Ask Dr. REX!

"No, I don't use loops. I create my own music."

Heard it before? In all likelihood, yes. Oddly enough, the use of loops still remains a somewhat controversial issue. It's in the twilight zone, that fuzzy grey area between artistry and "pushing a button". After all, a loop is the product of someone else's efforts, talent and creativity. But then again, isn't that true for any kind of art? Aren't you using sampled instruments that have been recorded – and played – by someone else? And synth presets that someone else programmed? Who do you owe credit to when you play a violin trill on the NN-XT – the guy who sampled it? The conductor? The violinist? Antonius Stradivarius, who built it? The truth is, at the hands of a skilled artist a bunch of raw loops can be molded into a unique piece of artwork that holds its own against any other musical accomplishment. And never was there a tool more powerful for tweaking, bending, carving and re-shaping loops than Dr.REX!

In this opening chapter of *Discovering Reason* we will explore this versatile device. We'll start off with the simpler and more common applications, and as the article progresses we dig deeper until we reach the very bottom of Dr.REX's bag of tricks!

Note: The mp3 audio examples in this article are loops, and we therefore recommend that you configure your default mp3 player to loop/repeat playback mode.

Dr.REX - the Subtle Sizzler

Using a drum loop upfront as the main rhythm body is not necessarily the way to go. In fact, one could say – arguably – it's unfashionable. In the late 1980s and early 90s sampled loops were all the rave, and excerpts from James Brown's "Funky Drummer" (featuring Clyde Stubblefield) were all over the place. But since then, more subtle and sophisticated ways of using drum loops have evolved and if you were to ask anyone in the know how they approach drum loops they would probably answer something along the lines of "as icing on the cake". The common approach is to program the drums on a drum machine or play them on a keyboard, and then use various sampled loops to spice up the main drums. Thanks to Dr.REX, this method is a no-brainer in Reason. Try this:

1. Create the "main meat" of the drums using single samples in ReDrum or any of the samplers (NN-XT, NN19).
2. Create a Dr. REX and find a drum loop that complements your programmed drums in terms of rhythm, feel etc, particularly in the higher frequency range.
3. Use the filter on the Dr.REX - set it to HP mode (high pass) and use the FREQ slider to find a good spot for driving a wedge between the wanted and the unwanted frequencies.
4. Insert a compressor with fairly aggressive settings (i.e. Ratio 4-8:1, Threshold 0, Release 0) to bring the subtle parts of the loop to the surface.

We have prepared a couple of examples of the above Dr.REX application (use track or mixer muting for quick comparison): hipass_sizzle1.rns | hipass_sizzle2.rns

Dr.REX - the Swing Supplier

Even if you don't want to use a loop for its sound, you can still find the rhythmic feel of the loop appealing. Let's say you find a certain conga loop that – even though you happen to loathe congas – has a nice groove to it that makes you wanna get up and do a funky little dance number. After all, it has this human feel to it that no stiffly programmed drum pattern can mimic. Well, go for it – just steal the groove and throw the congas on the fire! Here's the procedure:

1. Create a Dr.REX and open that groovy loop
2. Click "To Track" on the Dr.REX
3. In the sequencer window, right-click on the group you just created
4. On the context-menu, select "Get User Groove"

Now the groove is all yours and you can find it on the Quantize grid drop-down menu as the last item: "User". It remains there until the next time you do a "Get User Groove" and is saved with the song, ready to be applied to any riff or pattern you record.

Dr.REX - the Animation Animal

Another cool thing you can do, is to employ the Dr.REX as a rhythmic modulation source. Using CV/Gate, the groove of a REX loop can be applied to the timbre of a synth or sampler sound – adding life and pulse to even to the most static of textures. This doesn't necessarily involve the Dr.REX as an actual sound source, it can just as well

be kept silent – you only need it there to supply the groove, for "animation" purposes.

Here's an example of a rather uninteresting sequence: Now, by using a Dr.REX drum loop to control the filter of the synthesizer, the exact same sequence sounds like this: . (For setup reference, see animation1.rns or animation2.rns).

Dr.REX - the Shape Shifter

"Drums? What drums?" The fact that a loop features drums doesn't dictate the way to use it. With the unique tweaking features of Dr.REX together with external modulation through CV/Gate, you can shape a drum loop – or any loop, for that matter – into something barely recognizable. Allow us to demonstrate: Here is a basic drum loop from the Factory Sound Bank:

Not much going on there, right? But with some elaborate use of filters, FX and Matrix sequencers we can take this loop and turn it into this:
Here's a Reason Song version in case you want to see how it was done: abstracted.rns)

Dr.REX - the Melody Maker

At first glance it might seem a daunting task to shape a music loop into something useful: Wrong key, wrong chord progression, silly melody – how do I fit this into my song? So now we're going to teach you how to gain ultimate musical control of the Dr.REX. First off, there are no less than six different ways to control the pitch of a REX loop:

- Global Transpose (automation enabled)
- Pitch of individual REX slices
- Oscillator Pitch modulation by external source (CV)
- Pitch Bend
- LFO Pitch modulation
- Envelope Pitch modulation

Using a combination of two or more of these will allow you to tame a music loop and have it follow you anywhere you want to go. Let's begin shall we:

First, we open a simple bass loop called "Take Your Pick 01" from the Reason Essentials ReFill package. In its original state, the loop sounds like this:

Then we "flatten" the loop by adjusting the pitch of some individual slices, to make the loop more manageable. With this particular bass sound that also means altering the timbre on a couple of notes since the entire fabric of tones and overtones is changed, but that only makes it more interesting. Here's the result:

Dirty trick 1: And now for something completely different. We're going to create an entirely new bass line by using a Matrix to control the pitch. This is a much more flexible method than tweaking the pitch of individual slices because it allows the pitch to be changed right in the middle of a single slice, and it also lets you play variations on the same Dr.REX without having to create multiple instances with different sequences. Users who like this approach would probably utilize the Curve grid on the Matrix to program the pitch sequence. But we're not going to do that, we'll reserve the Curve part for other things and instead use the Keys grid! But how? Note CV doesn't correspond to pitch does it? Too right, it doesn't. While it comes out as correct notes when connected to the Note CV input of any of Reason's synths and samplers, the "voltage scale" doesn't map onto a chromatic scale. The tiny difference in "virtual voltage" between a C and a C# doesn't constitute a semitone increment if you connect Note CV out to Osc Pitch CV in. But through this little unorthodox trick, we'll make it happen:

First, flip to the back side of the Dr.REX and connect Note CV on the Matrix to Modulation Input: Osc Pitch on the Dr.REX. Then, turn the knob next to the Osc Pitch input all the way to the right (value 127). This will "boost" the incoming CV so that it translates with 100% accuracy to semi-note values. Unfortunately, it will also thrust the pitch to near-supersonic range, so you will have to flip back to the front panel of the Dr.REX and turn the Osc Pitch: Octave knob all the way to the right (Value 0). Now you're free to step program a new bass line in the Keys grid on the Matrix. Here's an example of what that might sound like: Neat! Now, just because a REX file contains a bass loop you don't have to use it as a bass. Combining the tweaking possibilities of the Dr.REX you can turn it into a guitar, a staccato chord, whatever you want. Here's an example of multiple Dr.REX, all playing the same old bass loop but in completely different ways:

Now let's try that with drums on top:

Dirty trick 2: Right, so now that we've exhausted the Matrix-to-Pitch option and the pitch-per-slice option, is that it? Nope, there's yet another layer in store. This trick is ac-

tually not 'dirty' at all, it's documented in the Dr.REX specification, but far from all users are aware of it: You can transpose the Dr.REX from your MIDI keyboard using standard Note-on messages. This is by far the easiest and most "musician friendly" method of controlling of the pitch. All you need to do is to adjust the Octave range of your MIDI keyboard so that you can play the very bottom notes C-2 to C0.

Now let's use that in combination with the earlier trick (Matrix controlled pitch) to first create a custom sequence and then trigger the overall pitch from the keyboard.

We start with a loop from the Synthotica refill called "80 manip". This is a straightforward sequence playing the same note, and sounds like this:

Using "dirty trick #1", we program a sequence on the Matrix that controls the Osc Pitch of the Dr.REX. We also add effects, have some with filters, and push the tempo. Result:

Last but not least, we record the "macrolevel transpose": a simple 4-note (whole notes) sequence on the Dr.REX sequencer track. Voilá: Instant techno! The only thing missing is some vulgar techno drums, so let's get that over with: There.

Bottom line

As you can see, there's more to Dr.REX and REX files than rigid loop playback. Dr.REX is arguably the most unique instrument in the Reason environment, and the real fun begins once you go beyond the drum loops and start exploring other possibilities - try slicing up a vocal, guitar, bass or hardware synth recording in ReCycle and see how far you can push it in Dr.REX with filters, modulation, quantizing, transposition, randomization - the Doctor is always on duty, just ask!

Text & Music by Fredrik Hägglund

Dial R for ReDrum

ReDrum, she wrote.

Drum machines and hardware sequencers met a swift and gruesome death in the late 1990s when hardware samplers, sample players, workstation synths and computer based software sequencers put their predecessors out of their misery.

The new generation introduced a multitude of "human feel" elements, in stark contrast to the robot-like stiffness of the obsoleted thingamajigs. The goal - or so it would seem at the time - was to create the ultimate emulation of real instruments and real musicians. Hardware manufacturers spared no effort to ensure that their latest gadgetry offered superior realism, and software sequencer manufacturers were busy conjuring up extra relaxed quantization matrices, 'natural groove' algorithms, increasing the PPQN resolution - all to make sure that no two notes would sound exactly the same even if you tried.

Ironically, it didn't take long before a retro wave swept the music world, and now all of a sudden people were toppling over each other and trading in their tooth fillings to get their hands on an old battered TR-909. The 'human feel' efforts of late now became an obstacle on the way to creating the perfect 'robot feel': fixed velocity, synthetic sounds and the joy of repetition. People were making dance music like crazy and it took several years for hardware synth manufacturers to shake off the idea that all musicians are dying to play fusion jazz.

It's almost like poetic justice that it took cutting edge computer software to bring it back to the roots. Drawing inspiration from the Roland TR-series of the early eighties, Reason's ReDrum Drum Computer pays tribute to the old school drum machines that you could always trust to offer a quirky programming method, rigid timing and harsh limitations - and in this part of Discovering Reason we will hand out a few tips and tricks that will help you play the ReDrum to your advantage!

