Review: Latest PSP Audioware Plug-ins
(MasterQ, EasyVerb, MixPack 1.7 Update)
by Rick Paul - *27th February 2004 -*

Back around mid-2001, I'd been recording, editing, and mixing on the computer for maybe 2-3 years. Initially, my solution was a hodgepodge of audio tools alongside Passport Designs' MasterTracks Pro sequencer. My move to the Cakewalk family of products, starting with Pro Audio 9 came near the end of 1999, after which point the integration of the audio and MIDI sides of my overall solution really started to pay off in greater productivity. Though I'd become extremely enamored with the computer-based environment for producing my recordings, in large part because I tend to be a very visual person, and don't lay claim to having anything near golden ears, there was a subtle (to me anyway) aural issue in my recordings. The best I could put my finger on it, they sounded a bit cold, or thin. Whether rightly so or not, I was beginning to attribute this issue to the need for a more "analog" quality. I'd been experimenting with various "warming" plug-ins available at the time, from Cakewalk's TapeSim and several shareware plug-ins that specialized in analog simulation to several function-specific plug-ins that had reputations for having a more analog character than the ones I was using at the time. While some of these experiments showed some promise, between budgetary limitations and the individual pros and cons of the various products I'd been considering, nothing quite addressed my needs.

It was during that period when I first learned of Poland's Professional Sound Projects (PSP) from Bill Stunt's June 2001 review of PSP MixPack and PSP StereoPack in *Recording* magazine. In discussing the MixPack bundle, Stunt had mentioned analog characteristics, and also dropped a few high-end analog processor names when trying to describe the characteristic sound of the plug-ins. However, the thing that really caught my eye was that, in his opening comments on MixPack, he said, "they were so good that I was forced to re-mix the whole CD using them." A short while later I downloaded the demo versions of the MixPack plug-ins. It took all of a few minutes to convince me that, whatever these plug-ins did, and however they might compare to other plug-ins or hardware processors, they imparted whatever subtle thing it may have been that I'd been missing in my computer-based mixes to that point. Call it "audio voodoo" if you like, but it didn't take me long to whip out the credit card and become a PSP customer.

While PSP may not be the most prolific plug-in developer out there, even as relatively small companies go, their product line has been augmented over the last few years, each time with plug-ins that have broken new ground and made the audio world take notice. Their PSP VintageWarmer has distinguished itself as an analog-style multiband compressor that can make almost anything that passes through it somehow sound better. Their initial digital delay plug-in emulates the classic Lexicon PCM 42 so well that Lexicon approved use of the Lexicon name for the

Lexicon PSP 42. The PSP 84 builds on the Lexicon PSP 42 design and adds additional functionality including a reverb unit that reproduces the sound of classic spring and plate units.

From the very end of 2003 to early 2004, PSP has introduced several new plug-ins in short order. In addition to filling some key functional areas new to PSP with the PSP MasterQ parametric equalizer and PSP EasyVerb reverb plug-ins, PSP has updated their classic MixPack line to improve sound quality, compatibility, and the user interface. A new multimode filter plug-in, PSP Nitro, has also been announced, but is not yet available as of this writing. In this review, we will take a look at the two new PSP plug-ins (i.e. MasterQ and EasyVerb), provide a high level overview of MixPack for those who may not already be familiar with those, and check out the enhancements delivered with MixPack 1.7.

As usual, since this is CakewalkNet, this review will be oriented mainly at users of Cakewalk products, especially SONAR, though it should be applicable to users of most DAWs. In particular, all hands-on product testing has been conducted using SONAR 3 Producer Edition.

Basics

PSP sent us three products for review:

PSP MasterQ 1.0.2
PSP EasyVerb 1.0.3
PSP MixPack 1.7

We'll get into more detail on each of these plug-ins (or plug-in bundles in the case of MixPack) below. For now, though, the short summary is that MasterQ is a high quality parametric equalizer, EasyVerb is a multi-algorithm reverb, and MixPack is a bundle of four plug-ins aimed at imparting analog recording characteristics on digital recordings.

MasterQ and EasyVerb are brand new products, introduced in very late 2003 and early 2004. MixPack has been around for awhile, but version 1.7 is a feature upgrade that brings several improvements (see below) to the table.

MasterQ is priced at $149, but registered PSP VintageWarmer users are eligible for a promotional price of $109. New customers who want both MasterQ and VintageWarmer can purchase the two-plug-in bundle for $249. EasyVerb is priced at $69. MixPack is priced at $149. Registered users of earlier versions of MixPack can upgrade to version 1.7 for $29.

All plug-ins covered here have VST and DirectX versions available for Windows (Windows 98SE and later), with RTAS versions for the PC planned for 1Q2004. For Mac, VST plug-ins for OS X are available now, with AudioUnits and RTAS plug-ins, also for OS X, planned for 1Q2004. The new plug-ins are not available for MacOS.

Software installation for all three products was extremely straightforward under Windows XP. The installer provides the option of installing the DirectX version, the VST version, or both. (I took both.) Options for picking non-default directories were available, and, after picking my choice of directories, all went flawlessly.

