supply type to DC. This will eliminate any ghosting and give you a more "ideal" response. Most modelers use a DC power supply model but I've found that an AC supply model is key to achieving that last few percent of realism. The supply ripple is a big part of why old amps sound the way they do. Most guitar players actually like a percussive attack so that's why about 10ms has become a de-facto standard." source (http://forum.fractalaudio.com/threads/i-had-a-b-time-constant-epiphany.106309/#post-1271768)
- Quantum 2.0 release notes: "For convenience the virtual power supply voltage (B+) can now be monitored on the PWR DYN tab of the amp block. When the Supply Sag control is selected the gain reduction meter will display the supply voltage in dB

relative to the idle voltage."
- Quantum 2.0) "B+ Time Constant controls the capacitance of the virtual power supply. The more capacitance the "slower" the supply and vice-versa. Most guitar players like a fast supply but too fast will cause excessive AC ripple and create ghost notes (although I think a little ghost note is cool). When the supply is fast it will sag rapidly accentuating the pick attack and compressing after. This parameter works in conjunction with Supply Sag parameter. The time constant remains constant so if you increase Supply Sag the virtual capacitance decreases." source (http://forum.fractalaudio.com/threads/quantum-2-00-tips-part-1.110275/)

BIAS EXCURSION

- Cliff's comments:
  - "Bias excursion occurs because the power tube grids forward conduct when the grid voltage is slightly greater than the cathode voltage. Now this isn't a problem by itself. However almost all tube amp designs use a capacitor coupled grid circuit. The phase inverter is coupled to the power tube grids via a capacitor. When the grid voltage exceeds the cathode voltage, which is typically zero volts in a fixed-bias topology, the grid will become forward-biased and look like a low resistance. This clamps the grid side of the coupling capacitor. This occurs when the phase inverter signal is large and swings toward the B+ supply. When the phase inverter signal swings the opposite direction the grid stops conducting and the capacitor is no longer clamped. However there is now excess charge on the capacitor. During the time the capacitor was clamped charge was building up on the phase inverter side. When the grid comes out of conduction that charge effectively reduces the power tube bias. For example, a typical 6L6 is biased around -50V. The clamping action would then push the bias voltage even more negative, say -75V. In some designs the the bias voltage can be reduced by nearly 100%! Since the bias voltage is shifted the phenomenon is referred to as "bias excursion". Like cathode squish, bias excursion pushes the power amp from Class-AB operation towards Class-B operation. As we know Class-B operation has lots of crossover distortion. Now this may seem bad but, in fact, there are positive attributes associated with bias excursion. When designed correctly bias excursion can actually help an amp sound more "open". This happens because as the bias shifts the gain of the power tubes decreases. This in turn prevents the power tube plates from saturating as easily. However too much bias excursion leads to what is referred to as "blocking distortion" which can make an amp sound farty and generally unpleasant. Blocking distortion occurs when the bias shifts so much that the tubes are basically shut off for a period of time. If the capacitor charges up rapidly but bleeds off slowly, combined with lots of excursion, this leads to blocking distortion. There are three associated parameters in the Axe-Fx II that allow you to alter the bias excursion behavior: Bias Excursion, Excursion Time and Recovery Time. Bias Excursion controls the amount of bias excursion. The higher the value the more the bias shifts when the virtual power tubes are overdriven. Excursion Time controls how rapidly the coupling capacitor charges when the virtual power tube grids are conducting. Recovery Time adjusts how quickly the excess charge bleeds off when the virtual grids are not conducting. Preamp tubes also exhibit bias excursion and too much of it can cause blocking distortion. Like power tube bias excursion, though, a little bit can help. The trick is getting the

right amount." source (http://forum.fractalaudio.com/tech-notes/99698-bias-excusion.html)

BRIGHT CAP

- See BRT (Bright switch).

CF (PREAMP) COMP, CF PREAMP TIME, CF PREAMP RATIO, CF PREAMP HARDNESS

- See PREAMP COMP.

CHARACTER TYPE, CHARACTER FREQUENCY, CHARACTER Q, CHARACTER AMOUNT

- These parameters control extremely powerful “inverse homomorphic filters”. When playing softly these dynamic filters have little effect on the sound. As the amount of distortion increases, the influence of these filters increases. The Character Frequency control sets the center frequency of the filters while the Character control sets how pronounced the effect is. For example, to darken the tone when playing harder, one might set the frequency to 10 kHz and the amount to -5. Setting the amount to +5 will make the tone brighter when playing hard. The amount defaults to zero whenever an amp type is selected.
- This control is similar to Dynamic Presence and Dynamic Depth but the frequency is adjustable.
- Cliff's comments:
  - "The "Character" parameters are two of the most powerful advanced parameters available but I bet almost no one uses them. My secret formula: Character Frequency: 3000 - 5000 Hz, adjust to taste, Character: -0.5 to -1.0, adjust to taste." More information (http://forum.fractalaudio.com/axe-fx-ii-discussion/85426-my-secret-formula.html#post1035708)
  - "It is highly dependent on the amount of gain. This formula is designed for an "80's" lead tone. I use on for my JCM800 preset because I find JCM800s get shrill as you turn the gain up. It also works well with the SLO 100 and Recto models. The Character parameters control an "inverse homomorphic filter" which is a term I coined to describe a type of homomorphic signal processing. This filter is distortion dependent. The more distortion there is the more pronounced the effect of the filter. It's analogous to contrast and edge detection in image processing. The processing is dependent on the dynamic range of the image." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/85426-my-secret-formula-2.html#post1036134)
- Firmware Quantum 1.00: "Added a new mode to the “Character” controls in the Amp block. A Char Type of “Dynamic” engages an exciting new mode of tone control. This can be used to fatten or scoop the tone as a function of picking strength. For example, set the Type to Dynamic, Char Freq to 450.0, Char Q to 0.7 and Char Amt to 4.0. This will cause the tone to get fatter and thicker as you play hard but without getting honky when playing soft. "

CRUNCH

- Introduced in firmware 18.

DEFINITION

- This parameter allows changing the fundamental character of the amp from vintage to modern or vice-versa. Positive values increase the amount of upper overtone saturation whilst negative values reinforce lower harmonics. It's a treble boost/cut ("Tilt EQ").

DYNAMIC DAMPING

- Firmware 13:
  - "Improved power amp modeling via improved modeling of the plate impedance of the power tubes. This gives tighter bass (less flub) and warmer highs when the virtual power amp is heavily driven (higher Master Volume settings). This also improves the feel and dramatically increases the “3-dimensionality” of the tone. The plate characteristics are adjustable via the new Dynamic Damping parameter. This parameter defaults to the appropriate value when an amp model or power tube type is selected."
  - "The power tube type presets the Dynamic Damping parameter as well as several internal parameters."

DYNAMIC PRESENCE, DYNAMIC DEPTH

- Dynamic Presence:
  - "This models the output transformer leakage inductance that results in a brightening of the tone when the power amp is pushed. This control is set to a default value when the model is selected corresponding to the real amp, if applicable. Increasing this value results in a brighter response as the virtual power amp is pushed. When playing softly or at lower gains, the influence of this control is lessened. Note that this only affects the power amp modeling and is dependent on the degree of power amp overdrive. This control can also be set negative to cause the tone to darken when playing hard. This control can also be used to help “dial in” the sweet spot of an amp model. As the MV is increased an amp becomes more liquid, compressed and easier to play. However, the highs may get overly compressed causing the amp to sound too dark. The Dynamic Presence control allows you to get the desired power amp drive and liquid feeling and then bring the highs back without affecting the rest of the spectrum."
  - Another way to look at it: a distortion-dependent treble filter.
  - Cliff: "Transformer Grind is what you want to get that top-end sizzle. Dynamic Presence is one of my "Inverse Homomorphic" filters and only approximates the dynamic presence boost found in some amps. Transformer Grind is an authentic model of what actually happens in those amps." source (http://forum.fractalaudio.com/threads/q3-01-to-2-04-conversion.114090/#post-1364689)
- Dynamic Depth:

- "Analogous to the Dynamic Presence control, this increases or decreases low frequencies when the virtual power amp is being pushed. While real amps don’t display this behavior, it is a valuable tone-shaping tool."
- Another way to look at it: a distortion-dependent bass filter.
- Firmware 13: "Changed Amp block Dynamic Depth behavior so that frequency of action is set by the Depth Freq parameter rather than fixed."

EQ, EQ TYPE, EQ LOCATION

- Press Enter to reset all sliders to 0 dB.
- Firmware:
  - 11: "Added EQ Type parameter to Amp block. This allows selecting between an 8-band, 7-band or 5-band EQ. The 7-band and 8-band types emulate popular graphic EQ pedals. The 5-band type emulates the response of the on-board EQ in the Mesa Boogie Mark series amplifiers. Note that 5- and 7-band types are non-constant-Q designs whereas the other types are constant-Q designs. When selecting amp models based on Mesa amps the type automatically changes to 5-band."
  - 15: "Added Variable-Q EQ types to Graphic EQ, Filter and Amp blocks. Many “classic” graphic equalizers use variable-Q designs which may be more familiar to some users as opposed to constant-Q filters. In the Filter block this type is selected by choosing the “Peaking2” type. The Graphic EQ block now has four constant-Q modes and four variable-Q modes. The Amp block now has three constant-Q modes and three variable-Q modes."
  - 15.04: "Added various “Passive EQ” types to Graphic EQ and Amp blocks. These EQ types are modeled after classic analog EQs and specifically tuned for guitar amp equalization."
  - 17:
    - "Added “3 Band Console” types to Graphic EQ and Amp blocks."
    - "The EQ page of the Amp block now supports changing the EQ Type using the Up/Down Nav buttons. The type of EQ will be briefly displayed after it has been changed. The type is also briefly displayed when first switching to that page."
  - 19: "Added EQ Location parameter to Amp block. The sets the position of the graphic EQ. The default value of “Post P.A.” places the EQ at the output of the power amp. “Pre P.A.” places the EQ between the preamp and power amp."
- EQ TYPE selects the type of EQ.
- Firmware 19: "Added EQ Location parameter to Amp block. The sets the position of the graphic EQ. The default value of “Post P.A.” places the EQ at the output of the power amp. “Pre P.A.” places the EQ between the preamp and power amp. The graphic EQ in the Amp block is fixed in position at the output of the power amp section."

HARMONICS

- Introduced in Quantum 2. Do NOT use with Modeling set to firmware before Quantum 2.
- It controls the amount of interaction. Higher values yield softer distortion.

HIGH FREQUENCY SLOPE

- This parameter allows fine adjustment of the high-frequency impedance of the virtual voice coil (which affects the slope of the impedance curve). A speaker voice coil is “semi-inductive” due to eddy current losses in the motor. This presents an impedance to the power amp that isn’t fully inductive nor fully resistive. The amount of resistive loss varies by brand and type. Reducing the Slope simulates a speaker that is less inductive, increasing Slope simulates a speaker that is more inductive. Typical speakers range from 3.0 to 4.5 with the median being about 3.7. Lower values yield greater midrange while higher values are more scooped and sizzly.

INPUT SELECT

- Selects which side of the incoming signal should be processed by the Amp block.
- Use this parameter with two Amp blocks in conjunction with Left and Right Cab blocks, when using a single preset for two separate guitar. Or for stereo separation when running dual Amp blocks with a single guitar. Or to keep the stereo spread of a stereo delay before the amps.