Note: The mp3 audio examples in this article are loops, and we therefore recommend

that you configure your default mp3 player to loop/repeat playback mode.

ReDrum by Numbers

Many favor the NN-XT sampler over ReDrum as the main drum device. While the NN-XT certainly has an impressive feature list, astounding flexibility and 16 separate outputs, ReDrum still offers a number of advantages:

1. A very direct, down-to earth design, providing instant access and excellent overview of all parameters.
2. Can be played via MIDI keyboard or the onboard pattern computer, or a combination of the two.
3. All parameters are automation enabled (the NN-XT is limited in this respect).
4. Dual FX sends on each of the 10 channels.
5. Gate In on all 10 channels (all other devices have a single Gate input for note control).
6. Gate Out on all 10 channels (apart from Dr.REX, ReDrum is the only instrument with this feature).
7. Notes can be randomized separately for each channel (ReDrum is the only polyphonic device in Reason that does this)
8. Easy "live browsing" - let the ReDrum play on while you use the channel specific Next/Previous sample buttons to quickly find the best kick or snare for the track.

Serial ReDrum

The presence of Gate in- and outputs on the ReDrum opens up a number of control routing possibilities. Let's try a couple of scenarios:

1. Sound layering (internal).

By connecting the Gate Out on one ReDrum channel to the Gate In on another channel on the same device, you can play two channels from one key. This of course allows you to link two or more sounds, but with the help of the Velocity controls you can simulate a velocity switch function. Just set the Velocity-to-Level controls to "opposite" values so that one sound decreases and one increases in volume the harder you hit the key. Here's an example file featuring a ReDrum with linked channels and velocity

switching: gate_velo_sw.rns |

In this setup, channel 1 trigs channel 2 (two different bass drum samples), channel 3 trigs channels 4+5 (three different snare drum components) and channel 6 trigs channels 7+8+9 (four different hihat samples, ranging from soft/closed to hard/wide open). You can of course link as many channels as you like - there's no shortage of cables in Reason - and if you run out of channels, just add another ReDrum unit to the chain.

2. Sound layering (external)

You can also connect other devices than ReDrum to the Gate outputs. If you like the pattern programming approach and/or synth drums, try this: Create a number of synth devices (Malström/Subtractor) and load the drum patches you want. Then flip over to the back side of the rack and connect Gate Out from each channel to Sequencer Control Gate on the back of each synth. Of course, you can still load samples into the 'puppetmaster' ReDrum, which allows you to layer sampled sounds with the analog style sounds. Example: gate_ext.rns |

(The track features some Mixer automation to showcase the two layers of sounds: Samples only / Synths only / Both / etc.)

Silence of the Mutes

(Author's note: Yes, "Silence of the Flams" is a better title, but doesn't make any sense.)

Each ReDrum channel has Mute and Solo buttons. And what's even better, Mute and Solo on/off can be controlled from a MIDI keyboard through plain note messages. This allows for a fun way to program drums; You can stuff the pattern full of notes on all channels, and then "play" the Mutes from your MIDI keyboard so that only the notes you "let through" are heard. This is cool because it gives the drum track a random and chaotic feel reminiscent of the likes of Aphex Twin, Autechre and Squarepusher. Observe: crazy_mutes.rns |

In the example track, there are three patterns playing *all* notes on *all* channels (1/4, 1/8T and 1/16 respectively), and flams on quite a few notes. It's the Mutes that shape it into music...! This is a very bang-for-buck method of quickly creating something that sounds like it's taken ages to program, when it's in fact done on the fly in a couple of minutes.

The result may also pleasantly surprise you, something which carefully calculated programming usually doesn't.

(The channel Mutes are controlled by the C2 through E3 keys on your MIDI keyboard.)

Bottom line

ReDrum embodies the Reason approach - keeping it simple, encouraging experimentation and incorporating the best of both the hardware and the software domain. The tricks showcased here are just a starting point for your tweaking adventures - we supply the ammo, you make the kill!

Text & Music by Fredrik Hägglund

Mastering Mastering

Version alert! This article was written before the release of Reason 3.0 with it's MClass Mastering Suite. Though the Reason mastering tricks have changed with that addition, the general mastering techniques are still valid.

Masterclass!

Careful with those subsonics, son.

This month we will be taking a break from Reason device dissection, and instead focus on a hot topic - mastering. Traditionally, mastering has been an isolated domain outside the music production perimeter, but today, more and more aspects of production and distribution are brought closer to home - mastering included. Artists are exploring ways of bypassing the traditional music distribution channels altogether, opting instead for mp3 files or home-made CDs - and if you're planning on going all the way with the do-it-yourself working model, you also need to master this final step of the process (as if being a composer, musician, producer and mixing engineer wasn't enough...!) Needless to say, there's a reason why people can make a career and a living on audio refinement - and if you're dead serious about your material you should consider taking it to a professional, as mastering would be considered by some as a "don't try this at home" thing. But if you're one of those adventurous spirits, here's a basic primer in the art and science of audio mastering - have your MasterCard ready and step up to the counter.

First, let's get this out of the way: If your burning question is "Why doesn't my music sound as loud as my commercial CDs? The peak meter tells me both sources are equally loud!", there are two things you should know: 1) This article will answer your question; in fact, it was written for you. 2) While we will focus on the subject of volume (real and imagined), there's so much more to mastering than just loudness. In fact, many mastering engineers resent the "loudness race" and favor a more conservative approach - but what's a poor home studio owner to do when every other CD you play is so loud it jumps straight out of the speakers and lumbers around the room like the Incredible Hulk? Let's crank it up!

**Perceptual loudness - blessing or curse?**

Have you ever found yourself jumping out of the sofa to hit the volume button on the remote whenever they cut to commercials? Audio tracks for commercials are usually "macho mastered", heavily treated with compression and limiting enough to suppress a nuclear blast. This is done to get the message across despite your vain attempts to seek refuge in the kitchen during the break - there's no escape! But, assuming the regular programming is played at maximum audio volume, how can commercials appear at least twice as loud? The long and short of it is: The human ear judges loudness not by peaks, but by average. Meet the concept of "perceptual loudness". One of the ear's imperfections is that it isn't fast enough to pick up extremely transient (=short) sounds in the 1-10 millisecond range and make an accurate interpretation of the volume. Modern audio science has taught engineers to exploit this shortcoming by developing techniques that ensure delivery of maximum sonic impact. "Normalization", however, is not one of them.

Normalization doesn't make it "normal"

You may have been offered the advice to "normalize" your tracks. All wave editors offer a normalization function. But what does normalization actually do? It searches for the highest peak in the audio file and adjusts the overall volume accordingly. If you've made a Reason track that stays barely at the safe side of the clipping limit, the highest peak is probably around 0 dB already, which means that normalization will accomplish absolutely nothing.

Let's cut right to the chase and look at a simple but effective demonstration before we get down to the nitty-gritty of it all (throughout the article we will be using snippets of well-known Reason demo songs for "target practice").

Note: The mp3 audio examples in this article are loops, and we therefore recommend that you configure your default mp3 player to loop/repeat playback mode.
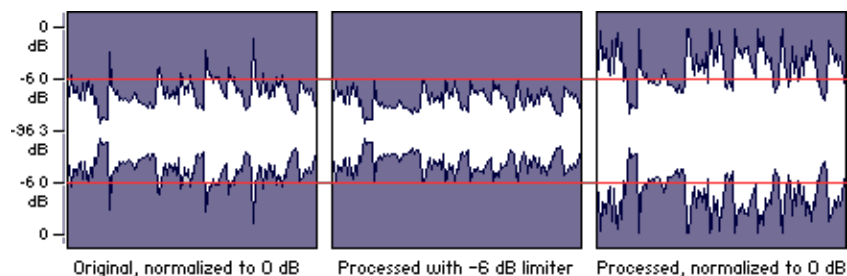
The left picture shows post-normalization audio, but as stated above, normalization is pointless if the level is already close to 0 dB (in this case, -0.21 dB, a negligible difference). If you look at the original (left) you can identify three peaks (those spikey, needle-like things sticking out over the red -6dB lines). In this case, they are caused by the bass drum. They're virtually irrelevant in terms of musical information, but they pose a problem in that they prevent you from increasing the average level. In the middle picture we have used a limiter to chop off everything above -6 dB. This may or may not be brutal, all depending on the material you're working on, but it serves the purpose here and now. Try listening to   the original and compare it to   the processed version - can you tell the difference? If not, you've made a bargain here, since a whopping 6 dB previously held hostage by the peaks has now been released. This leads us to the third picture (right), illustrating the processed sound subsequently normalized to 0 dB   . How's that for loudness?

This was not a complete mastering procedure by any stretch of the imagination, but it illustrates the fact that loudness is a very relative thing. Normalization is useful, but only after the appropriate processing has been done.

Clipping and meters

When analog was king, the most feared 'level enemy' was at the bottom of the scale - noise. Analog tape recorders could take moderate abuse at the top of the level ladder, but as soon as levels dropped, the noise was laid bare. Overloading an analog tape recorder didn't produce the nasty clipping artifacts you get in digital audio - in fact, a slight overload would often produce a pleasant sound. In the digital domain, a low audio signal isn't exactly a blessing either, but the side effects are nowhere near as destructive as digital overload is. Once the signal clips, the damage is irreversible - it's like an over-exposed photo, you can't bring back the parts of the picture that have already dissolved into white. So whatever you do, make sure that the raw, unprocessed audio doesn't clip. Keep an eye on the meter, but let the ear be the final judge - sometimes you can get away with clipping. However, if you don't have complete confidence in your ears, stay on the safe side and trust the red light.

The old school analog VU meters found on analog equipment were actually closer to the human ear's perception of sound level, because the response time was intentionally slow - around 0.3 seconds. A digital peak meter is something completely different from a VU meter. A digital peak meter is generally lightning fast - sample accurate - thus the

tiniest, most transient level spike will make it shoot straight to that dreaded red light, even though you could swear you didn't hear a peak - and in all likelihood, you didn't. A digital peak meter serves the interests of the digital audio device it speaks for, so perhaps "peak alarm" would be a more appropriate name. In other words, take it with a grain of salt.