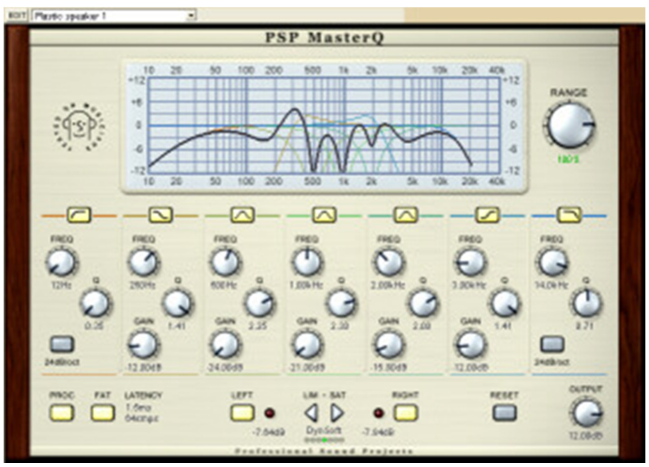
Software protection is via a combination of customer name and a serial number or authorization key. Since these are not system specific, as long as you save this information, there will be no issues later on if reinstallation is necessary, such as after replacing a hard disk.

Each of the products is supplied with a brief, but reasonably thorough, electronic manual in Adobe Acrobat (PDF) format. These manuals primarily serve as reference manuals for the controls provided in the plug-ins, as well as explaining the basic characteristics of any algorithms or special features. However, there are some additional materials. For example, the MasterQ manual also features a general discussion of equalization in mixing by *EQ* magazine's Craig Anderton, as well as a foreword by mastering engineer, Sakis Anastopoulos. The MixPack manual also features many block diagrams to describe the internal processing elements within each plug-in, response graphs where applicable, and general application-oriented tips. The tips don't quite go to tutorial level, but should be useful for getting a general idea of where to start with some of the more esoteric components. A generous helping of presets in each module also goes a long way toward getting users up and running quickly, not to mention providing some instant gratification.

My hands on testing of the PSP plug-ins focused initially on using the DirectX versions in SONAR 3 Producer. However, I did try out the VST versions, using Cakewalk's VST Adapter, also in SONAR 3 Producer, just to see if I would notice any differences. For most purposes, including automation of plug-in parameters and the ability to record automation using the plug-in interfaces, the DirectX and VST versions were essentially equivalent. The one key area of difference was with regard to automatic plug-in delay compensation (see the individual plug-in discussions below for more details).

Let's take a look at the individual products.

MasterQ



PSP MasterQ is PSP's new parametric equalizer. Sample rates from 44.1 kHz to 192 kHz are supported, and internal processing is 64-bit double precision floating point. These high-end specifications, along with MasterQ's name, might suggest an orientation toward pre-mastering activities. While MasterQ is suited to such activities, it is also equally usable for individual tracks and subgroups, though not the lightest weight of EQ plug-ins with respect to CPU usage. Thus, if you don't have CPU power to spare, you may not want to use it on every track of your 64 track mix, but might instead want to reserve it for tracks where its high quality and aural characteristics will be most useful.

MasterQ features seven fixed-function bands: low cut, low shelf, three fully parametric bands (nominally low, mid, and high), high shelf, and high cut. Frequencies and Q are sweepable for all bands, and gain can be adjusted for all except the high and low cut bands. The high and low cut filters can operate in either 12/dB per octave or 24/dB per octave modes, though adjusting the Q control for those bands, can change the slope of the filters. All bands may be individually enabled or bypassed.

A reasonably large-sized EQ graph display shows the graphs for both individual bands of EQ that are enabled, color coded by band to allow distinguishing which is which, and the net effect of combining all enabled bands. While the display can get fairly busy when many bands are enabled, this is a nice touch. In particular, it makes it relatively easy to figure out what is going on when you think one thing should be happening with the EQ curve, but what you're seeing isn't quite what you expected. Just look for a line on the graph that seems to be going in an opposing direction to what you thought should be happening, then find its band of EQ by color, and you've got your culprit. Note, though, that the graph only represents what is set by other controls. It cannot be directly manipulated.

MasterQ provides a range control to scale the EQ curve from -150% to 150%. You can think of this as a sort of exaggeration or understatement control. That is, if you turn the range up from its default of 100%, the peaks and valleys in the EQ curve get higher and deeper, respectively. If you turn the range down, anywhere between 0% and 100%, then, instead of steep mountains and deep valleys, you get rolling hills and less deep valleys. By using negative ranges, you can even invert the curve and scale it on the inverted side, so mountains become valleys and vice-versa. Note, though, that the effects of the high and low cut filters are left untouched. Thus, this control is really for dealing with parts of the EQ spectrum you are not rolling off. What is actually happening is that it serves as a multiplier for the gain controls. If you know the basic shape of what you want to do, this can be an easy way to maintain that shape while seeing what effect making more or less pronounced, or even inverted, adjustments might have. This can be a real timesaver compared to having to adjust each individual control to make similar changes.