LOW CUT FREQUENCY, HI CUT FREQUENCY

- Low Cut Freq controls the amount of lows the amp model sees. It's a blocking filter at the input (before distortion). Ranges from 10-1000Hz, with the lowest setting basically letting all the lows you feed it in. The main practical use for this is to tighten up a tubby bass end. Somewhere between 10-150Hz is generally where it will sound best for standard guitar tones.
- Hi Cut Freq is a low-pass 2nd order filter positioned at the end of the preamp section that will chop all frequencies above the value you select. Ranges from 2000-40000Hz. This will make your top end sound smooth and silky, the lower the value, brilliant and defined, the higher the value. Try changing values from stock when you want to fine tune a sound.
- Cliff's comments:
  - "Low Cut Freq sets the -3dB point of a highpass filter at the input to the preamp. The default value is usually due to the coupling cap between the input buffer and first triode stage (but not always)." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/96327-controlling-lows.html#post1154831)
  - "Low Cut Freq controls a 1st-order highpass before the first gain stage." When using a Filter block instead: "The filter block defaults to 2nd-order so the slope will be steeper." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/97963-low-cut-freq-tone-amp-page.html#post1175769)
  - (Quantum 3.00) "It's probably inaudible but there were a few amp models where the matching was indicating a loss of high-frequency response. This was traced to the high-cut filter. When the high-cut frequency is 20 kHz that means the response is 3 dB down at 20 kHz so you've still got some slight attenuation at, say, 15 kHz. So for the sake or accuracy it now goes to 40 kHz which pushes that pole well outside the audible range." source (http://forum.fractalaudio.com/threads/axe-fx-ii-quantum-rev-3-00-public-beta-2.113734/page-3#post-1360576)

LOW/HIGH RESONANCE FREQUENCY, LOW/HIGH RESONANCE Q, LOW/HI RESONANCE (SPEAKER IMPEDANCE)

- "The speaker tab is not an EQ. It allows you to adjust the impedance that the virtual speaker presents to the virtual power tubes. In most cases the resulting EQ is quite different than the impedance curve since negative feedback flattens the response. If you turn the damping all the way down then the EQ will be close to the impedance curve (but still influenced by the transformer)." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/47917-amp-speaker-tab-versus-amp-graphic-eq.html#post623896)
- "There are certain aspects that simply can't be modeled and require user intervention. For example, a speaker has a low-frequency resonance. A tube amp will create a higher output at that resonant frequency. The Axe-Fx has no way of knowing what that resonant frequency is and defaults to a value that is common for the speakers that are typically used with that amp. However, if you drive that speaker through a solid-state amp you won't excite the resonance unless you adjust the Speaker Resonant Frequency to match it."
- "**One way to find the SRF is to put a Filter block after the amp block. Set the type to Peaking, Q to 5 or so and Gain to 10 dB. Start with a Freq. of 50 Hz. Play some chugga-chugga and slowly adjust the Freq. until you hear and feel the cabinet resonate. Make a note of the frequency. Remove the filter block and set the amp block SRF to match. 4x12s typically have an SRF of between 80 and 120. Open back cabs are typically a bit lower**."
- "LF: in general the Q is between 2 and 2.5. The Hi Freq is usually between 1 and 1.5 kHz. Hi Freq sets the critical frequency (or corner frequency) of the inductive portion of the loudspeaker's response. The critical frequency is the frequency at which the reactive component of the impedance is equal to the resistive component. This is found by fc = R/(2*pi*L). For a typical speaker R is around 6 ohms and L is around 0.75 mH. Therefore fc = 1270. Jensens tend to have higher inductance so that would move this value down. Eminence speakers tend to have lower inductance so that would move this value up. Celestion does not publish their values so I used Eminence values when calculating the defaults. You'll notice the Marshally stuff has fc around 1500 which is consistent with a typical Eminence copy of a Greenback. => You cannot obtain speaker impedance via audio stimulus and microphone measurement. Impedance is defined as voltage divided by current so you need to measure the current vs. applied voltage across the frequency range of interest. I have the equipment to do it, and have measured many speakers, but the average person do esn't have the equipment nor the knowledge to use the data. The influence of speaker impedance is generally not that great. The exception are amps with no negative feedback. In these cases the speaker impedance has a much more pronounced effect on the overall response. These amps include Vox, Matchless and most other "Class-A" designs. As soon as you add negative feedback the response flattens considerably. However... Presence and Depth reduce negative feedback so if you dial significant amounts of those in then the speaker impedance becomes a factor again. All-in-all you only have to be in the ballpark. 1500 Hz is a good starting point for Hi Freq. Adjust up or down slightly by ear. I don't believe that 3000 Hz is accurate. I've never seen a speaker that would have the corner frequency that far out. => As I explained a few posts up I wouldn't set Hi Freq outside the range of 1.0 to 1.6 kHz. Vibroverb model is an exception (800 Hz) since it had a

more voice coil inductance. => I call it critical frequency since it is similar to the critical or corner frequency of a filter. I had to come up with some way of setting the loudspeaker inductance relative to the resistance. Frequency seemed to make more sense. I thought about an inductance parameter but figured that would be too nebulous. At the default settings the impedance rise of the simulated voice coil matches very close with published data. I have overlaid the modeled impedance curve with published data and it is a very good fit. For example, take the JCM800 model. The graph on the SPKR page has a scale of +20 dB at the top. Look at the response at 2kHz. It's roughly 1/4 of full-scale which equates to 5 dB. If we look at the impedance curve for a typical 8-ohm speaker we see that the impedance at 2 kHz is roughly 13 ohms. For a 6.5 ohm voice coil (typical) this means that the voltage at the speaker is 6 dB higher at 2 kHz. Pretty darn close to what the graph is showing. While there is no high-frequency resonance in the speaker itself, a resonance IS formed due to the winding capacitance of the transformer. This capacitance resonates with the voice-coil inductance. => The negative feedback is set in the Advanced menu. The SPKR page only sets the impedance curve of the speaker/OT combo. The values chosen are prototypical for the speaker used with the modeled amp. You should not need to vary these parameters much IMO. I only ever vary Low Freq and High Freq. Whenever I'm matching an amp I adjust Low Freq to match the resonance of my reference cabinet. I occasionally vary Hi Freq to get more or less midrange bite." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/55045-time-release-monster-speaker-resonance)

- "The default LF Resonance is based on the cab most likely to be used with that amp. For example, if you select a Twin Reverb model it will use the resonance data from the Twin Reverb amp that we used as the reference (we measured all the cabs as well as shooting IRs). Your 1960A cab probably has its resonance around 80 Hz so ideally you would want to do exactly what you did. The ultimate solution would be to measure the LF resonance but that requires special test equipment. I've found that you can usually find it by ear though. As you adjust the LF resonance you'll hear the cabinet sympathize. There is no way around this at the moment as no modeler can measure the impedance of a guitar cabinet (despite any claims to the contrary). Note that this is not applicable when using FRFR and IRs."
- "The resonant frequency goes up when mounted in a cabinet. It doesn't need to be spot-on. If you are within 10% you'll be fine. If you want to be anal about it you can use an impedance analyzer. This is what we use: Dayton Audio DATS Dayton Audio Test System 390-806." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/68009-speaker-tab-settings-amp-block-g12k100.html#post836372)
- "Guitar folklore has it that SRV and Joe Walsh intentionally mismatched their speaker impedance. I imagine others have done this. The general idea is you plug an 8-ohm speaker into the 4-ohm jack or vice-versa. The Axe-Fx allows you to replicate this behavior using the Transformer Match control. To simulate plugging an 8-ohm speaker into the 4-ohm jack set Transformer Match to twice it's current setting (i.e. 2.0). For the other way around set it to half (i.e. 0.5). And you don't have to worry about frying your OT. BTW, for this to be audible the Master Volume needs to be set very high so that you are clipping the (virtual) output tubes. Most amps set the impedance ratio of the OT so as to get maximum power from the power tubes (within the SOA). Some amps intentionally mismatch the OT to give more control over the distortion by riding the volume control, i.e. Trainwrecks. Most amps actually slightly undermatch the OT as the speaker impedance is greater than nominal outside of the midband. Trainwrecks

are overmatched." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/77944-simulating-speaker-impedance-mismatch.html#post949002) and source (http://forum.fractalaudio.com/cliffs-notes/77944-simulating-speaker-impedance-mismatch-2.html#post956205)

- "Most guitar speakers are roughly the same when it comes to the high-frequency reactive behavior. The impedance increases starting around 1 kHz and then increases at 3-4 dB/octave. This is due to the voice coil inductance. A pure inductance would increase at 6 dB/oct. but there are eddy current losses that make the voice coil look "semi-inductive". The Axe-Fx II models this with a high-order lossy inductor model. The low-frequency response of guitar speakers, however, varies greatly between speakers of different makes and models. This low-frequency response is a sharp resonance typically in the range of 50-150 Hz. The magnitude of this resonance varies from several to 20 times the nominal impedance. The impedance of a speaker influences the response of a tube amp since a tube power amp is essentially a transconductance amplifier. It creates a current for an applied voltage. This current in turn creates a voltage across the speaker terminals that is dependent upon the impedance of the speaker. Therefore the power amp will resonate at the resonant frequency of the speaker. This causes certain notes to become emphasized as they excite the resonant frequency. Negative feedback around the power amp will reduce the amount of resonance but not all amps use negative feedback (i.e. Vox). The increased voltage amplitude at the resonant frequency also causes the power amp to clip sooner at the resonant frequency. Think of it this way: if the power tubes are swinging, say, 200 V at the midrange frequencies, they will swing X times more at the resonant frequency where X is the ratio of the resonant impedance to the nominal impedance. So if the resonant impedance is 10 times the nominal impedance the power tubes will want to swing 2000 volts. This is impossible so they will clip. For high-gain tones this can cause the tone to sound muddy or feel spongy. For lower gain tones this can thicken the tone and make it feel, well, more spongy. Cabinet/speaker IR data does not contain the impedance information. The only way to obtain impedance data is to measure the current vs. voltage vs. frequency (despite what modeler advertising literature would like you to think). The Axe-Fx uses default values of LF Resonant frequency and impedance for each amp model. For models based on combo amps these values are derived from measurements of the actual amp's speaker. For models based on amp heads the values are based on measurements of the cabinet most likely to be used with that head. You can adjust the frequency and impedance to suit your taste. Reducing the impedance (Low Res) will reduce the bass response and can give tighter bass. Raising the impedance will increase the bass response and can give a fuller sound. Altering the frequency (Low Freq) will change the frequencies at which the power amp resonates and tuning this to the key you are playing in can be an effective strategy, e.g. set it to 82 Hz if playing in E. Don't be afraid to try drastic settings. Try turning Low Res all the way to zero. Compensate by adding some bass with the Bass knob or the EQ section. As I mentioned earlier the LF Resonance will cause the power amp to clip earlier than it will when amplifying midrange frequencies. Turning down the Master Volume will increase the headroom in the power amp and reduce this clipping. Furthermore the Transformer Match also influences when the power amp clips. So there is a relationship between LF Res, MV and Transformer Match. Many manufacturers publish impedance data for their speakers. Eminence and Jensen and probably others publish detailed impedance data. You can look at the impedance plots and set the resonance parameters to match (roughly). The Low Res parameter is indicated from 0 to 10 and sets the resonance in dB from 0 to 24 dB (dB is a ratio of powers so it's not really the proper units for this but that's semantics). For example, the Jensen P12N has resonant frequency of about 100 Hz so you would set Low Freq to 100 Hz. The impedance at this frequency is about 40 ohms. To get the Low Res amount use the formula (20 * log10(Zr/Rdc)) / 2.4 where Zr is the impedance at the resonant frequency and Rdc is the DC resistance. For this speaker Low Res is then (20 * log10(40/6.2)) / 2.4 = 6.7. A power amp isn't perfect though. Winding resistance in the output transformer increases Rdc, typically by a couple ohms. Therefore our above example would become (20 * log10(40/8.2)) / 2.4 = 5.7. The exact value isn't overly critical though and all this is subtle nuances. The resonance Q is a bit more difficult to calculate. It is derived from the bandwidth at the points where the impedance "gain" is the square root of the resonance impedance gain. IOW, if the impedance is, say, 10 times the nominal impedance then the bandwidth is given by the frequencies where the response is 3.16 times the nominal impedance. For our example the resonance gain is 5 (40 / 8 = 5). So the bandwidth is the frequencies at which the impedance equals sqrt(5) * 8 = 18. From the graph this is approximately 75 Hz and 130 Hz. Q is defined as f0 / bw so our resulting Q is 100/60 = 1.67. Most speakers have a Q of around 2.0 or so. Again the exact value isn't overly critical and don't be afraid to try extreme settings (you can't break anything). Finally, just because real speakers behave like this doesn't mean we have to adhere to this behavior. Perhaps a better speaker has no resonance (Low and High Res are zero), or maybe the Q is a lot lower or higher. In our virtual world we can design a speaker that is impossible to construct in the physical universe. tl;dr version: Mess with Low Freq and Res if you want, or not." source (http://forum.fractalaudio.com/cliffs-notes/78003-about-speaker-lf-resonance.html#post949557)