Monitoring

We love it loud, don't we? During the long hours of a studio session you turn it up a notch every time the ears have gone numb. It makes the music sound better, more powerful, and brings subtle details out in the open. A word of caution: Don't. First of all, the human ear has a built-in compressor/limiter that works in mysterious ways (part self-preservation mechanism, part imperfection); at near ear-splitting levels your ears will smooth out the roughness and give you the impression that the mix is reasonably balanced when it's not. The best way to discover if anything in the mix shoots straight off the charts is in fact to listen at very low levels. Only then will you discover that, for example, the bass drum is twice as loud as everything else. Another trick is to listen from a nearby room rather than being right in front of the speakers. Second, the louder the sound, the more bass you will hear - this is because the ear's response to bass energy is non-linear. Consequently, monitoring too loud will prompt you to cut away some bass when in fact you should leave it as it is, or even boost it.

Before mastering

Are you happy with the sound of your song, or do you expect the mastering process to solve all problems? Even the most deft mastering guru cannot save a sonic disaster from an eternity in hell. There are many things to be mindful of during the actual music production and mixing; this is where you lay the foundation for a professional sound - subsequent mastering is only the icing on the cake. Here are but a few issues worth considering long before you reach the mastering stage:

- **Distribute evenly.**

  There's a long way to go between 20 and 20.000 Hz, but the frequency spectrum can only take so much abuse in one place before the mix becomes muddy. Keep an eye on the low midrange, it's usually the first to become crowded. Don't forget the equalizer's ability to cut rather than just using it to boost. Take your time to analyze

each sound and examine its characteristics - what does it add to the mix? Does it bring something undesirable with the desirable? If so, can the undesirable aspects of it be eliminated?

- **Hands off those low octaves.**

 What? You mean... no bass? Of course not. But when keyboardists play piano sounds or pad/string sounds, they often play the chords with the right hand and 'show the bass' with the left. This can become a bad habit and is a classic source of muddy bass, simply because that pad/string/piano sound (or whatever it is you're playing the chords with) will compete with the bass line for the lower tonal range. That left hand is best left in your pocket! Generally speaking, it's good arrangement practice not to have too many instruments fooling around in the same tonal range - as with frequencies, try to distribute evenly.

- **Less is more**

Hate it or love it, this old cliché always applies. If you feel that a song (or a certain part of a song) lacks energy, the best solution might be to take away rather than add. Every addition to an arrangement will suck energy out of the existing arrangement - occasionally for the better, but often for the worse. Granted, it is possible to pull off a wall-of-sound type production, but it's a delicate balancing act - it takes a great producer, a masterful mixing engineer and an elite mastering engineer to get it right.

- **Stack with care.**

With todays unlimited polyphony and neverending supply of instruments, stacking sounds is a luxury that anyone can afford. Why choose between two snares when you can have three, four, eight at once? Be careful though, because the bitter equation still remains that you can't add something without taking something else away. If you stack two sounds, make sure that they complement eachother, not collide with eachother. If you play a sampled drum loop over programmed drums, maybe you can cut some frequencies on the loop to make room for the programmed drums?

- **Mix with focus.**

  All sounds cannot be upfront. An inherent problem in arranging and mixing is that you often concentrate on one sound at a time. When you turn your attention to one sound you will be tempted to bring it out in the open, enhance it, nurture it, make it stand out from the crowd. Soon enough you will have given all sounds this special treatment, and as a result, no sound stands out, instead you find yourself knee-deep in mud. Treat the music as a painting - you want to turn the viewer's focus to one particular spot. All else is secondary and must be treated accordingly - don't be afraid to sacrifice a sound by abstracting it or even removing it altogether, it will always be to the benefit of the sound you want to turn the spotlight on.

- **Careful with those subsonics.**

  Frequencies that you can't really hear on most systems are generally a waste of bandwidth. They will force the average level of your music down, and all you gain is, well, no gain. What's worse, a booming subsonic bass that happens to sound good over your speakers because they can handle it, may turn the speaker cones inside out on lesser systems, particularly those that boast "MegaBass" or some other voodooish pseudo-bass tomfoolery - cheap lightweight headphones and ghetto-blasters that by all physical accounts should be unfit to deliver any bass whatsoever. You can't build dance floor rattling earthquake bass into the actual mix, that task will be taken care of by a dance floor rattling earthquake P.A. system when the time comes. Conversely, don't try too hard to emulate a hi-fi sound by boosting the high frequencies - keep it neutral.

Mastering software tools

There is an abundance of software products capable of performing mastering tasks, whether they were tailor made for mastering or not. The first thing you need is a good wave editor. For Mac, there is Peak and Spark; for Windows there is WaveLab, Sound-Forge, CoolEdit Pro and others. In addition to this there are many VST and DirectX plugins, including...
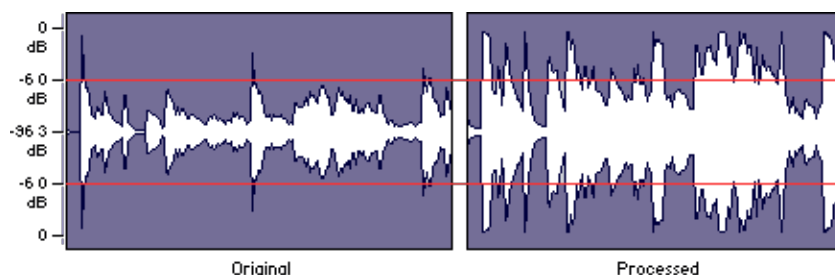
- **BBE Sonic Maximizer**
- **Steinberg Mastering Edition** - featuring Compressor (a multiband compressor), Loudness Maximizer, Spectralizer, PhaseScope, SpectroGraph and FreeFilter.

- **Waves Native Gold Bundl**e - featuring C4 Multiband Parametric Processor, Renaissance Reverberator, Renaissance Compressor, Renaissance Equalizer, L1 Ultra-maximizer, MaxxBass, Q10 Paragraphic, S1 Stereo Imager, C1 Parametric Compander, DeEsser, AudioTrack, PAZ Psychoacoustic Analyzer and much more.
- **db-audioware Mastering bundl**e - featuring dB-M Multiband Limiter, dB-L Mastering Limiter, dB-D Dynamics Processor, dB-S De-Esser.

There is also a T-Racks, a stand-alone mastering kit available for both Mac and Windows.

Of course, a plugin doesn't need to have "mastering" written all over it to be a worthy mastering tool - compressors, de-essers and dynamic processors are commonplace and can be used for mastering as well as other chores.

As an example of what kind of results you can expect from plugins like these, let's do an experiment.

First, we exported a snippet of the Reason demo track "Why Red" from Reason. Then we brought it into WaveLab. You can listen to the original, unprocessed sound here: We then used BBE Sonic Maximizer to bring more clarity and brilliance to the sound, and Loudness Maximizer to increase the perceived loudness. The result is here: You might also want to listen to this   A>B comparison which alters between the original and processed waveform every two bars.

This was a typical example of "macho mastering" to illustrate the huge difference you can make by processing a sound for maximum perceived loudness. If this is what you're after, look no further than the smart 'maximizer' plugins, but be careful not to over-use them - even those that are controlled by one single parameter are not foolproof. Unfortunately there's no "blanket procedure" you can apply to any and all tracks. You must listen to each track and identify its strengths and weaknesses. Play a commercial CD (preferably one you think sounds great...!) over the same system you're using for mas-

tering - once that sound has been imprinted in your mind, it's easier to go back to your own track(s) and spot the problems. A good eye-opener tool is Steinberg's FreeFilter (part of the Mastering Edition bundle). It's an equalizer that features a Learning function. You can play back a reference track ("source") and FreeFilter will analyze the sound characteristics. You then repeat the same procedure for your own track ("destination"). You will then be able to look at the difference between the frequency curves of the source and the destination, which gives you an opportunity to spot problems in the way you apply EQ in your tracks.

The quickfix FAQ

So, now we'e established the fact that there is no generic, failsafe, magic method for mastering. Where do we go from here? Perhaps the best approach is to turn the tables and provide some 'quickfixes' in the form of an FAQ which addresses some classic issues that can be fixed in mastering:

Q: *My tracks seem dull and quiet, they lack punch. What do I do?*

A: This is likely an issue with volume. Raw, unprocessed tracks produced solely in the digital domain can have a dry, lacklustre, almost 'papery' quality to them. Use a multi-band compressor, or - if you're not sure what all those parameters do - an advanced loudness plugin such as the Loudness Maximizer, L1 Ultramaximizer or BBE Sonic Maximizer. These generally produce better results than compressors because they don't add those unmistakable compression artifacts like 'pumping' etc. Keep in mind that the harder you push a Maximizer or Compressor, the more you squash the dynamic range. That carefully programmed hihat velocity might end up totally flattened, to the point where you might as well have used fixed velocity.

Q: *My songs seem to lack high end. I tried boosting the high frequencies but it just didn't sound good. Help...?*

A: Boosting the high frequencies will often make matters worse, as it can add unpleasant hiss. Experiment with an exciter-type plugin, these are designed to add pleasant sounding brilliance. Try BBE Sonic Maximizer, or the High Frequency Stimulator by RGC Audio (free).

Q; *My track sounds harsh. My ears hurt. Can this be fixed?*

A: Possibly. The nasty frequencies you're looking for are usually between 1 and 3 KHz. Use an equalizer to locate and cut.

Q: *I'm happy with the loudness, but the sound still lacks 'presence'. How...?*

A: The magic frequencies you're looking for are between 6 and 12 kHz. You can try a moderate boost in this range. You can also try Steinberg's Spectralizer, another 'magic box' that adds transparency and clarity using harmonic generators to produce synthesized overtones.

Q: *Not enough bass. Give me bass!*

A: As usual, go the equalizer way or the magic plugin way. Waves MaxxBass and BBE Sonic Maximizer are both capable of generating booming bass. If you prefer using a regular equalizer, there are multiple approaches and you might have to try them all to find the appropriate one. The problem could be that there is too much going on in the lower midrange, which gives a certain 'boxiness' to the sound. The culprit is in the 100-400 Hz range - try cutting. If the bass seems OK in terms of loudness but you'd like it to be deeper, try boosting gently in the 30-40 Hz range and cutting gently around 100-120 Hz.

Can Reason do it?