One curiously named button available in MasterQ is labeled "FAT". No, that is not "phat", but rather an acronym for PSP's proprietary "Frequency Authentication Technique" algorithm. Say what? According to PSP, the FAT algorithm adds an octave above the Nyquist frequency (i.e. half the sample rate) and shifts phase and linear errors to that frequency region, then truncates that frequency region just before the plug-in's output section, thus removing phase and linear errors from the signal. The intention is to yield sonic results more like a high-end analog equalizer, rather than like digital filters. In practice, the difference between engaging the FAT button or not is subtle, but audible. PSP recommends leaving it engaged all the time unless you need extra low latency (it does add 128 samples of internal latency) or are running short on CPU cycles. Since I wouldn't expect a need to track through MasterQ, I wouldn't think the latency would be a big issue in practice. The sound quality difference is definitely worth having, so, if you do disable it to conserve CPU cycles while setting up your mix, remember to turn it back on before you render the mix.

MasterQ also has Left and Right buttons to allow processing the left or right channel independently, while keeping the other channel synchronized with the channel being processed. There is no provision, however, to process the two channels differently. Of course, you could simulate that by setting up two sequential instances of MasterQ, one for the left channel and the other for the right channel. A Proc (for "Process") button -- basically, the inverse of a Bypass button -- is also available to allow enabling or disabling MasterQ's effect directly from the MasterQ interface.

In addition to the equalizer components of MasterQ, there is a limiter/saturation component to prevent digital overs on MasterQ's output. The LIM-SAT control provides 7 different limiter/saturation algorithms in addition to no limiting/saturation. These include algorithms with a range of characteristics, some borrowed from PSP's VintageWarmer and other PSP plug-ins, to suit various types of audio processing needs. An Output knob adjusts the output level prior to the limiter/saturation module, so you can control how hard the equalizer portion of MasterQ drives the limiter.

SONAR 3 Professional users will already have a variety of equalizer plug-ins at their disposal, from the old Cakewalk Pro Audio Parametric EQ (not installed by default by SONAR 3, but still available on the SONAR 3 CD for optional installation) to the Cakewalk/DSP-FX FxEq that dates back to SONAR 1, to the new Sonitus:fx Equalizer added in SONAR 3 Professional. Users who migrated through SONAR 2 XL will also have the TimeWorks Equalizer. I might add that, in addition to these, my personal equalizer arsenal also includes Waves Renaissance EQ, Waves Arts TrackPlug, and, for pre-mastering purposes, the EQ component of iZotope's Ozone 3. One might ask why SONAR users would need yet another equalizer with all these choices.

That is a very good question, not only because of the obvious notion of whether it is worth spending additional money on yet another equalizer, but also because there is such a thing as an overload of choices. Sometimes having too many choices can slow you down as you wrestle with which specific choice makes the most sense for any given situation. Are the subtle, or perhaps not so subtle, differences that may be encountered worth the extra effort involved in determining which similar module is best for the situation at hand?

I won't get into detailed equalizer comparisons, but let me start by addressing the basic question. Even with all the choices the typical SONAR user will have, MasterQ is a worthwhile addition. It isn't perfect, and it isn't likely to be an EQ for use on every track of a multi-track mix, but it does have something special going for it. As best I can characterize it, that something special is a smoothness that goes beyond what I hear in other equalizers. I suspect the difference has to do with PSP's FAT algorithm because I do hear a subtle difference between when that is switched in and out. I also noted that MasterQ's high cut filter's cutoff point can go up as high as 30 kHz where all the others stop at 20 kHz or 22 kHz. The lower cutoff frequency makes sense if we look at the CD quality sample rate of 44.1 kHz and the general frequency spectrum for human hearing. However, I can't help but wonder if taking the range up further provides a level of interplay in the higher ranges that has some side effects in the lower ranges. This is one of the classic comments I have heard analog diehards make in discussing analog's supposed superiority over digital, and perhaps this aspect could also relate to why MasterQ might be characterized as sounding a bit more analog?

(PSP indicates that, when typical filters are used in the digital domain, there is no way to have a low pass filter running at higher than half the sample rate. This produces a situation where filter range characteristics are compressed to a very narrow frequency range and the minimum available cut is still very deep into the usable frequency range. Using the FAT technique allows going beyond half the sample rate, and the result can be seen directly on a graph of the filter's true response. For example, the filter's slope just slightly attenuates high frequencies when the cut off is set at 24 kHz.)

The differences are subtle in any case. The best I can suggest is to try the demo version of MasterQ to see if you hear a difference and whether that difference can be used to improve the sound of your mixes. What I can say, though, is that, in addition to hearing subtle differences myself, when I played my first mix with MasterQ to my colleagues at CakewalkNet for feedback, they unanimously commented that the frequency spectrum side of things somehow sounded better than my past mixes. While I think there were a few other factors at work on that specific project, I do think MasterQ contributed significantly to the observed difference.

MasterQ isn't perfect, though. The EQ response graph can't be directly manipulated like a number of competitive products (e.g. Sonitus:fx Equalizer, Waves REQ, TimeWorks Equalizer). There is no handy embedded FFT meter like the Timeworks Equalizer has. While it has more EQ bands available than most competitive products (all of those mentioned except Timeworks and Wave Arts), its band configurations are not as flexible as some of the others (e.g. Waves, TimeWorks, Sonitus:fx), preconfiguring the two end bands to be cut and shelving filters, leaving only three parametric bands in the middle. The hardware look of the plug-in has knobs that are somewhat difficult to distinguish from one another. Also, due to the layout of those knobs, I sometimes found myself grabbing the frequency knob, which is the topmost knob, nearest the response graph, when I wanted to adjust the gain of that band. As mentioned above, MasterQ is also more demanding of CPU power than most, if not all of the others -- the key possible exception' being Ozone, but Ozone is part of a multi-module mastering plug-in, not a separate EQ plug-in.

