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hyu/Upstairs-Run-9827 inaudiophile Summary Discover top-rated DSP software used by audiophiles Learn how users feel about EQ, effects, and parametric on WIN10 Insights into popular choices like Equalizer APO, Dirac Live, and FabFilter Pro-Q Equalizer APO and Peace GUI Many audiophiles recommend Equalizer APO and Peace GUI combo with REW for precise EQ settings. The ease of use and flexibility make it a top choice for WIN10 users. Dirac Live Users praise Dirac Live for its outstanding room correction capabilities. Its ability to fine-tune audio settings and optimize sound quality is highly valued among enthusiasts. Jriver Media Center and VST Plugins Some users opt for Jriver Media Center and VST plugins like FabFilter for EQ purposes. The combination offers a comprehensive approach to customizing sound according to individual preferences. FabFilter Pro-Q and MusicBee Those using FabFilter Pro-Q as a 32-bit VST under MusicBee appreciate its smooth integration and advanced features. It provides a professional-grade EQ solution for music lovers. FXSound's 9-band graphic EQ and ambiance effects are popular among users seeking unique sound enhancements. Despite minor preset issues, the software helps in refining audio output effectively. Additionally, headphone-specific convolution filters in Roon cater to headphones-only usage on PCs, ensuring optimized sound for immersive listening experiences. Whether you want to admit it or not, we're now firmly into the age of the amp modeler. Most modern guitarists are unwilling to lug a heavy 2x12 tube amp around with them these days, and recording a tube amp at home? Forget about it. The convenience of modeling now ranges from pro guitarists playing on the biggest stages and recording in the studio, to us regular guitarists recording and practicing at home. With the best Neural DSP plugins as part of your setup, you can get mix-ready, ultra-realistic guitar tones at the click of a button.I've been using guitar plugins for a long time, with my first foray being Line 6's POD Farm back in the noughties, and ever since then, I've been using them for the majority of my home recording exploits. Amp plugins make it much easier to audition sounds, tweak settings, and get them fitting nicely into the mix, and with the recent advances in modeling tech, they sound nigh-on indistinguishable from the real thing.I was provided access to all the Neural DSP plugins in this guide, so you can rest assured I'm not reading from the spec sheet here. I've used, recorded, and tweaked every single amp and effect of each plugin, ensuring that they're firmly put to the test to determine which is the very best for your particular use case.In my opinion, if you're looking for the utmost versatility in a Neural DSP plugin, you should go for Archetype: Nolly X. For those who prefer their tones on the heavier side of the spectrum, then I'd definitely recommend checking out Archetype: Gajira X.Neural offers two-week free trials for each of their plugins, which is a great way to audition them before you commit to buying one. I've included sound samples for each plugin so you can listen to how they sound in the context of a mix as well as with solo guitar sounds to help you narrow them down. If you want to learn more about Neural's plugins before you buy, check out our extensive FAQs section, and our how-to-use guide for more info.My top picksBest overallFor pure versatility, Archetype: Nolly X takes some beating (Image credit: Future/Matt McCracken)At a Glance Buy if you want a single plugin to cover a lot of ground: I found Nolly X to be the most versatile of all the plugins I tested, making it a great shout for the player who wants to play in lots of different styles. Avoid if you need easy preset saving: This applies to all the Neural plugins, but there's no easy way to favorite plugins which can be annoying.Overview: Of all the guitar plugins I tried out for this guide, I found Neural DSP Archetype: Nolly X to be the most versatile. Whether you want pristine clean tones, mid-gain riffing, or searing leads, it has the versatility to cover everything that you need. With four amp models and seven effects (including the EQ section), it may not sound like much but you can cover a tremendous amount of ground with just this single plugin.Usability: The interface is incredibly easy to use with everything clearly labeled for you. Even if this is your first plugin it won't take you long to get up and running. Along the bottom, you have the ability to mix and match your amp and cab models, with them running left to right in order of dirtiness. This means you can quickly mix and match for your base level tone before diving deeper into the amp settings.Along the top, you have separate options for your pre-amp stompbox section, amplifier, cab, EQ, and post-amp effects. Before the amp, you have your usual selection of compressor, two overdrives, and interestingly, a delay stompbox for some more ethereal sound exploration. After the amp, you can choose from a single delay and reverb. It doesn't sound like much on paper, but sometimes having fewer options gets you to where you want to be much quicker.Sounds: The sounds here are simply superb, and whether you're playing