Reason is by no means a mastering tool, it should be used for composition and mixing, after which you should render the audio files at the highest possible bit depth and sampling frequency your software and hardware can handle, for further processing outside Reason. Having said that, Reason does feature the tools required for advanced EQ:ing and compression - this can be useful if you're a fan of the RPS publishing format and want to add that 'finalized' quality to songs played straight out of Reason. You will find that many songs in the song archive utilize a single COMP-01 as master compressor and a PEQ-2 as master EQ. However, a single compressor might not produce satisfactory results, especially not on extremely intense material. Since it works over the entire frequency spectrum you will often get a "ducking" effect - i.e. one sound pushes other sounds out of the way. For example, a dominant bass drum that looms high above the rest of the soundscape will prompt the compressor to inflict heavy damage on every beat, to the effect that all other sounds disappear out of 'hearsight' everytime the bass drum is triggered. To overcome this problem you need a multiband compressor - it works like a battery of compressors, each handling its own slice of the frequency spec-

trum. Three bands (low, mid and high) is usually more than enough. Thanks to a couple of new devices in Reason 2.5, you can now build your own multiband compressor in the Reason rack. The procedure is as follows:

1. Create a 14:2 Mixer.
2. Create a Spider Audio Merger & Splitter. This will be used both to split the stereo signal from the master mixer output and to merge the signals
3. Create three BV512 Vocoders. Hold down Shift to avoid auto-routing (Reason will not guess right here).
4. Set all three Vocoders to Equalizer mode, 512 bands (FFT).
5. To divide up the three frequency ranges, use the sliders on each BV512's display. You need to cut all the bands except the ones that the Vocoder/EQ will be handling, so for the low range unit, leave the leftmost third (e.g. bands 1-10) as-is and pull the remaining sliders down to zero.
6. Repeat step 4 for range assignment of the mid- and high range BV512 units - assign the bands on the middle of the display (e.g. bands 11-22) to one and the remaining bands on the right side of the display (e.g. bands 23-32) to the other. (see illustration and .rns example below)
7. Create three COMP-01 compressors.
8. Create a Spider Audio Merger & Splitter.
9. Routing time: Mixer out to Spider Audio split input / Spider Audio split output 1-3 to BV512 #1-#3 carrier inputs / BV512 #1-#3 outputs to compressors #1-#3 inputs / compressors #1-#3 outputs to Spider Audio merge inputs 1-3 / Spider Audio merge output to Hardware interface.

For variation on this theme, try adding another pair of COMP-01 and BV512 and you get a multiband compressor with four bands (for example, low + low midrange + high

midrange + high).

Or, you can try replacing each COMP-01 unit with a Scream 4 set to the Tape preset...

Here is a template Reason song with the multiband master compressor setup.

As a bonus, you can of course adjust the EQ bands on the Vocoders - together they serve as a master equalizer.

Online resources

- Digital Domain
- An Introduction to Mastering by Stephen J. Baldassarre
- 20 Tips on Home Mastering by Paul White
- What to Expect from Mastering by John Vestman

Text & Music by Fredrik Hägglund

What is the Matrix?

You have to see it for yourself. You either love it or ignore it; there is a lot of ambiguity surrounding this particular Reason device. Without generalizing too much, it can perhaps be said that traditional keyboard players are the first to ask "What's that thing for anyway?"

The fact is that the Matrix can be most anything you want it to be, whether you're a keyboard virtuoso or a 'paint-by-numbers' kind of composer. Think of it as your trusty house elf, the one that performs tedious automation chores while you concentrate on more important things. Or, think of it as the bug in the electrical system - plug it into any of those mysterious CV/Gate connectors on the back panel and interesting and unpredictable things may happen.

The Matrix Reloaded

The reason we're kicking off the new Reason era - 2.5 - with an in-depth look at the Matrix, is that it just got a whole lot better. Thanks to the new Spider CV Splitter & Merger unit introduced in 2.5, the signals from a single Matrix unit can now multiply and spread like wildfire over the entire rack. How about two Subtractors and two Malströms playing the same bassline? Or volume automation of 16 synths at once? The sky is the limit. But before we board the rocket and cram the rack chock full of Matrix units, let's go over what the Matrix can do, device by device:

In Reason 1.0:

- **Mixer 14:2:** Automation of Master Volume, Channel Level and Pan, per-channel *(curve)*.
- **RV-7 Reverb:** Automation of Decay *(curve)*.
- **CF-101 Chorus/Flanger: :** Automation of Delay and Rate *(curve)*.
- **DDL-1 Delay:** Automation of Pan and Feedback *(curve)*.

- **PEQ-2 Parametric EQ:** Automation of Frequency 1, Frequency 2 *(curve)*.
- **PH-90 Phaser:** Automation of Frequency and Rate *(curve)*.
- **COMP-01 Compressor:** No CV/Gate inputs.
- **D-11 Distortion:** Automation of Amount *(curve)*.
- **ECF-42 Filter:** Automation of Frequency, Resonance *(curve)* and Envelope trig *(gate)*.
- **Subtractor:** Monophonic sequencing of notes *(note, gate)*, automation of OSC Pitch, OSC Phase, FM Amount, Filter 1 and 2 Frequency and Resonance, Amp level, Mod Wheel *(curve)*, Amp Envelope trig, Filter Envelope trig, Mod Envelope trig *(gate)*.
- **NN-19:** Monophonic sequencing of notes *(note, gate)*, automation of OSC Pitch, Filter Cutoff and Resonance, Level, Mod Wheel *(curve)*, Amp Envelope trig, Filter Envelope trig *(gate)*.
- **Dr. REX:** Automation of OSC Pitch, Filter Cutoff and Resonance, Level, Mod Wheel *(curve)*, Amp Envelope trig, Filter Envelope trig *(gate)*.
- **ReDrum:** Sequencing of notes, per-channel *(gate)*, automation of Pitch, per-channel *(curve)*.

+ Reason 2.0:

- **Malström:** Monophonic sequencing of notes *(note, gate)*, automation of Pitch, Filter, Index, Shift, Level, Mod Amount, Mod Wheel *(curve)*, Amp Envelope trig, Filter Envelope trig *(gate)*.
- **NN-XT:** Monophonic sequencing of notes *(note, gate)*, automation of OSC Pitch, Filter Cutoff and Resonance, LFO1 Rate, Master Volume, Pan, Mod Wheel *(curve)*, Amp Envelope trig, Mod Envelope trig *(gate)*.

+ Reason 2.5:

- **BV-512 Vocoder:** Automation of Hold *(gate)*, Shift and individual Band levels *(curve)*.
- **RV-7000 Advanced Reverb:** Automation of Decay, HF Damp *(curve)* and Gate trig *(gate)*.
- **Scream 4:** Automation of Damage Control, Parameter 1 and 2, Scale *(curve)*.
- **UN-16 Unison:** Automation of Detune *(curve)*.
- **Spider CV:** Merge, split.

That's a total of 149 inputs just waiting for a Matrix connection - so what are we waiting

for?

A few ma-tricks up the sleeve

Since creating Matrix patterns can be time consuming, it's a good idea to have a preset Matrix unit in your customized default song. Naturally, musical preset patterns might not be very useful but you can make lots of handy generic curve and gate patterns ready for instant access. Some basic ones, like this:
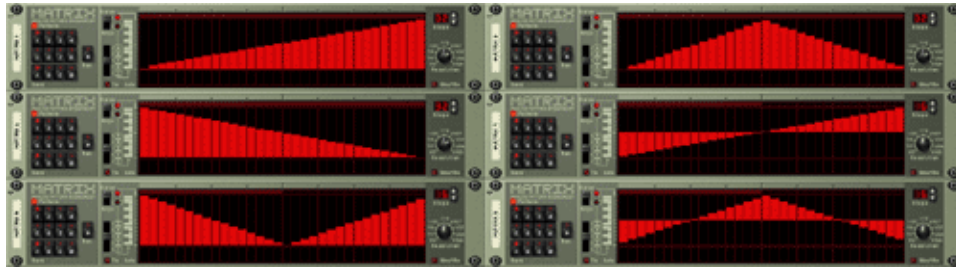
Tip: Hold down [Shift] while drawing, to draw continuous curves or note sequences

Curves like these are very handy for filter sweeps, pan automation, level automation and other things. If you keep these basic curves in mind and go back and read the above list of places to plug in the Matrix you're guaranteed to spot a dozen instant uses for them.

These curves only have to be drawn in one resolution, since you can easily change the speed by changing the time resolution of the pattern. In 1/32 mode, the above patterns play for one bar - change to 1/16 and the length will be two bars, 1/64 = half a bar, and so on.

Of course, you can also use Randomize Pattern and Alter Pattern. While randomized notes seldom produce a musically correct or appealing result, randomized curves and gates often turn out interesting and useful.

Using the new Spider CV module for merging two basic Matrix curves you can generate insane curve data by for example merging two identical curves played back at different speed!

Fuel Injection

Now for examples. First off, let's just let the Matrix flex its muscles in a simple yet effec-

tive "before and after" demonstration to show the profound effect that a little CV/gate injection can have on your sounds, even if you choose to play the actual notes from the keyboard and not the Matrix. History has shown that many Reason users overlook the power of the Matrix and of CV, and choose to limit their use of modulation sources to what each instrument offers.

In this example snippet we've taken a Matrix with a completely random pattern, split its signals through a Spider module and plugged it into virtually every available hole on one Malström and one Scream unit. Here's the little tune before it was fitted with a turbo: fuel.rns | And here's the after shot: fuel_injected.rns |

Which version is more spunky?