Of a bit more concern for using MasterQ as a track insert or on an aux bus is that MasterQ does not provide automatic plug-in delay compensation (PDC) for the DirectX version of the plug-in. It does, however, provide latency information in both milliseconds and samples. Thus, it possible to manually compensate for the latency by sliding tracks after rendering them to disk. The bad news is that, unless you have another plug-in to add elsewhere to do the sample compensation for other tracks, you will need to render to disk and manually slide the tracks, which can be a hassle. Thankfully, the VST version of MasterQ, configured via Cakewalk's latest VST Adapter, does provide automatic PDC. Thus, the simple workaround for SONAR users is to use the VST version of MasterQ whenever using the plug-in in a context where uncompensated delay could cause timing issues. (PSP indicated the lack of delay compensation in the DirectX version is a bug which they intend to fix in an upcoming update.)

The net is that MasterQ may not be the perfect all-purpose track insert equalizer, but its superb sound quality and characteristics make it an excellent choice for pre-mastering, mix-level processing, and featured track processing. While I had no way of testing its high sample rate support, I suspect that aspect of it could also make it especially suited for audiophile recordings, especially of solo acoustic instruments or small ensembles such as a string quartet.

EasyVerb

PSP EasyVerb is PSP's brand new reverb plug-in. As its name implies, it is designed to be easy to use. Probably the simplest way to get started is to select a preset that gets you in the ballpark from the 51 provided. If that doesn't quite do the trick, you can try various of the 9 reverb algorithms to see which you like best, then tweak the time and/or high frequency damping dials to taste. Finally, adjust the low and high frequency shelving dials (i.e. gain and frequency for each) if need be. There is also a wet/dry mix dial and an output level dial, as well as a Proc button to turn EasyVerb's processing on or off.

That's right, you won't find a zillion controls on EasyVerb -- for example, there are no controls for tweaking the high frequency damping curve, diffusion control, stereo width controls, early reflections versus reverb tails balances, or even pre-delay. That last one (i.e. pre-delay) seems like the most significant omission since it is often used to help lend clarity to a vocal or other signal where it may be desirable to use a fair amount of reverb but also to keep the articulation portion of the sound clear. However, if you really need that, you can always insert a 100% wet delay in-line before EasyVerb (i.e. in an aux bus configuration).

There are pros and cons to this approach, of course. Users who really know their reverb tweaking parameters, and perhaps are used to reverb plug-ins with an extremely large number of tunable parameters, such as SpinAudio's RoomVerb M2, or even Waves' Renaissance Reverb, Lexicon Pantheon, or Sonitus:fx Reverb, may feel a loss of control. In that case, though, perhaps it is best to consider EasyVerb to be "the reverb for the rest of us" (to paraphrase an old Apple Macintosh ad). To put it another way, EasyVerb's ease of use makes it harder for a less sophisticated reverb user to get into trouble.

Of course, ease of use wouldn't mean much if EasyVerb sounded bad. Thankfully, that is not the case. Also, don't let the simplicity of EasyVerb's interface fool you into thinking it is just a one trick pony.

EasyVerb's multiple algorithms make it a bit like getting 9 reverbs in one. Those algorithms are ambience, room, chamber, club, hall, arena, cathedral, spring, and plate. The EasyVerb user interface shows a diagram to give you a sense of the shape of the room (i.e. in the first 7 of the 9 algorithms), including sound source and mic positioning, or electronic device (i.e. in the last 2 of the 9 algorithms) being emulated. While these diagrams are fairly tiny and hard to see, the EasyVerb PDF manual provides slightly larger, more readable versions, and also provides textual descriptions of each room or device. For example, it may be useful to know that the difference between a "room" and a "chamber" has to do with room size, shape, and acoustics. The room is a mid-sized rectangular space, with the source material placed at one end of the room and the ambience mics at the back of the room. The chamber is a larger, more sonically rich space shaped more like a concert or echo chamber, with the source material placed at the front of the space and the ambience mics at the back. A "hall" might be described as a much larger version of a chamber, while a "club" would be a multi-room rectangular space, with the sound source on the "stage" and the ambience mics in the back of the larger room. All algorithms are true stereo -- i.e. there is no summing to mono before feeding the signal to the reverb effect.

Both the Time and Damp knobs are calibrated in percentages. For users trying to make surgical settings, for example to synchronize with a project tempo for reverb times or to damp from a specific frequency for the high frequency damping, this may not be ideal. However, if just using your ears, this manner of control should work just fine. Also, this type of setup makes it easier to ensure relatively natural sounding results. For example, PSP indicates that the most natural settings for a given room, or for emulating a specific electronic reverberation device, will be achieved when setting the Time dial between 30% and 70%.