- "The speaker page is the impedance vs. frequency. For a guitar amp with no negative feedback the voltage frequency response of the power amp will very closely match this since the power amp is basically a current source (V = I * Z). There will a slight reduction in the peaks as the output impedance isn't infinite but it is very high and will therefore be very close." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/79701-1-1-let%60s-talk-b-testing-actual-amps-their-axe-fx-ii-counterparts-3.html#post970197)
- "This post is aimed at those who use a solid-state power amp into a conventional guitar cab. As is described in the post "About Speaker LF Resonance" a guitar cabinet has an impedance resonance in the low frequencies. This typically falls in the range of 50 to 100 Hz. A tube amp, being essentially a current source, will have a voltage output that follows the impedance curve. Speakers, being electromotive devices, respond to applied electromotive force (EMF) which we know as voltage. A solid-state power amp is a voltage amplifier and, hence, will not be influenced by the impedance of the speaker. When using a solid-state power amp into a conventional guitar cabinet the experience will be different if the simulated speaker in the Axe-Fx II is not adjusted to match the actual speaker. Whether or not this is important is up to the individual but I imagine a lot of the posts about "in-the-room using power amp and cab is not the same" are due to this. Unfortunately the Axe-Fx II cannot measure the speaker impedance characteristics as it is not directly connected to the speaker. No device can measure the speaker impedance without being directly connected to the speaker, despite what their marketing claims may infer (cough, ahem...), since impedance is, by definition, V/I and we cannot measure these unless connected to the speaker terminals. The only truly accurate way to set the simulated speaker is to measure the speaker being used with an impedance measuring device. These can be had relatively inexpensively in the

form of the Woofer Tester 3 (from Dayton Audio IIRC). You can also make your own using a small value resistor (0.1 ohms or so) in series with you power amp and measure the voltage across the resistor. The next best method would be to estimate the impedance using published data from the speaker manufacturer. If the make and model are known the data may be available. Add approximately 10% to the published resonance frequency if the speaker is in a sealed box. The worst method, and the subject of some contention, is finding the resonance by "feel". No power amp has perfect damping. If you put a sine wave (use the Synth block) into the speaker you may be able to observe or feel the resonant frequency. The cone will have increased excursion at this frequency. Of course you may just be feeling the room resonance. I have used this technique successfully on several speakers but it takes practice. The main drawback is that the magnitude of the resonance is unknown. The Axe-Fx II's Low Res parameter is displayed in dimensionless units from 0 to 10. Each unit corresponds to 2.4 dB of impedance "gain". We define this as a gain since the our current source power amp will experience a voltage gain. This is relative to the DC resistance of the speaker. For example, if the speaker's resistance is 6 ohms and the impedance at resonance is 60 ohms then our impedance gain would be 20*log10(60/6) = 20 dB. Dividing by 2.4 gives a Low Res value of 8.3. Since a tube amp isn't a perfect current source these values should be reduced slightly. The exact value of the Q isn't too important. About 2.0 is a good starting point. Adjust up or down to taste. If you are anal more information is in the aforementioned post about deriving the value of Q. Once the simulated speaker is set correctly you may notice a difference in low-frequency behavior and pick attack." source (http://forum.fractalaudio.com/cliffs-notes/79816-about-matching-your-cabinets-resonant-frequency.html#post970496)

- "The default values in the amp block are based on the measured impedance of the most commonly used speaker for the selected amp. If it is a combo amp then the values are derived from the speaker that was in the cabinet. Most people who capture IRs are unable to also capture impedance data." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/75794-something-cool-ive-been-working-6.html#post927165)
- "In the SPKR page of the Amp block are various parameters. I've talked about low-frequency resonance in another post. In this post I will address high-frequency resonance. As with LF resonance the high output impedance of a tube power amp causes the frequency response to follow the impedance of the speaker. There are two primary components: LF resonance and a high-frequency boost. The HF boost is due to the inductance of the voice coil. At the frequency where the voice coil reactance is equal to its resistance the impedance will start to rise. If this were a "pure" inductance it would rise at 6 dB per octave. However eddy-current losses in the motor cause this inductance to be "semi-inductive" and the impedance typically rises between 3 and 4 dB per octave. Different brands and models of speakers behave differently. You can look at the spec sheet for a speaker to get an idea as to the behavior of the speaker. The formula for the break frequency is given by f = R / (2*pi*L). For example, if the voice coil inductance is 1 mH and the resistance is 7 ohms then the break frequency would be 7/(6.28*0.001) = 1.1 kHz. The Axe-Fx II allows you to adjust the virtual voice coil via the HI FREQ and HI RES parameters. The HI FREQ parameter sets the "break" frequency which is the frequency where the inductive reactance equals the voice coil resistance. For most speakers it is around 1000 Hz. It is lower for larger speakers and higher for smaller speakers usually. The HI RES parameter sets the rate at which the impedance increases. The default value of 5.83 is around 3.5 dB per octave. If you want a smoother sound you can increase HI FREQ and/or decrease HI RES. If you want more

highs or "chime" you can decrease HI FREQ and/or increase HI RES. Experiment with different values to get a feel for the response. Note that the amount of feedback (Damping parameter) will influence the behavior of these controls. With no feedback (Damping = 0) the frequency response follows the impedance curve virtually 1-for-1. As Damping is increased the frequency response flattens and the impedance curve has less influence on the response. As with all things in the Axe-Fx, use your ears." source (http://forum.fractalaudio.com/cliffs-notes/81121-about-hf-resonance.html#post985574)

- (answering: once I know what frequency to pick for my cab, and roughly which amount and Q to use, is it ok and expected to use the same settings for every amp model?): "Yes, keep the same settings for all models." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/91139-clarification-lf-resonance-real-cab-speaker-tab.html#post1097466)
- Firmware 5.02: "HF Resonance is similar to the previous control but only changes the slope of the resonance. The default value is consistent with the typical “semi-inductance” of a speaker voice-coil. Varying this value will change the high-frequency load presented to the virtual power tubes."
- "Negative feedback does not affect the speaker impedance. Speaker impedance is an independent variable. The default values are based on the speaker cabinet most commonly used with the amp. In the case of a combo it's the internal speaker. For heads it's the mating cabinet, if one. A typical speaker has a low-frequency resonance with some frequency, Q and magnitude. The Q is typically around 2, frequency around 100 and magnitude around 12 dB. These values are dependent upon the speaker construction and, to a lesser extent, the cabinet. The voice coil inductance causes the impedance to increase at high frequencies. Unlike a pure inductance which would increase at 6 dB/octave, voice coil inductance is semi-inductive and typically increases at 3-4 dB/octave. The "break" frequency is dependent upon the actual inductance and is adjustable. The Axe-Fx is unique in that it lets you adjust these values. Most products just use a fixed curve." source (http://forum.fractalaudio.com/threads/the-speaker-page.108172/page-2#post-1295724)
- "The voltage at the output of a tube amp is a function of the speaker's impedance curve. The impedance curve is set in the SPKR tab of the amp block. Real speakers respond to the VOLTAGE at their terminals, not the current. The amp block creates a virtual voltage which is a function of the amp model and the impedance curve. The cab block uses IRs which represent the measured sound from the speaker vs. an applied voltage at the speaker terminals. The aforementioned virtual voltage is sent to the cab block which then produces an audio signal by convolving the virtual voltage data with the IR. As I said in my previous post the IR is largely independent of it's impedance curve. The impedance curve can then be seen as just another tone and feel shaping tool. Bass too prominent? Turn down LF Resonance. Want more midrange? Lower HF Res Freq. The degree to which the impedance curve affects the output of the amp block is a function of the power tube type and the amount of negative feedback. The less negative feedback the more the impedance curve influences the output. Since Presence and Depth work by reducing negative feedback at high and low frequencies respectively, increasing them will also increase the influence of the impedance curve at those frequencies. The dynamic impedance of the power tubes also affects the output. A power triode, for example, has a much lower plate impedance than a tetrode or pentode and, therefore, the output will be mostly independent of the speaker impedance." source (http://forum.fractalaudio.com/threads/the-speaker-page.108172/#post-1295117)

- "You can simulate changing power tubes in the Axe-Fx by simply increasing or decreasing the LF and HF resonance values." source (http://forum.fractalaudio.com/threads/why-power-tubes-sound-different.79962/)
- "(Quantum 2.0) "Many of the amp models had their speaker parameters updated based on new measurements." source (http://forum.fractalaudio.com/threads/2-beta-3-possible-bug-with-speaker-low-res-not-a-bug.109890/#post-1315544)
- "If you want simulated power amp breakup, turn Power Amp Modeling on and then go to the SPKR tab in the amp block and turn the Low and High Res parameters all the way down so that you get a flat response. This will let you use amps that get their distortion from the power amp, i.e. Fenders, Vox, etc." source (http://forum.fractalaudio.com/threads/sorting-out-the-high-end.117233/#post-1399284)

MASTER VOLUME CAP

- This parameter sets the value of the bright cap across the Master Volume pot.
- Cliff's comments:
  - "Setting it to 1 pF defeats it." source (http://forum.fractalaudio.com/news/74402-axe-fx-ii-firmware-v11-01-new-preset-banks.html#post910428)

MASTER VOLUME LOCATION

- Cliff's comments:
  - "Most amps are Pre-PI, including Dumbles. Post-PI is rare and often does as a mod. This causes the PI to distort rather than the power tubes. It is a harsher sound." source (http://www.thegearpage.net/board/showpost.php?p=14422541&postcount=28)
  - "Moving the Master Volume after the Phase Inverter. This causes the phase inverter to distort. It is a popular mod on amps. In the Advanced menu change MV Location to "Post-PI"."source (http://forum.fractalaudio.com/axe-fx-ii-discussion/84166-post-pi-mv-try.html#post1020827)
  - "Post-PI MV, Try It! Turns a lot of mid-gain amps into ripping monsters. I just tried it on the JCM800 and, dayum... The only caveat is that, like a real amp, the more you turn the MV down the less effective Presence and Depth become (since the loop gain is reduced)." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/84166-post-pi-mv-try.html#post1020822)
  - "The swing of a properly designed PI is greater than the range of the power tube grids. For example a 6L6 will typically be biased around -50V. This gives a range of around 100V of swing at the power tube grids. A well-designed PI for an amp like this will typically swing at least 150V and usually closer to 200V or more. This means the grids clip before the PI clips. Then there are the power tube plates. Most designers choose the output transformer so that the plates just start to clip when the grids clip. This gives maximum power since current is maximum when the grids are driven to clipping so you want the plate voltage to be at its maximum excursion at this point. This is a matter of taste. Some designers slightly undermatch since the impedance of the speakers increases at low and high frequencies and this gives a more open tone. Others overmatch as this give a more touch-sensitive overdrive (i.e. Trainwrecks). When you put the MV after the

PI you attenuate the signal from the PI going to the grids. This allows the PI to clip before the grids clip. The PI has a fair amount of negative feedback so it's a somewhat hard clipper which gives a fairly aggressive distortion." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/84166-post-pi-mv-try.html#post1021130)

MODELING VERSION

- Selects the modeling version for new presets. Not available on Axe-Fx II Mark I and II or on AX8.
- If this parameter is set to anything other than Latest, the latest modeling version will still be used if the parameter Force Default Version in the Global menu is set to On.

NEGATIVE FEEDBACK (DAMPING)

- Changing Negative Feedback (Damping) between 0 and another value will change the display on the hardware, and switches between Presence and HiCut.
- Note that adjusting this parameter affects the amp's volume level. If the power amp is saturated, both damp and level must be increased to maintain level the same.
  - Cliff: "The Axe-Fx attempts to normalize the volume as you change the damping. (...) However, if you are driving the "power amp" hard the equation falls apart because it assumes linear operation. Therefore there may be some volume change. This is done since otherwise you would constantly have to adjust your output volume as you change the damping. Unfortunately it is impossible to predict how saturated the power amp is since that depends on input level. The compensation isn't perfect, the idea is to minimize the volume fluctuations since without compensation the volume would fluctuate wildly."
- Cliff's comments:
  - "When you set Damping to 0 Presence becomes a Hi-Cut so if you have the Presence turned up you'll lose high end when you turn Damping to 0. Also, Depth does not work at a Damping of 0 since Depth modifies the feedback and there is no feedback." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/59333-damping.html#post740665)
  - "Another factor which controls power amp hardness is Transformer Match. There are two primary distortion mechanisms in a power amp: grid clipping and plate clipping (PI clipping notwithstanding as this is only audible with a post-PI MV). Grid Clipping is extremely hard, almost a hard clipper (i.e. if(x>a) then x=a). Plate clipping is much softer. However most power amps are slightly undermatched which means the grids clip before the plates clip, but only at those frequencies where the speaker impedance is "nominal". At high frequencies (above 1kHz or so) the rising impedance of the speaker causes the plates to clip before the grids. At the low frequency resonance the plates also clip first. If you increase the transformer matching the plates will clip earlier and, since plate clipping is softer, the distortion will be softer. So turn up the Transformer Match and turn down Negative Feedback for softer power amp distortion." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/100622-scratchy-sound-patch-same-you-2.html#post1207496)
- Cliff talks about Negative Feedback (Damping) in this Tech Note

(http://forum.fractalaudio.com/cliffs-notes/79697-about-negative-feedback.html) .
Excerpt:

- "The Axe-Fx II allows the user to fine-tune the amount of negative feedback in the power via the Damping parameter. The term damping refers to the fact that increasing negative feedback lowers the output impedance and therefore "dampens" the response of the speaker. Power amps often specify their output impedance in terms of "Damping Factor" which is the ratio of the load impedance to the output impedance. The higher the damping factor the less the speaker impedance influences the frequency response. Let's examine what happens as you adjust the Damping parameter. As we increase the Damping we increase the negative feedback. This does several things:

1. LOWERS THE GAIN of the power amp. This causes the power amp to not distort as easily since the signal is amplified less and therefore it won't clip as easily.
2. Increases the linearity of the power amp. This reduces harmonic distortion but makes clipping "harder" as the transition to clipping is more abrupt.
3. Flattens the frequency response. This makes the frequency response more linear and widens the bandwidth. The peaks in the frequency response due to the speaker impedance are flattened and broadened.