A multi-tasking ReDrum

One of the inherent annoyances with pattern-style programming of drum machines is that you need a new pattern for each tiny variation. But why not use Matrix units to create one big "multi-tasking" pattern sequencer with one independent layer for each ReDrum channel? There are many advantages to this method:

- Pattern variations only need to occur on the involved sounds, the rest can play on independently (example: the snare drum Matrix can temporarily change to a snare roll pattern but the bass drum and hi-hat patterns remain unchanged)
- Each drum channel can have its own individual time resolution (example: the hi-hat is programmed at 1/32 resolution in a 32-step pattern but the bass drum in 1/8 resolution in an 8-step pattern)
- Velocity resolution is 1-127 instead of 1-3
- Triggering drums only requires Gate signals from the Matrix so you can use the Curve part to control the pitch of the drum if you like
- Instant visual representation of each individual drum pattern right inside the rack, no need to switch to Edit view
- You can combine ReDrum pattern data with Matrix pattern data - you don't need ten Matrix units for each ReDrum instance, you only need them for ReDrum channels where lots of variations occur

In this example song we've used three ReDrum channels for bass drum, snare and hihat. The bass drum pattern is static while the snare and hihat patterns switch between

two patterns each, while also using a different time resolution and pitch automation.

multidrum.rns |

The Return of the Dirty Trick

Remember "dirty trick 1" from the "Ask Dr. REX!" article? We're going to use this again to create a fake arpeggiator from a Matrix and a couple of synths. The idea is to turn the "Osc Pitch" knob next to the Osc Pitch input on the back panel of the instrument all the way to the right (value 127) which will "boost" the incoming CV so that it translates with 100% accuracy to semi-note values, allowing you to change the pitch with Note CV instead of Curve CV. This is not only more intuitive but it frees up the Curve CV so you can use it for other things.

Using a single Matrix we will control one Subtractor and one Malström, combining the Matrix data with MIDI note input. To do this we need to split the Matrix signal using a Spider CV Merger & Splitter. Here is the example file: digibubbles.rns |

Now open the song and explore the signal paths. The Matrix is sending Note CV to the Spider module which splits the signal in two, sending one to each synth where it's boosted to map onto the chromatic scale. The Matrix is also sending Gate data, split by the Spider and sent to the Filter Env input of the Subtractor and the Filter (not Filter Env) CV input of the Malström, adding a rhythmic texture to both sounds. The Curve output from the Matrix, finally, is connected to the Filter 1 Freq CV input on the Subtractor. But wait, it's not making sounds yet(!). What you now need to do is feed each synth with MIDI notes. A single sustained note is all it takes, and the Matrix will play the melody based on that 'root note'. You can play any single note or chord you want and the Matrix will take care of the rest. The example file already has MIDI notes in place but you can mute the tracks and play your own notes to get an idea of what it's all about. If you need more variations on the pattern (such as major/minor scale alternatives) just copy the original Matrix pattern to another slot and change whatever you like.

Bottom line

The truth is out there, Neo.

Text & Music by Fredrik Hägglund

Reason Vocoding 101

I am your father.

Yeah, but who's your daddy?

Look who's talking now.

Lo and behold: Reason 2.5 is out. The bad news is, it still won't do your homework or your household chores for you; the good news is, at least it can talk. The BV-512 Digital Vocoder vocoder is a cutting-edge space-age rendition of a hardware Vocoder, with a whopping 512 bands and an equalizer mode.

A brief history of the Vo(co)der

The year was 1928 and Homer W. Dudley of Bell Telephone Laboratories, New Jersey, embarked on his own private Homer's Odyssey in search of a way to reduce the band-width required for telephony, in order to boost transmission capacity. Little did he know that he was way ahead of his time - nowadays, a similar process is used by mobile telephone operators to squeeze more calls into the systems, by way of digital technology like Dynamic Half-Rate Allocation and Adaptive Multi-Rate Codecs. Homer's idea was to analyze the voice signal, break it down and resynthesize it into a less bandwidth-hungry signal. He called this process "parallel bandpass speech analysis and resynthesis" and conceptualized it through a prototype named *The Vocoder* (short for "voice coder"). The Vocoder evolved into a more commercially viable design, got renamed to "The Voder" and was unveiled in front of a large audience at the 1939 World Fair. You can listen to a demo of the original 1939 Voder here:

I just called... to say...

Sadly, the Voder never made the big time, commercially - perhaps the idea of transforming people's telephone conversations into hollow robotic blabber didn't sit well with phone company executives - however, a digital version of the Voder had a cameo appearance during World War II, where it served as a part of SIGSALY, a secure voice communication system used for enciphered speech transmissions between Franklin D. Roosevelt and Winston Churchill. The machine remained a closely guarded military secret until the seventies...!

Yup, the digital vocoder existed in 1942 - and now, some sixty years later, it's finally your turn to play with one. But how does a vocoder work?

It Takes Two to Tango

The very first thing you need to know about the concept of a vocoder, is this: It needs two sound sources to function. One *carrier*, one *modulator*. This doesn't necessarily mean that it requires two separate Reason devices, however - but, more on that later.

A good (albeit flawed) analogy would be to think of the carrier as the raw material, and the modulator as the mold; the Vocoder makes a cast of the modulator and pours the molten carrier into it. Here's how it works:
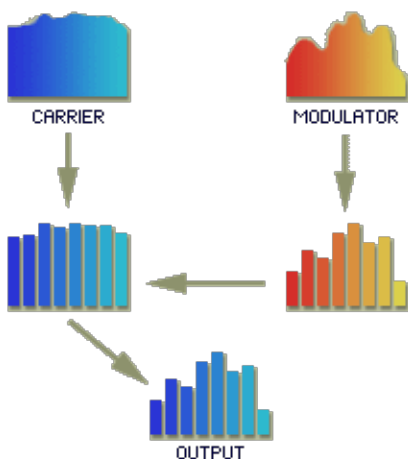
Much like a paper that goes through a document shredder, the *modulator* signal is sliced up in [X] number of bands, each representing one 'slot' of the time-varying frequency spectrum. With BV-512, the number [X] can be 4, 8, 16, 32 or 512. The analyzer then looks at the amplitude of each band. By this process, a spectral 'blueprint' is extracted. Simultaneously, the *carrier* signal is analyzed and divided in an equal number of slices. The frequency characteristics of the modulator can now be 'superimposed' onto the carrier.

It is therefore preferable that the carrier features a rich spectral content, so that the vocoder can always "find a match" for the modulator. For example, if the carrier sound is a heavily filtered pad sound with nothing going on in the high frequency range, and

the *modulator* is a recording of vocals, you will get a muffled and unintelligible result. This happens because there is nothing in the *carrier* material that corresponds to the high frequencies required to produce consonants such as "s" and "t".

Classic Vocoder

Now that we have an idea of how a vocoder works, let's try our luck with that good old trademark vocoder sound. For this we will need a vocal sample (the modulator) and an analog synth sound (the carrier). The Vocoder itself needs to be in 8- or 16-band mode in order to emulate an old school analog vocoder, since this is in the range of the number of bands these used to have. With the 32- or 512-band setting you tend to lose the 'robot' quality and the human quality takes over.

As for the carrier sound, instant results will be obtained by using a raw, unfiltered saw-tooth wave - with the emphasis on raw; the most effective carrier sounds for getting vocals across are usually sharp, loud and unrelenting, and sound pretty awful on their own.

A quick way to create such a carrier sound is this:

- Create a Subtractor.
- Using the "Init Patch" settings as a starting point, open up Filter 1 completely by moving the Freq slider to value 127.
- Make the patch more "sluggish" by adding some portamento, some attack time and some release time. This adds an organic touch to the carrier sound, since the human voice A) tends to glide slightly between notes, and B) doesn't open and close instantly like an envelope in gate mode.

From this basic setup you can go on to add variations: Maybe you want a monophonic carrier. Maybe you want to enable Oscillator 2 and detune it, or use it as a sub oscillator. Maybe you want to use the noise oscillator, which gives the voice a more raspy, whispering character and accentuates the consonants. Maybe you want to add a Scream 4, which really spices things up, either inserted after the carrier or after the vocoder. Experiment!

Here's a sample setup: classic_voco.rns

Almost Human

If you're looking for a less machine-like, more human vocoder flavor, you should move up to the 32- or 512-band setting where the details really shine through. The carrier sound needs two elements - a vocal timbre (preferably a sample) and noise. This is perhaps best done with an NN19 or NN-XT, but here we will take our chances with the Malström:

- Create a Malström.
- Use one oscillator for noise: Try the "Pink Noise" graintable. Route it through the filter. Set it to Bandpass mode and keep turning the frequency knob until you get something that sounds like a long "SSSSS".
- Use the second oscillator for the vocal timbre. Experiment with the various Voice graintables such as "MaleChoir".
- Like in the Subtractor example above, add a little portamento, attack and release time.

The Malström setup: humanoid_voco.rns

Keep toying with the different Malström graintables and you'll find many otherwordly carrier sounds. Or, if you're a vocalist yourself, sample your own voice producing a steady "aaaaaaaah", then "ssssssss", mix these two together and loop the sample. Use this as a carrier in 512-band mode and you can approximate the immensely popular (and by some, fiercely hated) 'auto-tune' effect known from Madonna's "Die Another Day", Cher's "Believe" and many others. Investigate!

Noise Pollution

Noise is fascinating carrier material thanks to its rich spectral content; White noise is all over the frequency spectrum, which means that there will always be something that corresponds with the modulator signal - it lets you "project a shadow" of the modulator. In theory, this means that extending the decay time on the vocoder will result in a reverb-like effect. Not that you're in dire need of another reverb now that you have RV-7000, but this is a reverb with a twist. Allow us to demonstrate:

- Download the example file reverb_voco.rns, open it in Reason 2.5 and hit Play.
- Examine the setup: The carrier is a Malström playing the "Pink Noise" graintable, the modulator is a Dr.REX playing a drum loop, and the Vocoder is in 512-band mode. Try turning the Attack time up for a bit of pre-delay on the Reverb effect.

- Turn the Decay knob on the BV-512 down to zero and the Dry/Wet knob to 127. Now you will hear the 'ghost' of the drum loop projected on the carrier sound (it resembles heavy mp3 or RealAudio compression).
- Now turn the Decay knob back to where it was, around 87, and do the same with the Dry/Wet knob, back to 50/50 (around 64) - or simply reload the song. Now for the fun:
- Experiment with the band setting on the vocoder. Notice how 4 or 8 bands produces a cool "old school Roland analog beatbox" effect.
- Fool around with the Shift knob on the vocoder. Consider that it can be automated and CV controlled, too.
- Enable Mod B on the Malström. It will control the Motion on the noise graintable.
- Browse through the graintables on the Malström and hear what "Thunder reverb" or "TibetanMonks reverb" sounds like. Explore!
- Bonus round: Switch MIDI input to the Vocoder track in the sequencer, and mute the Dr. REX track. Now you can play the Vocoder bands from your keyboard.

Same, Same But Different

Is there a way to use the same device as both carrier and modulator? Certainly. Is there a purpose? Definitely.