funk, rock, metal, blues, country, or anything else, you'll be able to dial in a tone that suits you. The four different amps each have their own character and there's plenty of range within each. So you can get cleanish tones from the first two models as well as more gain-y sounds if you turn up the gain or add some overdrive in front.The presets are fantastic too, which is where I encountered the first real issueIt seems like this plugin leaves something to be desired when it comes to saving your favorite presets within itself. You'll need to create folders on your computer, drag or drop the presets in, rename them if needed, and then find their correct place again. This can be frustrating for those with a lot of presets to manage. Neural DSP Archetype: Gajira X is perfect for modern metal players, offering classic or modern metal tones that are hard to replicate elsewhere. It's designed by one half of the guitar duo known for defining modern metal tone and covers everything from clean to heavy sounds. The Neural DSP Parallax X plugin offers an endlessly versatile sound for bass guitarists, with a single amp and cab selection alongside compression, two flavors of distortion, and an EQ section. Although the user interface can be a bit mystifying at first, each section has numerous options for tweaking your sounds. The cab screen allows you to miked up with varying condensers, dynamic, and ribbon microphones alongside the ability to load your own custom IRs. Rabea X: A Versatile Plugin for Crushing Djent Tones The versatility of studio-grade microphones, such as those used in professional recording environments, can be effectively emulated using different microphone types. In actual studios, engineers would experiment with various mic configurations to achieve desired sounds. By utilizing your own cab sims, you'll have more flexibility in creating unique tones. However, keep in mind that Neural plugins might not work optimally with third-party IRs, so be prepared to evaluate all options if you have a large collection. When analyzing a favorite guitar tone, it's essential to consider the role of EQ and compression. Typically, starting by cutting lows from 60 Hz and highs from around 14 KHz helps eliminate unwanted rumble and harsh high-end frequencies, resulting in a more natural amplifier tone. Compression is another factor to explore, as it can help control playing dynamics, especially when mixing multiple guitars together. FAQs are crucial for users of Neural DSP plugins. To run these plugins, you'll need a computer or laptop with specific minimum specifications: a MacOS Intel Core i3 Processor (i3-4130 / i5-2500 or higher) paired with Apple Silicon (M1 or higher), 8GB of RAM, and macOS 11 Big Sur (or higher). Similarly, Windows users require an Intel Core i3 Processor (i3-4130 / i5-2500 or higher), AMD Quad-Core Processor (R5 2200G or higher), 8GB of RAM, and Windows 10 (or higher). Neural DSP plugins are a one-time purchase, and once authenticated through your iLok account, you can access them across three different computers. However, these plugins do not require a DAW, as each has a standalone mode for easy practice. When setting up Neural DSP plugins, it's crucial to find the optimal audio buffer size. This setting affects latency and should be set as low as possible while still avoiding clicks, pops, or crackles. Oversampling can improve sound quality but requires increased CPU processing power. To test Neural DSP plugins effectively, you'll need a powerful machine capable of handling increased processing demands. When testing the plugins, I used Pro Tools on a MacBook Pro M3 Pro with Yamaha HS5 monitors and Beyerdynamic DT 990 Pro headphones. When building your own audio setup, consider creating a 2.0 or even 2.2 configuration for each speaker box using active crossovers. Tweeter, highs, mid, low, and sub stages can be employed, although it's not strictly necessary. For practice purposes, it's essential to explore available options and understand the required hardware and software. I'm setting up a home audio setup and want to get good results. My main computer is a Windows PC, but I have extra computers available if needed. I've found some hardware that might work: A USB Sound card from StarTech which could be low quality. Maybe a PCIe sound card like the Creative Blaster Audigy Performance Headphone would help instead? For amplification, I've looked for an 8 channel amplifier but only found this 12-channel Dayton Audio MA1240a Multi-Zone Amplifier. Would using a surround receiver with more channels work? One that has HDMI input could even skip a dedicated sound card since you don't need analog outputs. Additionally, newer PCs have decent on-board multi-channel audio so you might not need a dedicated sound card if your PC is well-designed. I'm also considering cables - four 3.5 MM to RCA adapters from Amazon Basics should work fine. As for microphones, the miniDSP UMIK-1 Omni-directional USB Measurement Calibrated Microphone would be a good choice. For software, REW and Peace equalizer seem like they'd do the job. Are there any other recommendations or essentials I'm missing besides speaker wire and speakers themselves? One important thing to consider is volume control - using the PC's volume control can introduce noise if you're