- As we decrease Damping we decrease the negative feedback which does:
    1. Increases the gain of the power amp. The causes the power amp to clip more readily.
    2. Decreases the linearity of the power amp. This increases harmonic distortion and softens the transition into clipping.
    3. Increases frequency response distortion. The response becomes more scooped and the bandwidth is reduced.
- Many guitar players like the sound of amps with no negative feedback. The Vox AC-30 is the classic example of an amp with no negative feedback. The power amp distortion is soft and the scooped response along with lots of harmonic distortion give a bell-like tone for high frequencies and warm low frequency response. The drawback to this is that the low end can get muddy as the low frequencies clip readily due to the frequency response distortion. These types of amps typically do not work well for high-gain tones although there are notable exceptions, i.e. the Dual Rectifier which uses a high-power power amp and bass reduction in the preamp to compensate for the increased bass response.
- Fender and Marshall amps (and their derivatives) use varying amounts of negative feedback. The amount of feedback in Marshall amps was all over the map in the early years and seems as though the builders didn't really adhere to rigorous documentation and revision control. As such there can be quite a bit of variation in the sound of these early Marshalls.
- So what is the correct amount? There is no definitive answer however there are some guidelines. For more vintage tones less Damping is typically desirable. This gives softer power amp breakup and more "baseline" distortion. For modern, high-gain tones more Damping may be desirable as these tones typically rely on preamp distortion and the power amp is desired to be neutral (which many players describe as "tight"). As stated before the Dual Rectifier in modern modes is a bit of an enigma. The power amp in this mode uses no negative feedback. You can hear this as an increase in volume when you flip the switch to Modern (remember that negative feedback reduces the gain so turning it off will increase the gain).

    - I have read some users recommending increasing the Damping to reduce the amount of power amp distortion in, for example, Fender models. I do not endorse this viewpoint. The distortion is primarily reduced because the gain is reduced but the power amp will sound more "sterile" due to the increased linearity and flatter frequency response. A better solution is to simply lower the Master Volume. This drives the power amp less while retaining the baseline harmonic distortion, softer transition into clipping and more scooped frequency response.
    - Modification of the error signal is commonly employed in guitar amps. This was first employed as the ubiquitous Presence control. The presence circuit reduces the amount of feedback at higher frequencies. Since the gain of the power amp is inversely proportional to the amount of feedback, reducing the amount of feedback over only a certain range of frequencies will therefore increase the gain of the amp at those frequencies. The presence control therefore boosts high frequencies. This concept was extended to low frequencies via the Depth or Resonance control. Basically these controls are bass and treble controls for the power amp but operate by reducing the feedback for those bands. On most amps setting these controls fully CCW will basically remove them from the circuit. Turning them CW will reduce the feedback in the prescribed bands thereby increasing the gain of these bands. An exception are Boogie (Mark series) power amps. The presence control in these amps is more complex and the flattest frequency response is achieved with the knob set to noon. Turning the knob CCW will reduce the treble response. Turning it CW will increase it.
    - Note that these all interact with the Damping control. To hear this reduce the Damping to a value just greater than zero. Note that the Presence and Depth controls will have almost no effect. This is logical since B (beta) is nearly zero and we can't reduce it further."

OUT COMP TYPE, OUT COMP AMOUNT, OUT COMP CLARITY, OUT COMP THRESHOLD

- This parameter controls a compressor specifically tailored to reducing the output dynamic range of the Amp block. Note that this compressor runs in the master DSP and if set to a non-zero value will increase CPU usage. The Out Comp parameter controls the amount of compression (compression ratio). In the Advanced menu the user can also adjust the compression threshold via the “Comp Thrshld” parameter, if desired. The bar graph at the bottom of the menu displays the amount of gain reduction.
- Firmware 19: "The Output Compressor in the Amp block now features two modes of operation: Output and Feedback. These modes are set using the new Comp Type parameter. “Output” is the previous type where the compressor acts on the output of the block. “Feedback” also compresses the block output but applies dynamics to the input of the block based on the output compression. The “Comp Clarity” parameter adjusts the bass response of the input dynamics and can be used to add clarity to the bass."
- Non-zero values increase CPU usage!
- The effect of the compressor setting can be watched on the DYN page of the Amp block.
- Cliff's comments:

    - "The compressor tries to apply make-up gain but it can only guess at the amount." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/87485-output-comp-strangeness.html#post1058658)
    - "Output Comp is compression ratio. Ratio = 1 + 3 * comp/10. Attack and release are fixed. Threshold is adjustable in the advanced menu." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/87494-ouput-comp-recommendations.html#post1058766)
    - "If you set the Compressor Type to Feedback and turn up the Output Compression you will get more distortion as you play harder. So you can create an amp that cleans up more when you play lightly or roll of the volume." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/101388-feedback-parameter-amp-block-output-compressor.html#post1216109)
- Output Comp Clarity affects OUTPUT COMP. It adjusts the bass response of the input dynamics and can be used to add clarity to the bass.

OUTPUT LEVEL

- The Output Level parameter in the Amp block (appears on several pages) controls the overall volume level of the preset, enables you to match it to other presets and prevents clipping the digital signal.
- It's the best place to set the preset's overall level. Unless you want define the volume level per scene, in which case you should use gain in the Output block.
- Cliff's comments:
    - "The amp block is always the place to set your volume. The Level control is repeated at several places in the amp block menus for convenience so you don't have to keep switching pages. The Level control has no affect on the tone." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/55340-so-many-volumes-options.html#post698522)

PI BIAS SHIFT (PHASER INVERTER BIAS SHIFT)

- Added in firmware Quantum 7.00:
    - "New algorithm also includes bias shifting which results in more harmonic spectrum variation with input amplitude. This improves feel, “knock” and creates sweeter single note soloing. The new “PI Bias Shift” parameter controls the amount of phase inverter bias shift. Note that some real amps are “spitty” in nature due to PI bias shifting, i.e. Trainwrecks, and the new algorithm is designed to replicate that behavior accurately. If you find the behavior undesirable reduce the PI Bias Shift value as desired although this will reduce authenticity."

PICK ATTACK

- Controls a sophisticated dynamic range processor that operates on leading edge transients. Negative values reduce pick attack while positive values enhance it.
- Not available on AX8.
- Cliff's comments:
    - "It doesn't have any particular frequency. Pick attack is impulsive so, by definition, it contains all frequencies. The standard approach to reducing attack is to use

dynamics processing. The Axe-Fx II has a Pick Attack parameter which can be used to reduce the attack but the AX-8 does not have this parameter. You can try using the Gate/Expander to soften the attack." source (http://forum.fractalaudio.com/threads/how-to-eq-out-pick-attack.118497/#post-1410629)

POWER AMP CATHODE RESISTANCE

- Cliff's comments:
  - "Sets the cathode resistance of the power tubes. Should only be used with "Class A" amps, i.e. AC-20 DLX, etc. Lower values increase the bias current. Note that some amps have separate bias resistors while others have a shared bias resistor. The choice of split/shared is not exposed to the user." source (http://forum.fractalaudio.com/threads/cathode-resistance.122674/#post-1459417)

POWER AMP HARDNESS

- This parameter controls the hardness of the virtual power tube grid clipping.
- Cliff's comments:
  - "The hardness is tied to the tube type. I.e., if a model uses EL84s the default hardness will be 2. Different tube types will default to different hardness." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/98572-fw-18-06-amps-automatically-set-power-amp-hardness-4-instead-3-medium.html#post1182388)
  - "You can control the "shape" of the preamp and power amp distortion. The Preamp Hardness parameter controls the shape of the triode emulations. The lower the value the softer the distortion. The Power Amp Hardness controls the power amp clipping but that often is not noticeable because negative feedback around the power amp makes the distortion harder. Therefore you can make the power amp distortion softer by reducing Negative Feedback. A good example of this is a JCM800. A JCM800 has very hard preamp distortion (since there is no cathode bypass cap on the last stage) but has low negative feedback which softens the power amp distortion. The trick with that amp is to get the amp into the sweet spot by increasing the MV until you are getting some power amp distortion which softens the preamp distortion. Another factor which controls power amp hardness is Transformer Match. There are two primary distortion mechanisms in a power amp: grid clipping and plate clipping (PI clipping notwithstanding as this is only audible with a post-PI MV). Grid Clipping is extremely hard, almost a hard clipper (i.e. if(x>a) then x=a). Plate clipping is much softer. However most power amps are slightly undermatched which means the grids clip before the plates clip, but only at those frequencies where the speaker impedance is "nominal". At high frequencies (above 1kHz or so) the rising impedance of the speaker causes the plates to clip before the grids. At the low frequency resonance the plates also clip first. If you increase the transformer matching the plates will clip earlier and, since plate clipping is softer, the distortion will be softer. So turn up the Transformer Match and turn down Negative Feedback for softer power amp distortion." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/100622-scratchy-sound-patch-same-you-2.html#post1207496)

POWER SUPPLY TYPE

- AC = Alternating Current.
- DC = Direct Current.
- Firmware 10: "Amp block power supply modeling now models AC rectification and resulting supply ripple (if Pwr Supply Type is set to 'AC'). The power supply type can be selected between AC and DC with the Pwr Supply Type parameter. The line frequency can be selected with the AC Line Freq parameter. Note that high values of Sag along with low B+ Time Constant values can cause "ghost notes" when the supply type is AC (as in a real amp). Lower B+ Time Constant values will make the amp feel "faster" but too low can cause ghost notes."
- Cliff's comments:
  - "The most notable thing would be the ghost notes. There is also a slight difference in feel." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/67073-ac-power-frequency.html#post826893)
  - "Those old amps make ghost notes. My 100W Plexi has some ghost notes that are louder than the fundamental. The easiest way to eliminate them (if you don't like/want them) is to simply set the Supply Type to DC. However, IMO, the ghost notes are a large part of the character of these designs and removing them isn't desirable. Don't over-analyze it. Recognize that certain designs produce ghost notes and embrace it." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/70962-can-someone-check-out-jtm-45-a.html#post870070)
  - "100 watters are usually worse because they have lighter filtering (50 uF vs. 100 uF) and draw more dynamic current." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/104471-ac-vs-dc-power-supply-parameter-try.html#post1250025)

POWER TUBE GRID BIAS (PA GRID BIAS)

- This parameter can be used to adjust the offset voltage of the virtual power amp (this should not be confused with the Power Tube Bias parameter which sets the quiescent operating current of the virtual power tubes). Power Amp Bias allows the user to vary the symmetry of the clipping of the virtual power amp. A value of zero produces nearly symmetrical clipping which will produce very little even harmonics. Higher values will produce increasingly asymmetrical clipping which increases the amount of even harmonics. Small amounts of even harmonics can make the power amp distortion sound "warmer" and more bell-like while higher amounts will give a "fuzzier" tone. Most amps have some amount of offset and the amp models will default to a typical value. Note that this parameter is only applicable for push-pull power amp types. For single-ended power amps the Power Tube Bias parameter sets the symmetry (as always).
- Cliff's comments:
  - "This sets the quiescent operating point of the virtual power tubes. What is the quiescent operating point you ask? It is the amount of idle current flowing through the virtual tubes when no signal is present. Power tubes are basically nonlinear controlled current sources. They can only sink current (current flow in one direction only) so to get an alternating signal you have to have some amount of idle current then you modulate that. The push-pull power amp was invented to

increase efficiency by allowing lower idle currents. The name "push-pull" refers to the fact that one tube is responsible for mainly the positive portion of the waveform and the other for the negative. Some overlap of the responsibility is required since power tubes are nonlinear and without that overlap crossover distortion occurs. Setting the idle current, or bias as it's commonly known, affects the resulting transfer function of the power amp. Too little bias and there can be excessive crossover distortion. So-called "Class A" amps bias the tubes quite high. The following graph depicts the transfer function of the Axe-Fx II for five different bias settings: 0, 0.2, 0.4, 0.6 and 0.8. These are the red, green, blue, cyan and black traces, respectively. (see source URL for graphic) Note the severe crossover distortion at zero bias. At a bias of 0.4 the transfer function is almost a perfectly straight line. At 0.8 the response gets softer. Amps like AC-30s run the bias around 0.7 which gives them the soft distortion characteristics. Note that this applies to power amp distortion which only occurs when the power amp is driven hard (Master Volume set high). Some amps are intentionally biased cold to generate crossover distortion. Small amounts add an aggressive distortion. Some amps (i.e. Boogies) are intentionally biased cold to avoid having to set the bias and thereby reducing maintenance and warranty costs. Negative feedback around the power amp (Damping) further linearizes the power amp. So the transfer functions depicted are only accurate when Damping is zero. In conclusion, Power Tube Bias is a powerful parameter that can allow you to fine-tune the power amp distortion characteristics to your particular style." source (http://forum.fractalaudio.com/cliffs-notes/79049-understanding-power-tube-bias.html)