1) Using the same sound as carrier and modulator

Needless to say, if the carrier and the modulator signals are identical, not much will happen. A sound imposing its characteristics on itself will result in status quo. But once you throw effects into the melting pot, things get interesting. Par example:

- Create an NN19, a BV-512, a Spider Audio and a Scream 4. Load a vocal sample into the NN19.
- Using the Spider, split the NN19 output and route one signal to the modulator input on the BV-512, and the other through the Scream 4 and into the carrier input on the BV-512.
- Play the NN19 from your keyboard and mess around with the Shift knob on the BV-512 for interesting results; the Shift parameter now controls the formants.
- (example file: shift_voco.rns)

2) Using the same device as carrier and modulator

Since the NN-XT has multiple outputs, it can be used as both carrier and modulator at the same time. In other words, It will need only one MIDI input source to work. Par ejemplo:

- Create a BV-512. Create an NN-XT and load a strings patch.
- Add one sample Zone and span it across the entire keyboard. Make a separate Group of it and route it to Output pair 3+4. Load a rhytmic sample into this Zone.
- Set the Group polyphony parameter to 1 (monophonic).
- Change the Pitch kbd tracking to zero (fixed pitch) for this Zone.
- Connect the 1+2 outputs to the carrier input of a BV-512. Connect output 3 to the modulator input.
- Now you can play chords on the keyboard and hear the rhythm sample affect the timbre of the sound. (example file: lazy_voco.rns)

You can of course also use two polyphonic NN-XT sounds in the above setup to "morph" any two sounds together: Strings and piano, effect sound and pad... it's your call.

Bottom line

Hopefully, this exercise has begun to open your mind to the fact that a vocoder is much more than a talking synth thingamabob. A vocoder is a set of bandpass filters, and it imposes the characteristics of one audio signal onto another audio signal. Either audio source can be any sound, and vocals is just one source material out of billions. Now get to work!

Text & Music by Fredrik Hägglund

Scream and Scream Again

DISCOVERING REASON **Oh, the Horror...**

There's a new breed of Reason FX units in town, and they're not to be messed with. The most lethal weapon of them all is the Scream 4 Sound Destruction Unit. Sure, it can play nice, gently warming up your sounds with a smooth tape saturation algorithm - but underneath the innocent facade lurks an audio assassin, a mad mangler that will tear your sounds to shreds and leave no one alive to tell the tale. Dare you read on?

Distorted Reality

Scream 4 is arguably one of the best sounding and most versatile distortion units in the software domain. It's capable of producing very realistic, natural distortion. But distortion comes in many shapes and sizes; even though the term in a musical context has become somewhat synonymous with the rock'n'roll-type distortion produced by a guitar amplifier pushed over the edge, the term encompasses much more than that. As a dictionary would have it, distortion is "the act of twisting something out of natural or regular shape". Some of the ten Scream 4 algorithms can not be attributed to analog or "real world" distortion emulation, they're something different altogether: Modulate, Warp and Digital. Here's a quick recap of the different algorithms (described in full detail in the Reason 2.5 Operation Manual, page 226):

- **Overdrive** *produces an analog-type overdrive effect. Overdrive is quite responsive to varying dynamics. Use lower Damage Control settings for more subtle "crunch" effects.* Overdrive is the quintessential "pleasant" distortion, the kind that analog gear produces when pushed to the limit and beyond. It was discovered in the infancy of rock-'n'roll - in the 1950s, guitar amps were usually low powered (5 to 35 watts) and easily frightened. This was long before guitar amplifier manufacturers had any idea that this was a *desirable* way to use an amp - they would have referred to it as abuse.
- **Distortion** *is similar to Overdrive, but produces denser, thicker distortion. The distortion is also more "even" across the Damage Control range compared to Overdrive.* This algo-

rithm appears to emulate transistor distortion which is more of a 1980s heavy metal type deal.

- **Fuzz** *produces a bright and distorted sound even at low Damage Control settings.* This is the angry, wasp-like 1960's Jimi Hendrix effect. An obtrusive buzzing sound with little to no low end.
- **Tape** *emulates the soft clipping distortion produced by magnetic tape saturation and also adds compression which adds "punch" to the sound.* This is the algorithm for anything you need to "de-digitalize" - individual instruments or the entire mix.
- **Tube** *emulates tube distortion.* Tube distortion is warmer, thicker and more musically appealing than transistor distortion. Tube amplifiers (originally called "valve" amps) were the original guitar amplifiers, later to be replaced by transistor amps, but soon resurfaced and remains the 'gourmet' choice to this day.
- **Feedback** *combines distortion in a feedback loop which can produce many interesting and sometimes unpredictable results. Feedback is basically when a sound source is fed back to itself.* This algorithm is great fun, but you need something that controls the P1 and P2 parameters to fully appreciate it.
- **Modulate** *first multiplies the signal with a filtered and compressed version of itself, and then adds distortion. This can produce resonant, ringing distortion effects.* Also called Multiplicative Synthesis, this effect is quite common on synthesizers (the Subtractor, for instance) but gets especially interesting when used on a natural sound such as a vocal or acoustic instrument sample.
- **Warp** *distorts and multiplies the incoming signal with itself.* Watch Star Trek for more information.
- **Digital** *reduces the bit resolution and sample rate for raw and dirty sounds or for emulating vintage digital gear.* This effect has so many potential applications, from emulation of crusty old arcade game samples to Aphex Twin type "digital meltdowns" where the entire mix is sucked into a vortex of pixelated audio.
- **Scream** *is similar to Fuzz, but with a bandpass filter with high resonance and gain settings placed before the distortion stage.* The cool thing about this algorithm is the bandpass filter. You can control its frequency with the P2 knob (or connect the Auto CV out to the P2 CV in) to produce a heavy wah-wah effect.

Basically, you should try to forget all the conventions about distortion, since: A) You can distort *any* sound, not just guitar, and B) The Scream 4 is more versatile than most distortion effects. Drums, bass, vocals, pads - anything goes. Here's a quick example of what the Scream 4 can do for a Vocoder sound - an excerpt from the Reason 2.5 Flash

showreel.

Since the Scream 4 is pretty straightforward we will not go over different examples of effects in this article - a wide range of Scream 4 presets is featured in the In Full Effect Sound Bank that installs with Reason 2.5. Instead, we will look at one of the "hidden" features - the Auto envelope follower which is a new type of CV signal source, previously unavailable in Reason and very useful for all sorts of tricks.

Grand Theft Auto

The Auto section of the Scream 4 is so useful that one simply has to steal it, or at least borrow it. You can use this even if you're not employing the Scream 4 as an actual effect. If you start thinking about scenarios where you'd like the volume (or inverted volume) of one sound to control something, you'll soon come up with a lot of possibilities. How about dynamic control of the detune parameter on the UN-16 Unison - the louder the sound, the heavier the detuning? Or how about controlling the Shift parameter on a Malström? Building your own compressor? There are hundreds of possibilities. Here are a few examples:

Ducking. You can send the Auto CV through a Spider CV and invert the signal. This means that the higher the amplitude of the input sound, the lower the Auto CV value. This application can be used for a "ducking" effect, i.e. when the volume of one sound source increases, the volume of another source decreases. In the example file duckand-cover.rns, we're using two Dr. REX units playing different drum loops. Increasing the volume level on one Dr. REX will "drown out" the other - but in this case, not completely; whenever there is silence between the drum loop slices on the governing Dr. REX, the "slave" Dr. REX will slip through. Try increasing the reverb decay or the delay feedback in the example file and you'll get the picture.

External source Auto Wah. How about letting a second source control the wah effect? For this we need two sound sources and one Scream 4 unit. In the example file blabber-mouth.rns we use a ReDrum as a source signal for the Auto section of a Scream 4 unit which in turn controls the cutoff frequency on a Subtractor. Since in this case we don't want a distortion effect on the ReDrum, we can either turn off all the effect sections on the Scream 4 (the auto CV out will be operational anyway), or we can use a Spider Audio to split the signal from the ReDrum, send one stereo signal to the mixer, and the other to the Scream 4 where it's "terminated". As a bonus, you will find one more Scream 4

unit in the example file: This one is used as an effect on the Subtractor, and it's also using its Auto CV output to control its own P2 parameter. The signal is inverted through a Spider CV, and the P2 parameter on the "Scream" algorithm controls a resonant bandpass filter, meaning that we get an "inverse wah-wah". Check it out.

ECF-42 Wah. If you can't seem to get quite the wah effect you're after, using the Auto section on the Scream 4 alone, you might want to try heavier artillery. The ECF-42 Envelope Controlled Filter might be your cup of tea. Soopah_wah.rns is an example of such a setup, where the ECF-42 is in bandpass mode. It can be tricky to get the Auto section working right, but in 9 out of 10 cases the problem is that the signal is too high, causing the envelope follower to fly wide open - consequently, you don't get a "wah" but rather a "weeeeeeeh" effect. There are different ways to remedy this problem, but a good place to start is to lower the volume level on the input source, i.e. the instrument. The Scream 4 can compensate for low levels, because it has some gain headroom; to "calibrate" the Scream 4, switch off Damage, Cut and Auto. Then set the Master pot to 100 (which is not the maximum volume; 127 is). Switch between On and Bypass and you'll find that there is no difference. Now you can experiment with the Auto section and perhaps lower the volume of the source sound; use the Master gain to compensate. If you're using Auto CV and get the same problem, use the CV adjustment pot on the target device to dampen the signal.

Bottom line

It's a killer device, no doubt. It's also a chameleon and highly useful as an insert effect on just about any instrument in the rack. You don't always have to use it at full blast, it can do wonders even at near unnoticeable settings. Don't set it loose unsupervised.

Text & Music by Fredrik Hägglund

Space Madness!

Once Upon a Time In Sweden...
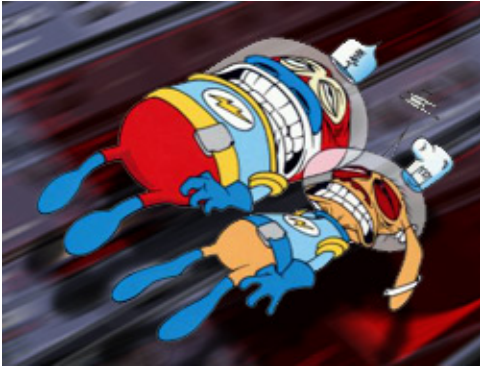
When the distinguished gentlemen of Propellerhead Software set out to create a new reverb unit for Reason, their goal was to be able to stand up and say: "This is one of the best software reverbs on this planet". They just wouldn't rest until RV7000 compared favorably to the crème-de-la-crème hardware units on the market. And to all intents and purposes, they succeeded.