A nice touch with EasyVerb's Mix and Output knobs is that their settings are not stored with EasyVerb presets, but those settings will be stored with projects. Thus, for example, if you are using EasyVerb on an aux bus, and thus switch the Mix control to 100% wet, you don't have to worry about the Mix control's being reset every time you try a different preset to audition different reverb sounds. This also avoids the whole quasi-religious battle between users who almost always use reverb plug-ins on an aux bus and those who almost always use them on a track insert, where each has decidedly different preferences for default wet/dry mix control settings.

The area of reverb plug-ins is another where SONAR 3 Professional users will have a number of choices. Not only are there the old Cakewalk Pro Audio vintage Reverb and the Cakewalk/DSP-FX FxReverb (based on DSP/FX's StudioVerb) that came with SONAR 1 and 2, but SONAR 3 adds the Lexicon Pantheon reverb, and SONAR 3 Professional also includes the Sonitus:fx Reverb. My own reverb collection adds SpinAudio's RoomVerb M2 2.0, Waves' Renaissance Reverb, Wave Arts' MasterVerb, and SilverSpike's Reverb-It, not to mention a few acoustic space simulators that aren't strictly reverbs per se, but can be used for similar purposes (e.g. Cakewalk's own FX3 SoundStage comes to mind). Choices, choices...

The thing about reverb, even more than with many other effects, is I find it to largely to be about flavors and tastes. As much as reverbs are all intended to add ambience and simulate either natural-sounding spaces or electronic devices, they all seem to do it somewhat differently, sometimes with markedly different flavors. This notion is compounded by the very different controls that many reverbs provide. Just try setting up the same basic program on even half the reverbs I mentioned above, and I can pretty much guarantee you'll end up frustrated before you get anything sounding remotely similar with those plug-ins. Come on, I dare you!

This makes it somewhat tough to compare reverb sound quality, especially when the basic algorithms two specific reverbs have don't have a lot of overlap. Even when they do, presets may be tweaked in wildly different ways. For example, comparing a few hall presets from EasyVerb to Pantheon hall presets might suggest that Pantheon is a much brighter reverb. However, raising the cutoff frequency of EasyVerb's high shelf filter in the EQ section got things more in the ballpark. But Pantheon doesn't have an equivalent EQ section, rather having a bass boost control that can have fractional values (i.e. such that it serves more like a bass cut), thus making it very difficult to compare apples to apples.

The bottom line here is that EasyVerb is pretty easy to make sound good, and, in fact, easy to use in general. For those of us who aren't diehard reverb tweakers, this may be a big advantage over more complex plug-ins, and it will be up to your ears to decide if EasyVerb sounds right for you (and there is a demo available).

I have only two real reservations about EasyVerb. First, I do wish it had a pre-delay in there. Having to add a delay plug-in to the chain for this common use seems to defeat the whole idea of keeping EasyVerb, well, easy. Second, EasyVerb is relatively demanding of CPU power. While this isn't unusual for good sounding reverb plug-ins, it seems significantly more demanding than all the other reverb plug-ins I tried, perhaps as much as 300-400% more demanding in some cases. This may not be a major issue for modern systems, but may be limiting for people with older systems. Also, it may well be that EasyVerb's simplicity could cause it to be in especially high demand among less sophisticated users, who may be less likely to have the latest greatest systems.

On balance, EasyVerb is dead easy to use, sounds great, and has a pretty reasonable price. That combination makes it an excellent value. Its ease of use may just help boost the productivity of those of us who want some flexibility for tweaking sounds, but are daunted by the number of parameters, and the number of ways to go wrong in tweaking those parameters, with more typical reverb plug-ins. That aspect of EasyVerb may just be the factor that makes EasyVerb become a first call reverb plug-in for many of my future projects, except, that is, on tracks where I would use a pre-delay.

MixPack 1.7 Update

PSP MixPack is a bundle of four plug-ins aimed at imparting analog characteristics on digital recordings. MixSaturator produces tube- and tape-type saturation. MixPressor is a soft knee compressor with sound characteristics similar to that of classic devices with tube and optoelectronic circuits. MixBass is a bass enhancer. MixTreble is a multi-function treble processor with capabilities aimed at both repairing and enhancing the signal.

For users who are not already familiar with the MixPack modules, we'll look at each of the modules in a bit more detail below. Existing MixPack users may wish to skip ahead to the "Version 1.7 Updates" section.

MixSaturator

PSP MixSaturator aims squarely at imparting analog-style saturation, such as is characteristic of tape recorders or tube-based circuits, to digital tracks. There are three key components of MixSaturator's processing. The main saturation algorithm allows choosing between 7 different non-linearity curve shapes (3 analog tape curves, 3 analog tube curves, and digital clipping) or disengaging the main saturation algorithm altogether. The bass processing algorithm, which can be disabled, can be used to add warmth to the frequencies below a specified frequency by adding harmonics and increasing (or decreasing) the amount of lower frequencies in the resulting audio. The treble processing algorithm, which can also be disabled, allows simulating tape compression in the treble range, with controls for adjusting the frequency above which the processing will be applied, the depth of compression, and the balance of the treble component in the overall signal.

VU-style meters provide average and peak level readings. The VU-style meters aren't for looks only, but rather are calibrated to work as real VU-style meters with 300 ms integration time and the VU meter's zero reference level set to -14 dBFS (with absolute digital zero -- i.e. 0 dBFS -- showing a reading of +14 on the VU meters' red peak needles). Pre and Post settings can be used to take readings either after MixSaturator's Input attenuation stage (but before any saturation processing) or at MixSaturator's output, respectively. If neither the Pre nor Post settings are engaged, then the meters reflect the drive levels of the saturation algorithm.