- "The only possible parameter I can see adjusting to thicken clean tones is Power Tube Bias. The higher the value the less crossover distortion. Note that many other modelers don't even model crossover distortion. Small amounts of crossover distortion add aggression to high-gain sounds. In fact it is said that EVH liked to bias his amps cold to get that extra bite. The infamous HM-2 pedal intentionally adds crossover distortion. However crossover distortion can make clean tones sound thin. Many amps renowned for their clean tones run the power tubes hot so there is little or no crossover distortion. These amps are commonly (and mistakenly) referred to as "Class A" but they are really cathode biased Class AB with the tubes biased hot. Many people bias Fender-type amps hot to get warmer clean tones (at the expense of tube life). The default bias value in most cases is such that the bias point is roughly 60% of full-power ("Class A" types not included). If you increase the bias value to 0.5 or so you'll be running the virtual tubes at around 75% of full power and clean tones will sound warmer but you will lose that sizzle on high-gain tones." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/74564-one-more-piece-puzzle-solved-thinness-individual-notes-3.html#post915848)
- (Quantum 2.0) "Tube Bias controls the idle current of the virtual power tubes. The models, in general, are biased on the hot side because that's how most people like their amps. Reduce Tube Bias to simulate a colder bias. It is rumored that EVH liked his amps biased cold. When the bias is reduced the amp sags and bounces more and the bass will be a little tighter and the tone edgier." source (http://forum.fractalaudio.com/threads/quantum-2-00-tips-part-1.110275/)

POWER TUBE TYPE

- Firmware 10: "Added Tube Type parameter to amp block. This allows selecting Tetrode (i.e. 6L6, KT66, etc.) or Pentode (i.e. EL34, 6BQ5, etc.) power tube types. The type defaults to the appropriate value when a model is chosen but may be overridden by the user."
- Firmware 13: "Added selectable power tube types for Amp block. Available types are: EL34, EL84, 6L6, 6V6, KT66, KT88, 6550, 6973, 6AQ5 and 300B (triode). Also available are an ideal tetrode and ideal pentode. The power tube type defaults to the appropriate type when the amp type is selected but may be overridden by the user. The power tube type presets the Dynamic Damping parameter as well as several internal parameters."
- Cliff's comments:
  - "The tubes are "normalized" so that you don't have to do anything to the transformer. The internal values for the power tubes are based on typical transformer values. For example, the values derived for the EL34 are based on a primary impedance of 3200 ohms (assuming one pair of tubes). The impedance ratio of the transformer affects the values slightly as the plate impedance is reflected to the secondary (or the load is reflected to the primary) and the ratio therefore affects the interaction." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/80497-axe-fx-ii-firmware-version-12-04-public-beta-4.html#post977955)
  - Tech Notes: "You'll often read that 6L6's sound "full" whereas EL34's have more midrange and other colloquial descriptions of the tone of a power tube. These myths are perpetuated by forum dwellers, uninformed tube "experts" and even amp manufacturers as marketing tools. Well, the fact is that power tubes do NOT sound different. They do not have any intrinsic tone. "But I can hear the difference when I change to a different type of power tube. How can that be?" A power tube has a very flat frequency response and they all clip roughly the same. If you put a resistive dummy load on a tube power amp (assuming it doesn't have any intentional frequency shaping) it will measure very flat. However a speaker is not a resistive load. A speaker is a highly reactive load. As I've mentioned in the other threads in this forum section a speaker has an impedance that is sort of scooped at the midrange frequencies. It is the impedance of the speaker that affects the tone of the amp and different types of power tubes react differently with that impedance. As I've mentioned before a power tube is nearly a current source. The operative word here is "nearly". No power tube has an infinite plate impedance and that's why power tubes sound different. A current source has infinite output impedance, an actual power tube has a finite output impedance. The output impedance of a power tube (or any active device for that matter) is defined as delta V / delta I which is the change in voltage vs. the change in current. Let's take a 6L6 for example. Let's assume that the tube has a quiescent operating point of 300V and let's assume we swing +/- 100V around that point. If we look at the plate graphs for a 6L6 at a bias of -10V we see that the plate current at 200V is 95 mA and at 400V it's 105 mA (roughly). Using our formula for impedance we get 200/0.01 = 20 Kohms. Now let's take an EL34. At 200V the current is 130 mA and at 400V the current is 150 mA. The plate impedance is therefore 10 Kohms which is half that of the 6L6. This lower output impedance "de-Q's", or flattens, the speaker impedance. Essentially the EL34 has a higher damping factor than a 6L6. This higher damping factor reduces the mid-scoop due to the speaker impedance. This makes the tone have more midrange. There's a little more to it as the output transformer plays a role as well and 6L6 power amps typically have a

slightly higher impedance ratio. There's also different operating voltages and bias points but I'm trying to keep this simple. You can simulate changing power tubes in the Axe-Fx by simply increasing or decreasing the LF and HF resonance values." source (http://forum.fractalaudio.com/cliffs-notes/79962-why-power-tubes-sound-different.html#post972283)

- "Changing the power tube between pentode and tetrode doesn't change the sound in the same way actually changing tubes would because it only changes the distortion curves. It does not change the transconductance so the transformer matching is constant. When you put different power tubes in an amp the difference in tone isn't due to some inherent difference in the "sound" of the tubes. It's mainly due to the different transconductances. The transconductance of an EL34 is about 30-40% more than the transconductance of a 6L6 (edited (http://forum.fractalaudio.com/axe-fx-ii-discussion/94404-preamp-tube-types.html#post1131521)). This means that the plate current will be twice as great for a given grid voltage. This makes EL34s sound "more midrangey" and 6L6s sound "tighter" or "fuller". The truth is that if you bias them correctly and compensate for the difference in transconductance you will hear very little difference. Unfortunately you can't compensate for the transconductance easily in a real amp without changing the gain of the phase inverter and/or putting in a different output transformer." source (http://forum.fractalaudio.com/axe-fx-ii-tone-match/75082-mark-v-tone-match.html#post924520)

PREAMP BIAS

- PREAMP BIAS sets the bias point of the last triode (cathode follower not counted). Depending on the bias points of the previous stages increasing or decreasing this value can alter both the harmonic content and the attack characteristics. Typically, if the previous stage has a negative bias then increasing this value will be more noticeable and vice-versa. This value is set to a default value for the model whenever the type is changed but can be overridden by the user.
- Cliff's comments:
  - "The further you move away from (roughly) zero the more even harmonics are introduced. It's an asymmetric transfer function, so you have to experiment." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/74457-more-gain-11-01-a.html#post911042)
  - "The primary controls to adjust the saturation behavior are Preamp Tube Type, Preamp Hardness and Preamp Bias. (...) Preamp Bias adjusts the bias point of the last triode stage which will control the ratio of even/odd harmonics. Values around zero will produce mostly odd harmonics. As you deviate from zero you'll produce less odd and more even." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/97925-suggestions-what-youve-done-enjoy-18-03-a-6.html#post1178319)
  - "Bias points in an amp are important. The bias point of the last preamp stage is the most important. Therefore Quantum firmware exposes this bias point on the Pre Dyn tab as the Preamp Bias parameter. Most amps are biased slightly towards cutoff (negative). The closer you get to 0.0 the more odd harmonics and the fewer even harmonics. Experiment with this to craft your "Ultimate Tone" (TM, all rights reserved, use only as directed, these statements have not been evaluated by the FDA)." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/104687-one-quantums-powerful-tools.html#post1252309)

- "The Preamp Bias control in the Amp block controls the operating point of the last virtual triode stage in the preamp. This is the most important stage wrt to the feel and texture of distortion. The earlier stages are important but generally not nearly as much and the bias points are not exposed to the user. The operating point of a tube determines the symmetry of the clipping. If the tube is biased exactly halfway between the supply voltage and ground then it will clip symmetrically (greatly simplified). If the quiescent current is reduced the tube will biased more towards cutoff. If it is increased it will be biased more towards saturation. In general cutoff is smoother than saturation but it depends on the external circuitry. Negative values of Preamp Bias bias the virtual tube towards cutoff and positive values toward saturation. Symmetrical distortion has lots of odd harmonics and very little even harmonics. The more asymmetrical the distortion the more even harmonics are introduced. Odd harmonics give clarity and a more aggressive, open tone but this can be cold and harsh. Adding even harmonics gives a warmer sound but too much and things can get muddy. Getting the right balance of even and odd harmonics is one of the keys to achieving "edge of breakup" tones. Experiment with the bias point to find your optimum tone. Things get especially interesting when a cathode follower is involved. You can tell if an amp has a cathode follower if the Preamp Comp parameter is not zero. The cathode follower interacts with the last stage and slight adjustments to the bias point can cause major changes in the distortion characteristics. For example, the Dizzy Blue models are biased near zero (0.08 IIRC). If you play lightly you'll hear the bass is kind of stuffy and tubby. Reduce the Preamp Bias a bit and you'll hear the bass clean up. Too negative, however, and the sound can get indistinct. The good amp designers understand the interaction between the last stage and the cathode follower and tune the bias point for the desired distortion characteristics. The cathode follower is a bit of an imperfect design though. It's great for vintage Plexi and other high gain sounds but its clipping behavior is not ideally suited to certain tones. Therefore the Comp Type parameter allows you to choose an idealized cathode follower with different distortion characteristics (Comp Type == Ideal). Note that the behavior of this type is similar to the algorithms used in profiling modelers and other products. Try using the Ideal mode. You will likely need to reduce the amount of Preamp Comp as this mode has much more compression. Even amps that rely mostly on power amp distortion can benefit from fine tweaks to the Preamp Bias point. Shifting the bias point changes the harmonics into the power amp which changes the distortion character of the power amp (albeit less significantly)." source (http://forum.fractalaudio.com/threads/preamp-bias.110726/)
- "It's one of the main tools that amp designers use in voicing Marshall-style amps. For these amps you'll notice the amp gets tighter as you set Preamp Bias negative and chunkier for positive values. Too negative and things get thin and sputtery. Too positive and the lows get farty." source (http://forum.fractalaudio.com/threads/axe-fx-ii-quantum-rev-5-02-firmware-released.120535/page-5#post-1434706)

PREAMP CF COMP TYPE, PREAMP CF COMP, PREAMP CF TIME, PREAMP CF RATIO, PREAMP CF HARDNESS

- Cathode follower algorithm.
- The CF Comp parameter has been renamed Preamp Comp to better explain its

function.
- Preamp Comp Type selects between “Authentic”, which accurately models the compression in a tube amp, and “Ideal” which is an idealized distorting compressor. The idealized type is more focused and has tighter bass whereas the authentic type is bolder and looser. High gain players may prefer the ideal type due to its tight character."
- Parameters:
  - Preamp Comp / CF Comp: sets the amount of compression.
  - Preamp CF Time: sets the attack time of the compressor.
  - Preamp CF Ratio: sets the maximum amount of compression with lower values giving more compression.
  - Preamp CF Hardness: adjusts the shape of the cathode follower distortion.