Now, it is particularly crucial that a *reverb* unit is really, really good; unlike other more overtly synthetic, alien sounding devices in Reason, RV7000 needs to convincingly emulate something that we experience in natural environments every day. Consequently, RV7000 has a harder job than any other device in Reason. And what a bang-up job it does!

What is reverb, anyway?

Simply put, *reverberation is a natural acoustic environment's response to sound*. Yet, many people view reverb as just another effect, and often use it accordingly. But the term "effect" doesn't even begin to cover it: Literally *everything* we hear in real life is accompanied by acoustic reflections. Therefore, a dry, mute, dead sound with no reverberation whatsoever is in fact more of an effect than reverb is, simply because it's, well, unnatural.

Reverb units are at your service to (re-)establish an air of realism. Having said that, drenching sound completely in reverb isn't quite the way to go either - one has to take into account the different listening environments the music might face. The room where a pair of speakers reside may add some measure of reverberation all by itself, while as a pair of headphones need all the space simulation help they can get (unless of course the listener has very *VERY* large ears), so - as usual - a compromise is in order.

Now we will go over a few parameters of the RV7000, specifically those which - while having most appropriate names - aren't entirely self-explanatory.

- **Pre-delay**. Cleverly applied pre-delay can provide a sense of depth. For example, a very long pre-delay gives the impression that you are close to the sound source, let's say a lead vocalist. It creates a feeling of intimacy and places the lead vocals in focus. A very short pre-delay, on the other hand, sells the idea that the singer is positioned far away at the opposite end of a long tunnel, since the reverb and the voice reach your ears at the same time. This is suitable for ambient sounds, or anything you want to move further into the background.
- **Early Reflections.** The first response a room will give to sound is the primary, or early, reflections. The reverb tail that comes directly thereafter is made up of secondary reflections, i.e. not reflections of the original sound source, but reflections of reflections (of reflections... etc). Early reflections (and the intervals between them) are important because they are what establishes the perception of room size, the clue the ear is looking for to roughly determine the size of a room. The larger the room, the further the source is from the walls of the room, and the longer it takes for the early reflections to 'return to the sender'. While humans are not quite as proficient as bats, we do pick these things up...
- **Diffusion.** ...yes, we even pick up the character of the reflections; if the early reflections resemble a 'clean' echo, we know that the surfaces of the room are solid and flat; if the reflections are more blurry and diffused, we can tell that the room is irregularly shaped and/or contains different objects of different texture, causing reflections to bounce all over the place. The Diffusion parameter featured in some RV7000 algorithms is the 'space irregularity booster'. If you find a room simulation too harsh and clinical, you can add a little diffusion - think of it as putting up wallpaper on marble tile walls.
- **HF Damp.** The reason why most reverbs offer the option to dampen high frequencies - so that they fade out more quickly than their counterparts - is that in the real world, high frequencies are absorbed by the air and the surfaces of the room. This phenomenon is more pronounced the larger the room is, so if you have a long reverb with no HF damping whatsoever, you're basically simulating a rather strange empty hall with mirrors for walls and some sort of gas in place of air...!

Hand Over the Goods

Surrender your music to room ambience! Seriously, a little reverb on everything does a whole lot. Sadly, many people use reverb as an "either/or" tool. Maybe you recognize this scenario: You open Reason, create a mixer and put a 2-second hall reverb on one of

the aux sends. Then you start arranging the song, but you leave most of the sounds dry (at least the bass and the drums) except for pads, strings and pianos which all get a nice dose of reverb. Well, that's the classic home studio way - maybe it's just a question of being lazy, maybe it's about CPU economy, or maybe it's the trauma of only owning one cheap hardware reverb before software studios came to town - however, in many a professional recording you'll find that there's some kind of room ambience on just about every sound in the mix. Producers will often use lots and lots of short and barely noticeable reverb programs (the kind that usually goes by patch names like "studio", "closet", "small room", et cetera). The kind you don't notice when it's there, but miss when it's gone.

Here's a demonstration: In this track, there are four different short reverbs. On bars 1-4, there's reverb on all sounds; on bars 5-8, all reverbs are muted, and the mix you thought was 'dry' suddenly takes 'dry' to a new level...! Take this opportunity to observe what we mentioned earlier about headphones vs. speakers; in headphones, the reverb effects are rather obvious, but over speakers they're very subtle. Therefore, always do some A/B monitoring when you're determining how much reverb a mix should have

Now, consider for a moment that Reason is the ideal tool for sticking a bit o'reverb on anything you like: You can have as many reverbs as you please (as long as the CPU can cope) and you can use them both as sends and inserts. Is there any reason not to go reverb crazy? No!

Automation Elation

You may have noticed that the remote panel controls on the NN-XT cannot be automated, and this might lead one to assume that no remote panels in Reason are automation enabled. Not so with the RV7000 however! Here, *all* parameters can be automated, even all the "soft parameters" on the remote panel. Much fun can be had, so we decided to have some.

Please enjoy this piece of musically ignorant mayhem: 7000_automad.rns

Pin the Tail

If you have been following the *Discovering Reason* series, it shouldn't be news to you that we're big fans of CV and Gate. It's one of the unique features of Reason - after all, what

can be more fun that hot-wiring virtual electronics with no risk of frying the circuitry? RV7000 does not offer an abundance of CV/Gate connectivity, but it has a couple of interesting items. The Gate, for example, can be opened via CV (or MIDI). Now we're going to show you how to pin a new reverb tail onto a sound. Here's what we did: First, we routed a few different samples through an RV7000. Then we created a Dr.REX and connected the Slice Gate Output to the Gate Trig input on the RV7000 and enabled the Gate section. Now we have a reverb that only sounds when triggered by the Dr.REX loop slices. Effectively, this means we're adding reverb to the Dr.REX loop, only the reverberation is not derived from the sound of the REX loop but by a completely different sound. Here: 7000_cvgate.rns. If you happen to have a MIDI keyboard connected, you can try pressing any key on the keyboard and you'll find that the gate stays open until you release the key. You may also want to adjust the release time of the gate to adjust the length of the reverb tail.

Bottom line

Always remember that reverb is more than a special effect: It's also a simulation of a natural occurence. Forget the term 'effect' and try to picture it as an 'acoustic backdrop' instead. Observe the mood and the character of the music, and decide what kind of room you would like to put it in. A church? A closet? A hotel suite? RV7000 is ready to take your room reservation.

Text & Music by Fredrik Hägglund

Six Strings Attached

DISCOVERING REASON

For years I've experimented with MIDI guitar with mixed results (so much hardware, yikes!). After getting Reason I've decided to dust that guitar controller off and see what damage can be done.

Hooking it up

A lot of people starting out on MIDI guitar use it where every string transmits out on a common channel. After a short while sonically it becomes one dimensional. However, with your MIDI guitar set to "mono mode" each string transmits on its own separate MIDI channel. My Parker MidiFly transmits over channels 2-7 in this mode. The high E string transmits over channel 2, the B string transmits over channel 3, and so on. Check the manual for your controller to see how it is set up for transmitting over six different channels.

The great thing about using a MIDI guitar with Reason is that each string can go to a separate device that you create. Your low E can go to a Subtractor for bass sounds, while the A string can be sent to a Malstrom, D string to a NN-XT sampler, etc. When you have your controller set to this "mono mode" and you are connected to Reason thru a MIDI interface, you should be able to see six red lights lighting up individually on the "MIDI In Device." This means you are set up correctly and ready to start creating!

Let's start out with an empty rack, then create a mixer and a reverb. Next, let's create a Subtractor and load in "Warm Pads" from the Factory Sound Bank > Subtractor > Pads folder. I usually start out by setting the Subtractor polyphony value to 1 to prevent extraneous notes. Pitch bend for now can be set at zero.

Next hold down the Shift and Options keys (for Mac) or Shift and Control (for PC), click on one end of the Subtractor's rack "ears" and drag it down. Do this four more times and you've got six Subtractors with the same patch and settings, ready to be assigned. Now you can go up to the top of your rack and assign each individual string its own Subtractor by using the pull-down menus on the MIDI In Device. I usually rename the Subtractors after the string that they'll be assigned to, like "E string," "B string," all the way

down to "Low E string."

Make Some Noise

Now you can load up different patches into each of the six Subtractors, pan them around, change the octaves on each voice if you'd like- in other words, try some things and see what you get! Here's an example of six Subtractors plus a groove to play to. I ran the Subtractors and part of the drums through a Scream module just to add some character. For this example I wanted to include a short sequence of me playing the MIDI guitar. Since Reason won't record multiple channels at once, I used a sequencer program to record the parts that you hear all in one pass, and imported the files into Reason. Here are the example files: HexSubtractors.rns |

Crank it Up!

Let's do the same process again but this time it's all Malstroms and while we're just experimenting (no one is getting hurt, right?) let's load all six with different rhythmic patches. I loaded some patches without listening to them just to see what I'd get. However, I did go through the Malstrom patches and set the polyphony to 1, pitch bend to 0 and changed the Modulator rates from triplet values to straight eights or quarter notes. Here's an example of the six Malstroms played over a groove along with an added a bass synth line from the MIDI guitar that I overdubbed. It's just a one take attempt at a bass line that hasn't been fixed, but the input quantize feature really helps get the idea across quickly. (Solo it - you'll hear that it's a one take attempt!) Here are the example files: SixMalstroms.rns |

Hey! Keep it Down Over There!

The next example of using MIDI guitar with Reason involves a happy accident that occurred when I MIDI'd up to a Redrum module. I had my MIDI guitar set to transpose up an octave, and as I played the guitar, I noticed that the high E string on my guitar would mute/solo different channels on the Redrum module. Here's what was happening: Reason is setup so that if you play a C4 into the Redrum, it will solo channel 1. Play a D4, and it will solo channel 2, an E4 solos channel 3, and so on. (Check out the MIDI Implementation pdf chart that came with the program.) The high E string can be used to solo Redrum channels 3 thru 10. How is this helpful? With my MIDI guitar I can now use the first string (the high E) to control different gate patterns for other modules.