not careful. A sound card with low noise would be a good idea in that case. I've heard great things about Asus sound cards, like the Essence STX II 7.1, but they're pricier. I also have some experience with active crossovers and think it's essential to consider path differences between drivers when using multi-way speakers up close. A surround receiver can be a good option if you don't need analog outputs, and newer PCs' on-board audio is getting better. The Behringer 204hd, Minidsp 2x4hd, and Asus xonar u7mkii sound cards are all options I've tried - the U7mkII stood out for its absolute quiet performance. For equalization, I've used Equalizer APO half the time and Pulse Xoverrack the other half. They're both great and free, but I prefer the Pulse Xoverrack due to its ease of use and customization options.so you want to know if its possible to have other DSP filters/effects on top of the crossover like room eq, time correction and summing low passed portion of the crossover to mono for sub? the thing is that yes you can do multiple EQs/filters/delays as you like but you cant make pulseaudio handle multiple soundcards well. its better to get a nice multi-channel soundcard so all channels are perfectly in sync. time correction makes sense if theres zero time dependant skew between channels which means no delay changing over time... but you can always test it with second soundcard report back cheap and easy as for other questions: its possible to sum output for sub to mono, thanks for reply. sorry maybe i didnt use correct terminology with regards to time. i was referring to time alignment so the sound from speakers and sub reach listening position at same time. either way sounds like this does everything you need saved me £250 from buying miniDSP that do same job. last edited: 2020-06-10 10:21 am well if sub lags behind then add delay to tweeter so its time aligned, but will not work properly if two outputs for sub/tweeter dont have fixed delay between them at any time like living on different sound cards / DACs driven by different clocks. in recording studios this problem is solved by using some kind of clock synchronization aka "Word Clock" distribution. so my advice would be to get decent multi-channel sound card well if sub lags behind then add delay to tweeter so its time aligned, but will not work properly if two outputs for sub/tweeter dont have fixed delay between them at any time like living on different sound cards / DACs driven by different clocks. in recording studios this problem is solved by using some kind of clock synchronization aka "Word Clock" distribution. so my advice would be to get decent multi-channel sound card. it just so happens i produce music and have interface that does just this so can test full solution with studio interface before considering getting another DAC for more permanent set up. either way though your software looks fantastic cant wait to finish work and get tinkering this evening Page 2 so you say each time start playback might introduce different fixed delay between two cards? well pretty much cant recommend that anyways repeat myself once mor - get multi-channel soundcard or I2s interface if you want use own DACs. these are readily available. ill use rme audio interface from studio system it has more than enough analogue outputs cant wait start playing the alsa multi plugin splits multichannel stream to individual RAM buffers for each soundcard, but i do not think theres any way drivers can tell two devices start DMA transfers very same moment, plus each device has small FIFO between incoming stream supplied by DMA and outgoing stream to DAC clocked synced clock, there is no way tell two soundcards output internal FIFOs very same moment. a multichannel card reads whole frame i.e all channels in same DMA transfer samples for each channel held together all way I2S. good to know so for speaker crossover applications alsa multi card route not recommended then. I tried this out of curiosity and can confirm it drifts hell could literally hear around seconds worth delay between primary speakers one interface and sub another interface. if i play one track they both start same time but if skip pause then play one interface starts much later other and you can hear all over place, obviously specific time difference depends on interface combination being used but its definitely not usable with multiple interfaces. that being said im very impressed with results had this running perfectly with interface borrowed studio system. had basic 2.1 set up going using HPF LPF added some EQ and was really impressed with results. ive ordered used xonar u7 because it was only £40 so can take studio interface back. going to use software part of permanent set up. do you have any suggestions how might EQ correct for sub? can take measurements with calibrated mic and use REW room eq wizard software measure peaks troughs say below 100hz created by subs interaction with room. but is there way import this into your software apply Looking forward to seeing everyone at the meeting tomorrow and discuss our strategies. I'm aware this can be done with the software you recommended, it's a gamechanger for me, and saved me money from going down the route of using MiniDSP. I'll keep posting my progress and feedback as