PREAMP DYNAMICS

- Dynamics processor that can be used to alter the dynamic response of the amp algorithms. When set below zero the amp compresses resulting in a smoother, less dynamic sound. When set greater than zero the amp expands resulting in a punchier, crunchier and more dynamic sound. Note that extreme values can have undesirable side-effects such as pumping and clipping.
- Cliff's comments:
  - "Dynamics works at the input to the block. Negative values compress the input, positive values expand." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/96779-output-comp-love.html#post1160978)
  - "The Dynamics knob in the Amp block does the same thing as the Dynamics mode of the compressors so you can save a block that way." source (http://forum.fractalaudio.com/threads/comp-vs-multicomp.107343/#post-1284495)

PREAMP HARDNESS

- Previously called: Triode Hardness (before firmware 16). source (http://forum.fractalaudio.com/axe-fx-ii-discussion/91241-triode-hardness.html#post1098468)
- This parameter controls how sharply the triodes enter saturation and can be used to simulate softer or harder tubes. The default value is 5.0 and is set to this value whenever the type is changed. The effect of this is subtle and most apparent at edge of breakup. Lower values give softer saturation, higher values give a more aggressive breakup.
- Cliff's comments:
  - "What the parameter does now is control the asymmetry of the triode model. The higher the value, the more asymmetrical the clipping." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/67091-v-10-triode-hardness.html#post826849)
  - "Lower values will have less even and more odd harmonics. The smoothness/harshness of distortion is a function of the ratio of even to odd harmonics. The more symmetrical the clipping, the more odd and less even harmonics." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/67091-

v-10-triode-hardness.html#post826877)

- "The default hardness value is based on the tubes that were in the amp being modeled. Most tubes fall in the range of 8 to 9. Perhaps old Mullards or Gold Lions or whatever are softer and would be equivalent to 5 or less. I don't know, I've never tested any. In general lower values will sound softer (naturally) but have less note separation. Higher values will give a more aggressive distortion and better note separation. There are no rules. Adjust the value to your personal preference. I doubt a real tube would ever be able to get to a value of zero but that doesn't mean it isn't a useable sound." source (http://forum.fractalaudio.com/news/74402-axe-fx-ii-firmware-v11-01-new-preset-banks-3.html#post910797)
- "If you are right on the edge of breakup the triode hardness is very powerful as it controls the harmonic series. Higher values will cause the overtone series to have a less steep decay and will increase perceived "sparkle". Together with the preamp bias you can control how chimey and "round" the tone is (preamp bias effectively controls the ratio of even/odd harmonics)." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/74457-more-gain-11-01-a.html#post911002)
- "Triode Hardness at zero gives a smoother distortion with reduced upper harmonics. However if you carefully compare a real tube preamp with the Axe-Fx models you'll clearly hear the difference as you reduce Triode Hardness. It's even more apparent when you compare the distortion spectrum. It's yet even more apparent when you use measurement techniques that learn the proper value." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/76826-downgrading-fw9-4.html#post938344)
- "The primary controls to adjust the saturation behavior are Preamp Tube Type, Preamp Hardness and Preamp Bias. (...) Preamp Hardness allows you to adjust how soft the saturation is." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/97925-suggestions-what-youve-done-enjoy-18-03-a-6.html#post1178319)
- "You can control the "shape" of the preamp and power amp distortion. The Preamp Hardness parameter controls the shape of the triode emulations. The lower the value the softer the distortion. The Power Amp Hardness controls the power amp clipping but that often is not noticeable because negative feedback around the power amp makes the distortion harder. Therefore you can make the power amp distortion softer by reducing Negative Feedback. A good example of this is a JCM800. A JCM800 has very hard preamp distortion (since there is no cathode bypass cap on the last stage) but has low negative feedback which softens the power amp distortion. The trick with that amp is to get the amp into the sweet spot by increasing the MV until you are getting some power amp distortion which softens the preamp distortion. source (http://forum.fractalaudio.com/axe-fx-ii-discussion/100622-scratchy-sound-patch-same-you-2.html#post1207496)
- "There are two primary parameters associated with our preamp tube models. "Preamp Hardness" determines how abrupt the tube clips when it enters the saturation region. There is another parameter that determines how nonlinear the tube is between cutoff and saturation. This is currently not exposed to the user." source (http://forum.fractalaudio.com/threads/anyone-not-a-fan-of-quantum-6.121353/page-3#post-1446007)
- "0.00 is not symmetrical. In fact there is no value that is symmetrical." source (http://forum.fractalaudio.com/threads/preamp-tube-type.122604/#post-1459051)

PREAMP SAG

- This parameter allows turning Preamp Sag modeling on or off. Turning it off replicates the behavior of separate preamp and power amp. Turning it on replicates the behavior of an integrated tube head or combo amp.

PREAMP TUBE TYPE

- Firmware 18.04: "Added “Preamp Tube Type” parameter to Amp block. “Modern” (default) selects a triode characteristic representative of modern production tubes. “Vintage” selects a characteristic typical of tubes produced in the 50’s and 60’s. “Long Plate” replicates the softer saturation characteristic of so-called “Long Plate” triodes."
- Firmware 18.06: "There are now only two Preamp Tube Type options in the Amp block: Short Plate and Long Plate. The Vintage type has been removed. Short Plate is similar to the previous Modern model but has the improved saturation characteristics that were developed for the Long Plate model."
- Firmware 18.08: "There are now four Preamp Tube Types:
  - Short Plate: an accurate model of a modern “short plate” production 12AX7.
  - Long Plate: an accurate model of a classic “long plate” 12AX7.
  - Modern: An idealized 12AX7 model. This is the same model from version 18.04 firmware. This model is useful for high gain tones that can benefit from the increased clarity and string separation.
  - Vintage: Another idealized 12AX7 model also from version 18.04 firmware. Has a softer breakup than the Modern model. Useful for “vintage” tones where more base nonlinearity is present."
- Firmware Quantum 1.00: "New RTS triode models. There are three new triode models based on our new algorithms: 12AX7A (default), ECC83 and 7025. The previous models are still available and may be selected with the Pre Tube Type parameter."
- Firmware Quantum 2.01: "Further improvement of preamp tube models based on measurements. The existing theoretical models, i.e. “Modern”, “Vintage”, etc., have been removed. There are now six extremely accurate preamp tube types: 12AX7A, ECC83, 7025, 12AX7B, ECC803 and EF86. Note that the EF86 type has been normalized to have roughly the same gain as the triode types."
- Firmware Quantum 6.01: "Updated all models to use an appropriate Preamp Tube Type when selected. I.e. British models will now use the ECC83 when selected."
- Cliff's comments:
  - "The primary controls to adjust the saturation behavior are Preamp Tube Type, Preamp Hardness and Preamp Bias." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/97925-suggestions-what-youve-done-enjoy-18-03-a-6.html#post1178319)
  - "The parameters for the 12AX7 were extracted from an RCA 12AX7A. The ECC83 was a Mullard." source (http://forum.fractalaudio.com/threads/if-12ax7-and-ecc83-are-the-same.110515/#post-1323322)
  - "The 7025 was a Sylvania." source (http://forum.fractalaudio.com/threads/if-12ax7-and-ecc83-are-the-same.110515/#post-1323463)
  - "The preamp modeling in 6.00 is the same as 5.xx except the parameters for the default tube type (12AX7A SYL) are different. The Sylvania 12AX7A is more nonlinear than other 12AX7As which results in more dynamics but will also result

in more "background" distortion because the waveform is being distorted even when it isn't being clipped. The JJ version is more linear which will result in a tighter tone and less background distortion but less dynamics. For 6.01 I've also added back the old 12AX7B type which is the most linear of the types and clips hard. People who play with lots of gain tend to like this as it results in tighter tone and more aggressive harmonic content. There are two primary parameters associated with our preamp tube models. "Preamp Hardness" determines how abrupt the tube clips when it enters the saturation region. There is another parameter that determines how nonlinear the tube is between cutoff and saturation. This is currently not exposed to the user but I've been contemplating adding it. I've also changed the default type for British amps to the ECC83 model as these amps typically were equipped with ECC83s (duh). The ECC83 was the European equivalent of the 12AX7A and tended to be a bit more linear and clip a little harder." source (http://forum.fractalaudio.com/threads/anyone-not-a-fan-of-quantum-6.121353/page-3#post-1446007)

PRESENCE FREQUENCY, DEPTH FREQUENCY

- Sets the frequency range of the Presence and Depth controls.

SATURATION SWITCH, SATURATION DRIVE

- Enabling the SAT switch decreases power amp smoothing which results in meaner distortion. This switch is enabled by default in certain models such as Cameron ch. 2 amp. Try it with amp types such as Plexi, JCM800, Friedman and Mesa Mark.
- Firmware 15.01: "Amp block Sat Switch now has three settings: Off, On (Auth) and On (Ideal). On (Auth) replicates authentic saturation circuit behavior and will lower the volume out of the virtual preamp. On (Ideal) replicates the idealized behavior present in Version 14.xx and earlier firmware."
- Firmware 18: "The parameter Sat Drive controls the amount of "saturation" when the Sat Switch is on."
- Cliff's comments:
  - "It switches in a zener diode clipping stage right before the tone stack. This is the so-called Jose Arrendondo Mod." source (http://forum.fractalaudio.com/axe-fx-ii/35997-axe-fx-ii-technical-questions-thread-76.html#post494304)
  - "The Sat (saturation) circuit is located between the preamp and power amp. If the model doesn't have much preamp gain, e.g. 59 Bassguy, then the sat switch will have little effect. A real amp would exhibit the same behavior. Amps like this get all their distortion from the power amp." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/87420-may-stupid-questions-fw15.html#post1058088)
  - "Sat switch works on all amps. However if an amp is getting its distortion from the power amp, i.e. Plexis, etc., then the affect may not be noticeable since the power amp distortion will mask any distortion occurring prior. Sat switches are typically employed on amps with master volumes for just this reason." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/104357-saturation-switch-does-nothing-authentic-most-amps-not-g3-yet-maybe.html#post1248576)
  - It always works. If you turn down the MV its affects may be more noticeable. Be aware that a Plexi has low preamp gain and you may need to set Sat Drive quite

high to get any saturation. The classic "Arredondo mod" involves adding a "saturation circuit" and a master volume as the sat circuit's affects are masked if the power amp is heavily distorted." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/104357-saturation-switch-does-nothing-authentic-most-amps-not-g3-yet-maybe.html#post1248582)

- Sat Drive controls the amount of "saturation" when the Sat Switch is on. The value is different for each model.

SPEAKER DRIVE

- This parameter models speaker overdrive. It interacts with Master Volume. If you crank it, you'll get the sound of a blown speaker.
- Firmware 6.02: "Improved speaker overdrive modeling in Amp block. New algorithm captures the “throaty” sound of an overdriven speaker along with the gentle compression. The “Spkr Drv” (Speaker Drive) parameter has been moved to the Spkr tab on the Amp menu.
- Firmware Quantum 7.00: "Changed default value of Speaker Drive to 0.5. When selecting an amp model the Speaker Drive parameter will now default to a value of 0.5 which is commensurate with a small amount of speaker breakup. Adjust this value to taste, if desired. If using the Axe-Fx II with a power amp and conventional guitar speakers you may want to reduce this value as the guitar speakers will impart their own distortion."
- Cliff's comments:
  - "The range of the speaker drive parameter is far greater than you would be able to push any real speaker before it self-destructed. If it doesn't sound good set that high, simply turn it down." source (http://forum.fractalaudio.com/axe-fx-ii-bugs/48341-weird-speaker-drive-5-07-a-2.html#post632608)
  - (about using Speaker Drive with a traditional cab) "I would say no. Your guitar cab is already distorting so you would be adding more distortion on top. As always, though, let your ears decide." source (http://forum.fractalaudio.com/threads/axe-fx-ii-quantum-rev-6-02-public-beta.122103/page-7#post-1454119)

SUPPLY SAG (MAINS IMP.)