PSP provides 31 presets to get you started, and they are meaningfully named, and suitable for various track-specific applications. MixSaturator's controls are pretty intuitive, though. Thus, it is easy to tune a preset to your liking, or even to just create your own track-specific setting from scratch. The key to doing the latter is to decide which algorithm most suits the needs at hand, set the Drive control appropriately to get the saturation level in the ballpark, then set the controls for the treble and/or bass algorithms if needed.

When I first came across the PSP MixPack plug-ins several years back, I was feeling my digital tracks lacked a certain something. I wasn't quite sure what it was, though I had a feeling it had something to do with sounding "less digital" and "more analog". However, I'd tried various other tube and tape simulation plug-ins available at the time, and had not quite found that elusive "something". Plugging MixSaturator in seemed a bit like audio voodoo at the time. My tracks were instantly transformed in a way that brought them much closer to what I'd been seeking. It didn't take long to get out my credit card, nor to start seeing MixSaturator pop up in pretty much every project I did. Since that time, a number of other options have become available, but MixSaturator still finds frequent use in my projects.

MixSaturator is highly complementary to SONAR 3 and the plug-ins it provides. The closest thing available in the SONAR box is Cakewalk's FX2 TapeSim plug-in. However, TapeSim has only a subset of MixSaturator's functionality, and really isn't even in the same league with respect to sound quality and control. Perhaps the primary competition for MixSaturator in a SONAR environment would come from Antares Tube or PSP's own VintageWarmer. Tube does a great job for its highly specific purpose (emulating a tube preamp, with two tube models with very different characteristics), but isn't as far reaching as MixSaturator, and both products can be extremely useful. As for VintageWarmer, that is a great plug-in, but would be overkill, both in terms of functionality and in terms of CPU drain, for many situations for which MixSaturator would be ideally suited. In short, I would still classify MixSaturator as a "must have" plug-in.

MixPressor

PSP MixPressor is a soft knee compressor, plus optional limiter/saturator, with a decided focus on vintage-style characteristics. Rather than providing ratio and threshold controls, there is a single Compress slider to determine the amount of compression.

MixPressor's Attack and Release controls are augmented by a Hold control. The Attack and Release knobs, besides featuring manual settings, also each provide "fast" and "slow" automatic settings. The Hold control ensures that, when the compressor circuit is engaged, it is held open for at least the amount of time specified in the Hold control, thus minimizing artifacts when a short release time is needed with bass frequencies. A Make-up knob provides make-up gain. In addition to allowing manual gain settings, it includes an "auto" setting to let MixPressor determine the amount of gain needed to make up for the amount of compression dialed in.

An SCL (for "side chain listening") button allows keying the compressor off a portion of the signal around a specified frequency as determined by Freq ("frequency") and Q (bandwidth) controls. A Del (for "delay" button) activates an internal look-ahead delay to allow the compressor to reduce signal peaks during the attack portion of the sound. An RMS button determines whether the compressor is activated based on RMS or peak detection. A Sat/Lim toggle button determines whether the limiter/saturator circuit is engaged, and, if so, whether it operates as a saturator or limiter.

I'm generally not a big fan of VU-style meters because they tend to take up more user interface space than most other options for showing the same type of information. However, MixPressor's VU meters, which work like those for MixSaturator (i.e. in terms of emulating real VU meters), are a key exception to this due to their unique use of three separate needles for showing average level, peak level, and gain reduction all in the same interface. There are also LEDs available to show peak overloads (or, if the limiter is engaged, when it is engaged to the point of being audible). Pre/Post toggles are available to determine whether the meters show the signal after the input gain (but before the dynamics processing circuits) or at the output of MixPressor.

In the crowded world of compressor plug-ins, there are two basic categories of compressors. One is the relatively transparent compressor aimed at controlling dynamics while altering the character of the signal being processed as little as possible. The other is the compressor that is used as much for its character as it is for dynamic control. MixPressor is decidedly in the second camp. Call it "warm", "phat", or whatever "analog-ish" term you prefer, but the key is that MixPressor tends to color the signal to some degree, and generally in a pleasing way. Of course, you can also take advantage of this aspect for more extreme effects if that suits your needs, but MixPressor is also at home processing vocals and even full mixes. A Mix slider, which goes from fully dry signal to 100% compressed signal, makes it easy to use MixPressor to achieve the kinds of drum submixes that blend the dry signal with some amount of relatively hypercompressed signal without using an extra bus.

My biggest complaint about MixPressor is that it currently does not have automatic delay compensation. Because it also has variable internal latency depending on its setting, and does not let the user know what that latency is, this makes the user have to work a lot harder to compensate if MixPressor is used on extremely timing-sensitive tracks. Since the range of latencies we're talking about can be from at or near zero to approximately 23 ms, it is certainly possible to get into territory where this becomes a practical issue. This is especially the case when dealing with multi-level submixes (e.g. individually compressed background vocals, which are then fed to a submix bus where they are further compressed) and/or when dealing with additional plug-ins requiring manual delay compensation, such as MixTreble. While there are workarounds available, such as rendering individual tracks that use MixPressor then lining up the rendered tracks visually with the original tracks, the workarounds are not convenient, especially when there are many tracks involved. As a result, I have found myself using MixPressor considerably less than I otherwise might had I not had to revert to left brain mode to deal with manual delay compensation. (PSP indicates addressing this issue is on their high priority tasks list, but they do not yet have a forecast for when automatic delay compensation might be available.)