I integrate it into my main setup. Several seconds are way too much (many magnitudes), there must be something setup wrongly. Devices linked by the alsa multi plugin are started/stopped/forwarded in one for loop, the difference should be just a few samples at most. IMO less. Such setup has been used for years e.g. /quicktoots.linux-audio.com/toots/el-cheapo/. It was up to a single second, not several seconds, sorry if i hadn't made that clear. You shouldn't show me things like that, I've only just got my latest amp build off my desk. I dont need another DIY thing to try. My current focus is to get the listening room setup finalised. One question, I'm relatively new to Linux, how do i set the bit depth and sample rate of my interface in Linux? Do you have any suggestions about how I might EQ correct for my sub? PAXOR can show and work with an FRD file. There will then be three curves... the *frd, the filter response and the (combined) result. One basic approach could aim for a flat result curve by applying filters. Further to this, in creating the reference file you will probably take multiple spatial measurements for sub usage and average them somewhere else. You may also come to the decision to use further EQ, which you can either combine into the final reference curve, or apply independently. You can use a screen trace to capture a result and make a further target. If you want to separate your standard EQ from crossing, you can cross blind after combining the standard rolloffs with your target file, but don't do this for higher Q effects for accuracy. You'll want to confirm phase/timing later using separate measurements. Sorry to repeat my question, but you guys seem like you'd know the answer. How do I set the bit depth & sample rate of an interface in Linux. When I google it I seem to get conflicting answers. I'll look for that tonight. I did see on stack overflow changing the config file was mentioned, then there were conflicting responses saying that wasn't correct in new versions of Ubuntu, the tread seemed to descend into an argument, which was confusing for a linux noob like me. Thanks mate, I'll check it out and look for the appropriate settings. First I should say I'm on Debian. These arguments might be for one of two reasons. Either there is a problem you need to avoid... or both ways will work but only one is proper/tdy/elegant. Not to sound dismissive, it is best to learn to do things right, but on the other hand you still need to learn and try things. Page 3 Perfect, thank you so much for you help and advice. The U7 turned up so I'm gonna have a tinker at lunch time. Do you have a paypal donation link on the website though? Do you think there would be any advantage if I had a pateron profile? So this software serves such a small niche market. Well that was a productive lunch break. The U7 worked first time It's running at 192khz as per your instructions So everything is spot on, thanks for your help You've got 30 euros too so have some beers on me cheers @futurama92! in total disregard of your request I decided to go with a White Russian instead of beer this evening. Btw - Futurama - best comic series ever, still in aw that it was canceled eventually! I do love a good white russian. I miss so many of my favourite bars and pubs during lock down. I cant wait for them to open again. Yeah, Futurama is hands down one of my favourite shows. I usually rewatch an episode or 2 every week, and have done for quite a few years so I'd estimate I've probably seen every episode at least 10 times. I think I can relate to it, as a kid all I wanted was to travel into the future and have a robot as a best friend. Also in my 20's I worked in a pizza shop @ Very nice, thanks! Did you actually use the support for FRD files in the software? Up to now it seems that nobody ever used that, at least I have no feedback whatsoever. FYI: The link to open the image does not work. Yeah some weirdness with google drive links, I've found forums dont like them, I'm not sure of a workaround. I haven't used the FRD files yet, I'm still waiting for my measurement mic to arrive, but I'll keep you posted. For now I just tested the setup with pink noise and a spectrum analyser on my phone. I found the ideal cutoff for the sub where I could see the speaker bass response falling, then I level matched the speakers and sub. just something as simple as that has made the world of difference. I'll share further adjustments when the measurement mic arrives and I'll get to grips with REW. I have a technical question. How are levels and clipping handled by your software? If i apply an EQ boost for instance, should i lower the overall output gain of that EQ to prevent clipping, or is there internal headroom or something within the processing? In my setups I adjust the overall gain so the gain curve of the ##Using float32le as the sampling format for pulseaudio server eliminates clipping issues until the final resampling step. Monitoring levels and peak detection can be done at various stages in the filter chain, reducing guesswork. S24le is suitable for 24-bit interfaces, but 32-bit may cause stuttering with high sample rates. LADSPA plugins internally work with floats, making float32le a recommended choice.

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