- There are two dynamics controls for the power amp section. SUPPLY SAG controls how much the virtual power supply sags. This is a complex interaction between the master volume (MSTR), transformer matching (XFRMR MATCH) and screen network. Depending upon the amp you may even feel the screen voltage bounce if the screen network is underdamped (amps with chokes can often be underdamped). The screen network parameters are automatically set when the model is selected and cannot be altered by the user. DYNAMICS is an idealized dynamic range processor which controls the power amp response independently of the aforementioned parameters although it is still somewhat dependent on master volume. In general, the more heavily driven the power amp section, the more effect the SUPPLY SAG and DYNAMICS controls have.
- Turning Sag to 0 disables the entire power amp modeling for the Amp block in the preset. The Sag control has no effect at all when Power Amp Modeling is switched off in the Global menu. Cliff: "If you shut the power amp modeling off from the Global menu it is not exactly the same as turning it off by setting Supply Sag to zero. This is

because the virtual power amp always runs. So if you shut the power amp modeling off from the Global menu the supply will still sag resulting in a more compressed response. If the Master is set high the sag can be quite pronounced." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/82815-questioning-axe-fx-ii-4.html#post1004405)

- High values of Sag along with low B+ Time Constant values can cause “ghost notes” when the supply type is AC (as in a real amp). Lower B+ Time Constant values will make the amp feel “faster” but too low can cause ghost notes.
- Owner's manual about switching off power amp simulation: "In this mode, MASTER works as a simple volume, DEPTH is deactivated, and PRESENCE turns into a simple shelving filter".
- You use Supply Sag to simulate different rectifiers. Cliff: ""Reduce to simulate solid-state, increase to simulate tube." source (http://forum.fractalaudio.com/threads/rectifier-solidstate-vs-tube.102553/#post-1228660)
- Cliff's comments:
  - "Supply Sag models the power supply resistance. This includes the power transformer, rectifier and any other resistances before the filter caps. The higher the resistance, the more the supply droops when current is pulled from it by the power tubes. The more the supply droops, the spongier the feel." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/56648-im-addicted-supply-sag-2.html#post711961)
  - "If you turn off power amp modeling always check the Presence control. It changes from a "classic" control to a shelving type where 5.00 is neutral. I just spent an hour trying to figure out why this preamp model I am working on wasn't matching. Forgot to set the presence control to 5.00.” source (http://forum.fractalaudio.com/axe-fx-ii-discussion/45574-one-thing-always-check.html#post595595)
  - "Reduce to simulate solid-state, increase to simulate tube." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/102553-rectifier-solidstate-vs-tube.html#post1228660)
  - (Quantum 2.0) "The new power supply modeling is a significant improvement over previous versions. The power supply sag is now incredibly realistic and far superior to the "expander/compressor" techniques used in other products. Learning to adjust the pertinent parameters can make a good tone great. Supply Sag is the most fundamental of the power supply controls. It controls the virtual resistance of the AC input. In a real tube amp the supply sags due to a combination of power transformer resistance and rectifier resistance. Increasing Supply Sag increases this resistance and vice-versa. The higher the resistance the more the supply sags and the more bouncy and spongey the amp will feel. I like to increase Supply Sag a bit and reduce gain. You can monitor the virtual supply on the hardware by selecting the Supply Sag parameter. The gain reduction meter will display the supply voltage in dB relative to idle." source (http://forum.fractalaudio.com/threads/quantum-2-00-tips-part-1.110275/)
- Quantum 2.0 release notes: "For convenience the virtual power supply voltage (B+) can now be monitored on the PWR DYN tab of the amp block. When the Supply Sag control is selected the gain reduction meter will display the supply voltage in dB relative to the idle voltage."

TONESTACK TYPE, TONESTACK FREQUENCY,

TONESTACK LOCATION

- The tonestack is the set of tone controls on an amplifier. Use the ADV page in the Amp block to select a different tonestack.
- Firmware 6: "Reworked most tone stacks based on amp matching results. In general most knobs now behave exactly like the actual amp when possible. In a few instances there may be minor discrepancies between the knob position of the model and actual amp due to programming constraints and/or peculiarities of the actual amp (such as poor potentiometer tolerance). Due to variations in presence circuit topologies the taper of the Presence parameter, in particular, may vary between the model and the actual amp. In other words, a different setting on the model may be required to achieve the same response as the actual amp. In most cases however, the Drive, Treble, Mid, and Bass knobs will be accurate to within 10% of the actual amp."
- With some amp sims, such as the Lonestar, moving the tonestack location results in loss of volume.
- Cliff's comments:
  - "In most cases the knobs do translate. Usually within 10%." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/58613-axe-sounding-terrible-through-guitar-cab.html#post732551)
  - "Whenever you set the bass and treble to zero the tone stack becomes basically "flat" and the mid becomes a volume control. Most tone stacks behave this way." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/71226-interesting-find-eqs-zero-brit-pre-sounds-pretty-good.html)
  - "Every amp model uses one of these tone stacks. Obviously as there are more models than tone stacks some models share tone stacks. If you set the Tone Stack Type to Default, the amp block will use the tone stack appropriate for that model. For example, the Plexi 50W models use the "Plexi" tone stack (not surprisingly). If you set the Type to Default then the selected tone stack will be the Plexi tone stack. If you set the Type to Plexi it will be the same tone stack. The reason I did it this way is so you don't have to remember what the default tone stack is for the model. Simply set the Type back to Default. BTW, the tone stack is one of the main things that gives an amp its particular voice. People wax on about NOS tubes and "vintage iron" and cloth insulation and other nonsense but at the end of the day it's 99% frequency response. The tone stack shapes the frequency response pretty drastically. Many so-called boutique amps are nothing more than a classic design with a tweaked tone stack. The Axe-Fx II is unique in that it is the only modeling device that accurately replicates a tone stack along with the interaction of the controls and influence of surrounding circuitry. I had to solve the mesh equations for each of the major tone stack types which wasn't easy. A tone stack is a 3rd-order network and coding that was a real challenge." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/79267-when-change-tonestack-type.html#post965179)
  - "A Deluxe Reverb, for example, has no Mid pot but a fixed resistor. The value of that resistor is 6.8K. If you use a "Fixed Mid" tone stack the value of the virtual resistor will be 6.8K when the Mid control is at noon." And: "If you use a "Fixed Mid" tone stack the value of the virtual resistor will be 6.8K when the Mid control is at noon." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/98757-what-bf-fixed-mid-tone-stack.html#post1185050)

TREMOLO FREQUENCY, TREMOLO DEPTH

- Not supported on the AX8.
- This is a true bias tremolo and works by varying the bias of the virtual power tubes. The tremolo action is therefore different than other types of tremolo and the amount of tremolo varies with a multitude of variables, most importantly the tremolo is “self-ducking” and decreases at higher signal amplitudes. Note particularly that bias tremolo is a somewhat crude tremolo circuit and it’s interaction with the power amp depends on many things including damping, bias, etc. On some amps high values of bias trem depth can result in excessive crossover distortion. On other amps the amount of tremolo can vary greatly between loud and soft playing. All this, however, is part of the allure of bias tremolo as it results in a particularly “organic” sound. Control of the bias tremolo is afforded by the Trem Freq and Trem Depth parameters. A modifier can be attached to Trem Depth to facilitate engaging and disengaging the tremolo via footswitch or for other applications.
- When power amp modelling is disabled, the amp tremolo won't function.
- Cliff's comments:
  - "If the power tubes are being overdriven the bias tremolo can add lots of crossover distortion."
  - "Bias trem doesn't work well with all amp models. It depends on the model and this is precisely why bias trem wasn't offered on every amp ever made. It works best on amps that are biased hotter and that don't have much gain. Even a Deluxe Reverb doesn't work that well with bias trem because the power amp overdrives too easily. That is why an actual Deluxe Reverb uses an optical trem instead." source (http://forum.fractalaudio.com/axe-fx-ii-bugs/87474-65-bassguy-sv-bass-citrus-bass-fas-bass-usa-bas-1-2-bias-trmolo-bug.html#post1058577)
- "Bias trem works by modulating the power tube grid bias. One of the side-effects is that the effect becomes less pronounced as you play harder which makes it basically "auto ducking". Also since it's modulating the bias it gives an almost Univibe like effect since the phase changes a bit too." source (http://forum.fractalaudio.com/threads/patch-62-nuclear-tone-weird-tremolo-effect.122258/#post-1454602)

TRIODE1 PLATE FREQUENCY, TRIODE2 PLATE FREQUENCY

- Cliff's comments:
  - "It sets the cutoff frequency of the resistor/cap combination on the plate of the last triode stage (the previous stages are not user adjustable). Most amps have no cap on the last stage but a few do. You can vary this parameter to simulate increasing/decreasing the capacitor value. The frequency is only approximate since the actual frequency varies with the bias point/cathode impedance/drive/etc."
- Groovenut: "It will give you some control over the high harmonics that are created during clipping. The cap in question forms a low pass filter with the plate resistor on the triode stage. In English, it will allow you to control the buzziness that sometimes occurs with higher gain settings. It can also serve as a gain dependent tone control of sorts." source (http://forum.fractalaudio.com/axe-fx-ii/38144-amp-block-triode-plate-frequency.html)

- Firmware 2.0 and up also expose Triode1 Plate Freq. and Triode2 Plate Freq. Release notes: "This parameter sets the cutoff frequency of the plate impedance for the next-to-last triode in the chain. Many amps have a capacitor across this triode’s plate resistor. This capacitor is used to smooth the response and reduce noise. You can adjust the amount of capacitance, and the resulting frequency, using this parameter. The last triode plate capacitor is also exposed: Triode2 Plate Freq."

XFRMR GRIND (TRANSFORMER GRIND)

- Quantum 3.0: "Improved Amp block output transformer modeling. New model more accurately simulates dynamic core losses and leakage inductance. The “Xfrmr Grind” knob controls the intensity of the effect. Higher values result in more high frequency response and a more “open” sound. Very high values can yield a raspy, spitty tone common in vintage and/or low wattage amps. Modern “big iron” amps tend to have low values. Note that the audibility is dependent upon how hard the virtual power amp is driven and is more noticeable as the MV is increased. Also note that the effect in real amps is highly dependent on the speaker. Some speaker/transformer combinations exhibit significant high frequency dynamic boost while other combinations yield almost none. As always use your ears as the final determinant. Note: The Transformer Grind parameter will be set to a default value and the Dynamic Presence parameter will be reset to 0.0 for any presets created with previous firmware."
- Cliff's comments:
  - "Transformer Grind is what you want to get that top-end sizzle. Dynamic Presence is one of my "Inverse Homomorphic" filters and only approximates the dynamic presence boost found in some amps. Transformer Grind is an authentic model of what actually happens in those amps." source (http://forum.fractalaudio.com/threads/q3-01-to-2-04-conversion.114090/#post-1364689)
  - "Some amps interact with some speakers resulting in a dynamic high frequency boost. It creates an aggressive, biting distortion. It depends on the amp (the amp's output transformer in particular) and the behavior of the speaker as it deviates from it's rest position. You can simulate this using the Transformer Grind parameter." source (http://forum.fractalaudio.com/threads/raw-amp-sounds-anyone-know-how-to-get-them.121460/page-2#post-1445818)

XFRMR LF, XFRMR HF, XFRMR DRV (TRANSFORMER FREQUENCY, TRANSFORMER DRIVE)

- Cliff's comments:
  - "Transformer Drive models the core saturation in the output transformer. The Drive increases the amount of core saturation." source (http://forum.fractalaudio.com/axe-fx-ii/39352-does-transformer-drive-parameter.html#post535233)
  - "Don't overlook this when striving for "vintage" tones. I was playing around with this last night and it's very powerful in making edge-of-breakup tones sound like an old, well-played amp (if that's your thing). The higher you set the drive the more it saturates the virtual transformer's core. It doesn't affect the B+, that's done with the Sag parameter." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/41252-tranformer-drive.html#post553646)

    - "The size of the transformer is dictated by the necessary power handling. You can simulate smaller/larger transformers by adjusting the Transformer Drive parameter." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/63263-x-former-match-3.html#post784589)
    - "Part of the sound of certain tube amps, particularly those who derive their distortion from power amp overdrive, is attributable to the output transformer. The distortion produced by an overdriven output transformer isn't particularly pretty but it does play a role in the tone and without it the distortion would not be authentic. When a transformer is overdriven the iron core saturates. This happens because all the magnetic domains are aligned with the field and no more can be "rotated". In engineering terms the flux density (B) no longer increases linearly with the flux intensity (H). Since a transformer presents an inductance to the power tubes the flux intensity is inversely proportional to the frequency applied. Therefore the distortion increases at lower frequencies. Manufacturers frequently specify the frequency response of the transformer at its rated power. For example Hammond specifies most of its output transformers of having a bandwidth of 70 Hz to 15 kHz (re. 1 kHz +/- 1 dB) at rated power. The bandwidth of the transformer, however, will be much greater when operated below it's rated power. The Axe-Fx II allows the user to adjust the amount of output transformer saturation via a parameter called XFRMR DRV (Transformer Drive). Lower values decreases the amount of distortion, higher values increase it. The parameter is normalized to a rated-power lower-frequency cutoff of 40 Hz, i.e. a value of 1.0 means that the virtual output transformer will have a lower cutoff frequency (-3 dB point) of 40 Hz when the virtual power amp is operating at the rated power of the transformer. So, if the transformer has a rated power of 50W and the lower cutoff frequency is 40 Hz at that power, setting XFRMR DRV to 1.0 will duplicate that behavior. The formula for rated power cutoff frequency is simply D = f_c / 40, where D is the drive level and f_c is the desired cutoff frequency. For example if we wanted to duplicate the aforementioned Hammond transformer we would first need to find the equivalent -3 dB frequency which is roughly 3/4 assuming it's -2 dB at 70 Hz (since they strangely specify +/- 1 dB) which would be about 50 Hz. Plugging into the formula we get D = 50 / 40 = 1.2. As always use your ears. I personally prefer a setting of around 1.5 - 2.0 for clean-to-lightly distorted tones. I find it adds a bit of richness to the bass frequencies. For higher gain tones I prefer less as it can sound muddy. Note that the effect of output transformer distortion is highly dependent upon the how hard the virtual power amp is driven which is a function of Master Volume and overall gain. There are lots of strange things that happen with an OT saturates but those are trade secrets and I can't elaborate further." source (http://forum.fractalaudio.com/cliffs-notes/85592-transformer-drive.html#post1037637)
    - ""XFRMR LF" sets the low frequency -3dB point of the output transformer. Most transformers actually have a very low -3dB point (contrary to internet wisdom) however their full-power -3dB point is significantly higher. The Xfrmr Drive control sets the full-power -3dB point." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/96327-controlling-lows.html#post1154831)
- Firmware 10.10: "Transformer distortion modeling is now independent of transformer match value. Before the amount of distortion was also dependent on the match value making adjustment more difficult."