With so many compressors on the market, SONAR users generally have at least several choices in their plug-in arsenals. SONAR 3 Producer Edition itself comes with the excellent Sonitus:fx Compressor and its multiband sibling, Sonitus:fx Multiband. SONAR also includes Cakewalk's historical FX1 range of dynamics processors, and SONAR 2 XL users will also have Timeworks' CompressorX, which itself has some vintage characteristics. My personal collection of compressors also includes Waves Renaissance Compressor, Wave Arts' TrackPlug, dB-audioware's dB-D, and PSP's VintageWarmer, among others.

For me, the bottom line is that MixPressor's character provides another flavor, and that dictates when it is the best compressor for the job. I might add that, were it not for the manual delay compensation issue, the decision on which compressor to use would probably go MixPressor's way for at least a few tracks on every project.

MixBass

PSP MixBass is a low frequency processor that provides compression and harmonic enhancement to audio signals below a specified crossover frequency. Its controls are about as simple as they come. Simply set the crossover frequency, set the Threshold for the fixed ratio (2:1) compressor, dial in any Color (i.e. the level of harmonic enhancement), and turn the Bass knob to control the mix of the bass component in MixBass' output. A Saturation button is also available to determine whether soft clipping is used when the output level gets high enough. Finally, a Mix slider allows setting the enhanced component of the bass signal to be anywhere from non-existent to 100% for applications where it may be desirable to heavily process the bass signal but only mix a small amount of the processed signal in with the original signal.

One note with MixBass is that delay compensation is automatic with the VST version of the plug-in, but not with the DirectX version. (PSP indicates that they expected the DirectX version of MixBass should have had automatic delay compensation, and they are checking into why there is a difference in this area, with the intent of fixing the DirectX version as soon as possible.) While the MixPack manual suggests the internal latency of MixBass is a fixed 256 samples, my measurements showed a fixed 1023 samples (approximately 23 ms at 44.1 kHz) for the DirectX plug-in. Needless to say, this could affect timing, so it makes sense to just use the VST plug-in, even in SONAR. In practice, this worked just fine via Cakewalk's latest VST Adapter.

The basic purpose of MixBass is to allow beefing up bass signals, be it on individual tracks or a mix. It does this job well, and there aren't a whole lot of other plug-ins that cover this territory. Perhaps the closest is Waves' Renaissance Bass, which does provide harmonic enhancement, but is not as flexible as MixBass for fine-tuning the signal.

MixTreble

PSP MixTreble is a multi-function treble frequency processor. Those functions include hiss removal, transient restoration, stereo enhancement, and exciter-style harmonic enhancement. The various components of MixTreble can be selectively engaged in order to facilitate a number of diverse applications.

Don't let the name "Hiss Remover" fool you. While MixTreble's Hiss Remover component can be used for hiss reduction, it can also be used for a number of other applications, such as reverb reduction, and creative processing of percussive tracks. For example, I used it on some mixed acoustic drum tracks to "dry up" the tracks, reduce the cymbal content, and more closely approximate a drum machine sound. The Hiss Remover allows you to specify the minimum frequency above which will be considered the "hiss" portion of the signal, the signal level threshold below which the Hiss Remover to kick in, the speed of the filter's cutoff frequency, and the amount of attenuation to be applied. A general rule of thumb is basic recommended settings will make the Hiss Remover perform its named function, while more extreme settings will allow creative sound mangling.

The Transients component is geared toward trying to simulate the restoration of lost transients (e.g. from over-compression of the source material). A Ratio knob provides for continuous expansion ratio control between 1:1 and 2:1. You can also tune the filter slope, apply high damping to avoid overly bright signals, and adjust the balance of processed signal content in the output of the module.

The Enhancer module provides stereo field enhancement using a high-pass filter and a controlled XY->MS->XY matrix. You set the slope of the filter and the degree of enhancement. There is an LED to show when it is possible too much enhancement is being applied (i.e. based on a high level of the side component in the output signal).

The Harmonics module generates additional harmonics in aural exciter fashion. You set the filter's center frequency and Q (i.e. bandwidth), then adjust the Drive control to set the depth of the effect. There is also a First Out control, which allows removing all or part of the fundamental from further processing. Finally, an Adjust slider allows controlling the amount of the generated harmonics that gets mixed in with the output signal. The general effect is to make the signal more "exciting", though too much can make it "brittle" or "thin".

MixTreble is a fairly powerful processor, both for making individual tracks stand out more in a crowded mix and for restorative processing. For example, I used MixTreble on a mix that came off cassette. The Hiss Remover component allowed removing some of the cassette hiss, while the Transients module "undid" some of the tape compression and the Harmonics module brightened things up a bit. While the mix may not have ended up being quite as good as I could have done had I the original multitrack masters to remix within the digital realm, it was a major improvement over the cassette-based source material.

MixTreble also has the issue with uncompensated plug-in delay; however, it will depend on what modules are in use. According to the MixPack documentation, internal latency is 256 samples (i.e. approximately 6.2 ms at 44.1 kHz) with the Hiss Remover turned on, and zero otherwise. My measurements gave slightly different results in this area, but the difference was only a few samples. The obvious workaround here is to avoid the Hiss Remover section if uncompensated latency will be an issue in your specific application.

While there are a number of plug-ins available that do various portions of what MixTreble does, I can't think of a single one that does all those jobs. Nor does SONAR provide any plug-ins for these functions. Thus, MixTreble represents another highly complementary addition for SONAR users.

Version 1.7 Updates

The first, and probably the most significant, area addressed by the MixPack 1.7 update is sound quality. Sample rate support now extends to 192 kHz, 64 bit double precision floating point computations are now used throughout each of these plug-ins, and DC removal filters have been added at the plug-ins' inputs. PSP also indicates that computation precision has been improved in some of the algorithms used.

Unfortunately, I did not have the opportunity to check the older versions of the plug-ins against the new ones to directly compare sound quality as I'd uninstalled the old versions prior to installing the new ones. However, my general sense is that the newer ones may well be subtly smoother. Checking around various bulletin boards to see what other early MixPack update customers are saying seemed to confirm this impression.

A second area of enhancement was in the graphical user interface (GUI) of the plug-ins. PSP indicates they improved the GUI colors and improved the meters' needles, both for better readability. To be quite honest, I can't say I had any problems reading the meters previously, but the updated meters are certainly very readable. As for the "improved" colors, I have mixed feelings on these. On the one hand, the colors of the knobs in some of the plug-ins (e.g. MixPressor, MixTreble, and MixBass) help group related controls together. On the other hand, the user interfaces for these plug-ins already had dividing lines to accomplish that grouping, and the colors are extremely bright, making it harder to see the text labels for the knobs. Also, the knob colors in MixSaturator seem pretty random, or at least using a different color assignment logic than the rest of the plug-ins use. Check out the screen shots above in the discussion of the individual plug-ins to draw your own opinion.

PSP also lists compatibility improvements, mainly in the area of VST conformance and bug fixing, among the improvements made in version 1.7. Users should note that presets written with MixPack 1.6 or earlier will not load properly in MixPack 1.7 because a stereo/mono switch was removed from the plug-ins' parameter lists.

Notably missing from the list of improvements in version 1.7 is automatic plug-in delay compensation for all plug-ins. The VST version of MixBass does now support automatic delay compensation, and MixSaturator does not introduce any latency anyway. This leaves both MixTreble and MixPressor, as well as the DirectX version of MixBass, without automatic delay compensation, or even any indication of the plug-in-induced latency for the given setting. Hopefully PSP's understanding of users' priorities on this issue, and resolving any further technical problems that remain via their parallel experience with MasterQ, will result in a solution to this problem in the not-too-distant future.

On balance, the sound quality improvements made in MixPack 1.7 fairly easily justify the $29 upgrade price for existing MixPack users, even if it is disappointing that the plug-in latency issues were not fully resolved in this release.

Closing Notes

Back when I first became aware of PSP, their MixPack suite of plug-ins filled a unique, but widely useful, niche at a bargain price. It was a no brainer for me to add MixPack to my then relatively small set of plug-ins. A few years have passed, and my plug-in collection has grown sufficiently that I'd hate to have to count the plug-ins, or really even to estimate their number. Even so, MixPack still fills a relatively unique niche, and individual plug-ins from MixPack find their way into many, if not most, of my projects. In my book, that says something about value. The latest updates to MixPack enhance that value, perhaps in relatively subtle ways, but it's tough to improve upon a good thing. Okay, so I still have one key gripe with MixPack in its lack of automatic latency compensation for MixTreble and, especially, MixPressor. However, PSP's start on the latency compensation with the VST version of MasterQ, suggests they may have some good news on this somewhere on the horizon, and the one issue won't stop me from seeing strong value from MixPack in the interim.

While MixPack is still relatively unique overall, MasterQ and EasyVerb enter into already crowded markets, populated by everyone from the most established plug-in and hardware vendors to the smallest startups and even some freeware-producing hobbyists. Yet, even with the many plug-in equalizers and reverbs out there, MasterQ and EasyVerb each bring something unique to the table. In the case of MasterQ, it is that ultra smooth, or should I say "FAT", sound that raises the bar for plug-in equalizers. With EasyVerb, it is the combination of multiple high quality algorithms and ease of use. These attributes combine to make EasyVerb a superb first reverb for anyone just starting out. Perhaps less obviously, they also make it a good first call reverb for more experienced users when they're in a hurry or just don't feel like tweaking lots of parameters but still want to tailor the reverb sound to their needs.

PSP is one of the few plug-in vendors out there about whom it is possible to say that every product they've introduced has been a winner and, in most cases, a "must have". Those who haven't yet encountered PSP or their plug-ins owe it to themselves to check them out. For those who already are using PSP plug-ins, you probably won't be surprised to hear that MasterQ and EasyVerb continue the PSP tradition for providing quality and value, and the MixPack upgrade makes a good thing even better.

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