XFRMR MATCH (TRANSFORMER MATCH)

- This parameter controls power amp clipping. Similar to adjusting MV and it can be used to mismatch a real amp and speaker. Decrease to make the amps sound more broad.
- Firmware 3.0: "This is an extremely powerful parameter that sets the relative output transformer primary impedance which in turn controls how easily the power tubes are driven into clipping. The higher the Master Volume setting the more pronounced the effect of this parameter. Decreasing the matching causes the power tubes to clip later and therefore the phase inverter and grid clipping becomes more predominant. Increasing the matching causes the power tubes to clip sooner. At lower settings the speaker resonance will be more pronounced, at higher settings the speaker resonance will be less pronounced. For optimum results bring up the Master until the desired amount of power amp distortion is achieved, then adjust the matching until the character of the distortion is as desired. The various LF and HF resonance parameters interact strongly with this parameter so be sure to experiment with those as well when crafting your ideal tone. The value of this parameter is relative to the actual transformer matching which is set internally and not directly exposed. The value is reset to 1.0 whenever they amp type is selected."
- Cliff's comments:
  - "Very powerful control. Use in moderation. It changes the turns ratio of the virtual output transformer. Primary impedance is a function of turns ratio. As you increase the turns ratio you increase the impedance by the square of the turns ratio: $Zp = N^2 * Zs$. An easier description: Increasing Transformer Match -> Thick. Decreasing Transformer Match -> Scooped." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/48005-5-04-adv-parameters-xformer-match.html)
  - "One of the most powerful controls in the Amp Block Is Transformer Match. If you want a more "open" sound and feel, turn it down. If you want more compressed sound and feel, turn it up. A little goes a long way. Note that this control has more or less effect depending upon the setting of the Master Volume. Transformer Match has more influence at higher MV values and vice-versa. If you turn TM down, you may want to turn MV up to compensate and vice-versa. Turning it way up (around 2.0), for example, simulates the sound of running an 8-ohm speaker on the 4-ohm tap."
  - "Don't overlook this parameter when your MV is set high. It is extremely powerful. A little in either direction can make a big difference. If you want a more open tone, turn it down slightly. If you want more compression and sustain, turn it up a bit. This parameter is essentially a "turns ratio" for the OT." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/56382-transformer-match.html#post709437)
  - "Higher values are "warmer" but more compressed. Lower values are more open but harsher. Only small adjustments are needed. Transformer Match is the single most powerful advanced parameter when dealing with non-MV amps (i.e. when you have the MV cranked)." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/59934-warmth-6-0-9-0-a.html#post747355)
  - "The most powerful advanced parameter is Transformer Match. When people try different tube brands or rebias their amp to use a different type of tube they make all kinds of hyperbolic claims about those tubes but it isn't really the tube that

made the difference. Well it is but it's not because the tube is doing something special. It's simply because the tube has a different transconductance (gain). Amp designers choose an OT turns ratio such that the amp is "matched" to the load. However "matched" is a nebulous term since tube gains vary, speaker impedance is variable and bias point is adjustable. Therefore there is no absolute turns ratio that ensures perfect matching. Matching implies that the swing at the power tube grids just pushes the plates to the rails. If the output transformer is undermatched, the grids will clip before the plates hit the rails and vice-versa. Designers also select the turns ratio based on personal preference. Some designers prefer undermatched OT since this gives a more "open" sound, while others prefer overmatched since this gives more touch response. For example, a Trainwreck is highly overmatched. For a given OT, if the tubes have higher gain than originally then this effectively overmatches the OT and vice-versa. Now this matters most for non-MV amps that get their distortion from the power amp, i.e. old Marshall, Fender, etc. So... if you are going to experiment with any advanced parameter, start with Transformer Match. A little bit in either direction can make a big difference. Note that the Transformer Match parameter is relative to the internal value." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/61820-most-powerful-advanced-amp-parameter.html#post768301)

- "Transformer match has nothing to do with the physical size of the transformer. It is the turns ratio. The higher the turns ratio (higher Transformer Match) the higher the reflected impedance from the speaker and vice-versa. The higher the value the sooner the power tubes distort. The optimum turns ratio is such that the maximum power can be obtained. Tube amps tend to be slightly undermatched though since the speaker impedance is not constant. This varies with the make/model of amp and is encoded in the model data. The size of the transformer is dictated by the necessary power handling. You can simulate smaller/larger transformers by adjusting the Transformer Drive parameter." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/63263-x-former-match-3.html#post784589)
- "The internal default value is based on the amp that was modeled and an assumed speaker voice coil resistance of 0.8 times the nominal impedance. I.e., if the speaker is rated at 8 ohms the assumed voice coil resistance is 6.4 ohms. Some speakers are slightly below this, others are above. 16-ohm Celestion Greenbacks, for example, are about 12 ohms so that would 0.75 times the nominal impedance. To simulate this you would reduce the matching to 0.75/0.8 = 0.9375. If you find yourself lowering this value consistently then your Master Volume is too high (assuming it's a MV amp). If it's a non-MV amp and you still find yourself lowering this value then you'll probably find the tone harsh or too scooped at loud volumes. In general I find people set the MV too high on MV amps. I think they don't realize that most MV amps achieve full volume around 2-4 on the MV knob and then it's just compression after that. Amp makers are partly to blame here as they do this on purpose to make their amps seem louder than they really are. Of course the sweet spot is that point at which the power amp starts to compress so you want to set the MV high enough to get into the sweet spot. It's a psychological thing. People always like a more "open" sound even though they don't really understand what makes a tone "open". When you lower the Transformer Match you reduce the power tube compression of the lows and highs. The problem is humans naturally gravitate to this to the point that they will make the tone excessively "open" and

then it doesn't fit in the mix. I do not recommend deviating much from 0.9 - 1.1. Of course there are no rules. With real amps some people like that more open sound and achieve it by plugging their cab into the higher impedance output, i.e. plugging an 8-ohm cab into the 16-ohm jack. This would be equivalent to setting the match to 0.5. SRV liked it the other way round IIRC which would equate to a match value of 2.0." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/75161-fw-11-03-xformer-match-settings.html#post919805)

- "Guitar folklore has it that SRV and Joe Walsh intentionally mismatched their speaker impedance. I imagine others have done this. The general idea is you plug an 8-ohm speaker into the 4-ohm jack or vice-versa. The Axe-Fx allows you to replicate this behavior using the Transformer Match control. To simulate plugging an 8-ohm speaker into the 4-ohm jack set Transformer Match to twice it's current setting (i.e. 2.0). For the other way around set it to half (i.e. 0.5). And you don't have to worry about frying your OT." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/77944-simulating-speaker-impedance-mismatch.html#post949002)
- Tech NOte: "One of the most important "advanced tweaks" in the amp block is the Transformer Match parameter (XFRMR MATCH). This control sets the relative turns ratio of the virtual output transformer. Each amp model has a default turns ratio embedded in the model data. The Transformer Match parameter adjusts that ratio relative to this default value. Turning it down reduces the ratio and makes the transformer "undermatched". Turning it up increases the ratio and makes the transformer "overmatched". Why is this important? In a "classic" designed power amp the transformer turns ratio is selected to provide the full power the output tubes are capable of to the load. IOW, when the tubes are delivering their maximum current the voltage at the plates is near zero (the voltage swing is maximum) for the NOMINAL speaker impedance. If the transformer is undermatched the tubes won't fully saturate. If the transformer is overmatched the tubes will saturate early. Most guitar amps are slightly undermatched. This wouldn't really matter than much if the load were a simple resistance but a speaker is a reactive load. As mentioned in my other posts the impedance has a low-frequency resonance and a high-frequency boost. If the transformer were perfectly matched then the plate voltages would reach maximum excursion when the current was at a maximum only at those frequencies where the speaker impedance equals the nominal impedance. When the speaker impedance is greater than the nominal impedance the tubes saturate early. This has the effect of distorting the highs and lows before the mids. If you decrease the matching the highs and lows don't distort as quickly and the amp will sound more "open". However the distortion can be harsher since power tube current limiting is harder than voltage limiting. If you increase the transformer matching the highs and lows will distort sooner and the amp will sound more "compressed. The resulting distortion will be smoother due to the softer nature of the voltage limiting. As explained above a classic design selects the turns ratio to give the maximum power for the nominal speaker impedance. However the "nominal" speaker impedance is just that and the actual speaker impedance varies considerably from model to model. The impedance of a typical 8-ohm speaker can vary 20% or so. If the actual impedance is lower then this effectively undermatches the transformer and vice-versa. Furthermore the transconductance and maximum current capability of the power tubes varies. For example, the amp models in the Axe-Fx that use EL34s were modeled with Mullard tubes (as these are considered the

best sounding). A set of JJ EL34s will actually produce slightly more current and saturate earlier. This effectively overmatches the output transformer. Some amps deliberately overmatch the output transformer, i.e. Trainwrecks. This results in a smoother power amp distortion with more compression of the highs and lows. The amount of overmatching is considerable, typically about 50% Now, the effect of the transformer matching is only evident when the power amp is distorting. Non-MV amps get most of the distortion from the power amp so the effects of altering this parameter should be readily apparent. Master Volume amps get most of their distortion from the preamp so the effects of altering this control may not be as noticeable if the MV is turned down. Small adjustments can make a big difference. I typically never adjust more than 20% (0.8 to 1.2) and usually less than 10%. If you find your tones are slightly too open and harsh turn up the Transformer Match slightly. Conversely if you find your tones too compressed turn it down a bit. Be warned that turning it down may seem to sound "better" because the volume will increase (our old friends Fletcher and Munson again) but then when you play loud it will be boomy and harsh. You want some of those high and lows to be distorted early so that things aren't too scooped." source (http://forum.fractalaudio.com/tech-notes/98527-secret-weapon-transformer-match.html#post1181829)

- "Another factor which controls power amp hardness is Transformer Match. There are two primary distortion mechanisms in a power amp: grid clipping and plate clipping (PI clipping notwithstanding as this is only audible with a post-PI MV). Grid Clipping is extremely hard, almost a hard clipper (i.e. if(x>a) then x=a). Plate clipping is much softer. However most power amps are slightly undermatched which means the grids clip before the plates clip, but only at those frequencies where the speaker impedance is "nominal". At high frequencies (above 1kHz or so) the rising impedance of the speaker causes the plates to clip before the grids. At the low frequency resonance the plates also clip first. If you increase the transformer matching the plates will clip earlier and, since plate clipping is softer, the distortion will be softer. So turn up the Transformer Match and turn down Negative Feedback for softer power amp distortion. However... designers know all this and they design an amp to sound best in a mix (at least the good ones do). Soft clipping sounds great when you are playing by yourself but as soon as you are in a band context the sound gets lost since hard clipping helps cut through the mix. Amps designed for rock typically have harder clipping than an amp designed for blues or jazz. A 5150, for example, has an extreme amount of negative feedback which makes the power amp very linear and clips very hard. A Deluxe Reverb, otoh, has low negative feedback and large cathode bypass caps on the last preamp stage. This makes the clipping softer and the sound less "clear"." source (http://forum.fractalaudio.com/axe-fx-ii-discussion/100622-scratchy-sound-patch-same-you-2.html#post1207496)