You are probably aware that you can use a Redrum module to gate the audio output of another module. With a Redrum's Gate Out connected to a Subtractor's Amp Env Gate Input, you can program different rhythm patterns on a Redrum channel, and the Subtractor's output will be controlled rhythmically by Redrum. I merged several Redrum channel "Gate Outs" to a Spider CV and patched that to a Subtractor's Amp Envelope In. Check out the audio example and the rsn file that goes with it to check it out. (Note: you won't hear the electric guitar in the .rsn file, only in the audio example): TriggeredPatterns.rns |

Musically the example is simple, but it shows how even when strumming the MIDI guitar, the Redrum rhythmic gates can make for a sync'd bass pattern. It can be hard to grasp initially what's going on here, but it can open up new thinking on how to use Reason in a live situation where you use a MIDI guitar to trigger events in tempo. Notice that channel 2 on the Redrum is not muted, but that I've muted channels 3 thru 5. If I avoid the high E string, channel 2 is the "default" rhythm pattern. As soon as I play an E4, F4 or G4, channel 2 mutes and another pattern takes its place, all live and in sync! You can use this to mute and unmute loops depending on where you are fretting the high E string. Add a few other modules (like in the earlier examples) to this setup, and you've got quite a palette for music making. Good luck!

Text & Music by Jerry McPherson

Take it to the NN-XT level

DISCOVERING REASON   **Productivity**

This time around the keyword is Productivity – we're going to learn how to work the NN-XT at warp speed, streamlining the workflow, boosting the power. If you've only scratched the surface before, here is an invitation to dig deeper.

The NN-XT is more than a big brother to NN19; it is by far the most advanced, capable, adaptive and deep Reason workhorse. When Propellerhead got around to creating Reason's flagship sampler, they got a chance to once and for all tackle their main gripe with hardware samplers – the lack of a fast and intuitive interface. With their invariably tiny LCD displays, awkward menu systems, keypad parameter editing and absence of a qwerty keyboard (meaning you had to painstakingly enter sample names like you enter high score initials on an arcade game), most hardware samplers suffered from a bottleneck syndrome in the user interface department – it took aeons to program them.

NN-XT became the opposite – with tons of rotary dials for swift parameter access, a generous display and super-fast macro functions like pitch detection and automapping, NN-XT lets you do in a coffee break what used to take a weekend. Now, let's have a slice of that pie...

Multiplicity

As we know, a sequencer track in Reason can only control a single device. But if that device happens to be an NN-XT, the sky is the limit. While you can't load multiple patches into an NN-XT through the Patch browser, it is still possible to get any number of patches into it. The trick is to use an intermediary "scratch" NN-XT to load the patches, and then copy them one by one into the primary NN-XT. Through this process you can put build a combo instrument featuring dozens of patches. Here's how:

Create two NN-XTs.
Load a patch into the first NN-XT.
Right-click the Group column (labelled "G") on the far left of

the display and select Copy Zones. If the Patch consists of multiple Groups, high-light them all and select Create Group.
- Right-click the display on the second (empty) NN-XT and select Paste Zones.

Done. All parameters pertaining to Group (such as polyphony, portamento) and Zones (envelopes, filters, outputs etc) have been preserved, so essentially the patch has been cloned – with the exception of Global Controls and Main Volume.

Now repeat this process for each new patch you want to add to the "combo" patch. Each time you paste in a new one, it will automatically be created as a new Group. By click-ing in the "G" column you can easily select all Zones within a Group, and when you change a parameter such as Output it will affect all selected Zones – a quick and effi-cient way to route each patch to a separate output pair. Typically, when you create a "combo" patch on a hardware synth you will lose the FX settings for individual Patches, but not so with the NN-XT because you can route each Group to a separate output and any number of FX units.

Note also that there is some measure of selective automation possible, because you can switch Filter on/off for each Zone – hence, the Global Controls for Filter will only affect Zones where the Filter is enabled. If you for example have a Piano+Strings combo, you can switch the Filter off for the Piano, so when you automate the global Freq knob it will only affect the Strings. You can also experiment with different Filter types for different Groups, e.g. HP on the first, BP on the second, LP on the third etc, and sweeping the Global Frequency will yield interesting results.

Here's a Reason Song file featuring an NN-XT with a combination of two bass patches and some additional samples: quadrabass.rns.

Separability

In case you didn't know it, here's a newsflash: NN-XT (as well as NN19) can load REX and REX2 files as Patches. Just like in Dr.REX, the slices are laid out over the keyboard range – one per key – but the possibilities are much greater thanks to all the additional parameters and functions NN-XT offers: per-slice filter, full ADSR envelope, individual outputs and much more. Editing slice parameters in the Dr.REX can be tedious if there are many slices, but in the NN-XT you can save a lot of time by dividing the slices in Groups. The NN-XT is a REX powerhouse.

- Create an NN-XT. Click the browse patch button and load a REX drum loop.
- Create a "partner" Dr. REX. Load up the same REX file you just loaded into the NN-XT.
- Click the To Track button on the Dr. REX.
- Drag the created part from the Dr. REX track to the NN-XT track (you can now delete the Dr. REX, or keep it around in case you need it later).
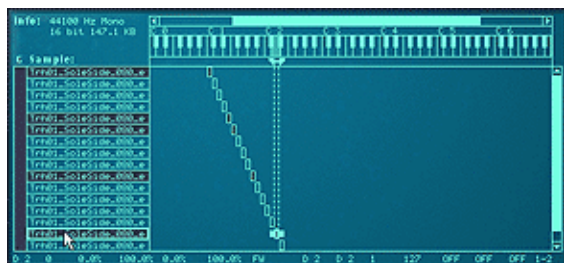
Now onto the fun. First we need to isolate bass drum and snare slices from "the rest". To do this, hold down the [Alt] key and click on each sample name in the left column to audition the slices. Whenever you encounter a bass drum slice, hold [Shift] or [Ctrl] and click on the sample name.

Keep doing this as you scroll down the list, and once you're done you should have all bass drum slices highlighted.

Go to the edit menu and select Group Selected Zones. Now repeat the above steps for the snare drum slices. Voilá: Now you have three Groups – Bass drum, Snare drum and "other" – which will make the rest of the task a breeze.

To the left of the name list you have the Group column. This column now has three vertical bars and by clicking on either bar you highlight snare, bass drum or "other" for editing.
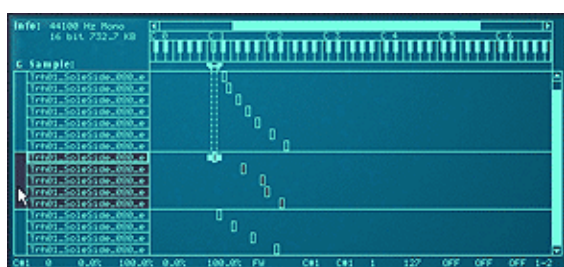
Click on the snare group. Now you can easily route all snare slices to a separate output – turn the Out dial to 5-6. Select the bass drum group. Turn the Out dial to 3-4. Now you have three separate stereo pairs that can be treated externally with individual levels, panning, effects, EQ, mut-

ing etc. You can also now macro edit Groups – changing pitch, level, envelope, filtering, modulation, LFO etc will affect all Zones in the Group.

Here's a couple of examples where the slices have been divided in three Groups with individual FX and other settings (A-B alternating between Dr. REX and NN-XT to showcase the difference): nnxt_rex1.rns | nnxt_rex2.rns

Relativity

The NN-XT is a great starting point for getting a grip on your ReFills. As we know, Re-Fill is a read-only format. And once your sound library has grown to include half a dozen ReFills, some converted Akai ROMs, a bunch of SoundFont banks, a few home made REX files and patches and plenty of WAV samples – phew – it becomes a challenge to maintain a good overview of the library contents. "Where was that great drumkit again...?"

What many overlook is that while samples are locked inside the ReFill, patches are not, even if they use samples from ReFills. This opens up the possibility to build a completely new "personal favorites" patch library for NN-XT/ReDrum/NN19 outside of the Re-Fills, while the references to the actual sample sources will be kept track of by Reason's index. Malström, Subtractor, RV7000 and Scream 4 patches are not ReFill dependent at all, so these can be re-saved and reorganized completely.
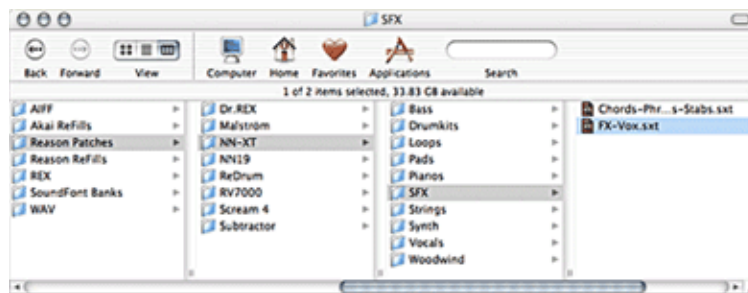
Here's one way to do it:

- Before starting Reason, create a new folder and call it something that makes sense to you, e.g. "Reason Library" or "Reason Patches".
- Create any number of subfolders and organize these in any way you want. You might want to sort your sounds by genre, BPM, device, file type, instrument type – that's all up to you. If you can't make up your mind, don't worry – you can make as many "favorite libraries" as you want.
- At this point you might want to consider gathering all your ReFills, samples, REX files, SoundFonts and other raw materials in one place. The idea is that you shouldn't have to use their location as the primary access point to your sound library again.
- Now start Reason and start working your way through all your ReFills and other library sources, device by device. Every time you stumble upon a "keeper", save

the instrument Patch to the corresponding subdirectory in your patch library. If you can't be bothered with navigating right then, just save all patches to the desktop or some other scratch location – you can always organize the patches neatly later (they can be moved around, as long as you don't move the ReFills they point to).

This procedure may require a few late nights sessions, all depending on the scope of your library, but you will thank yourself later. Think about it – in order to check out a ReFill, you must load the sounds one by one anyway, right? So while you're at it, why not just pick up the habit to hit the Save Patch button each time? Soon enough you've built up an extensive library of favorite patches, organized just like you like it. Now make this one of your four main sound location in Reason Preferences, and you're done.

It should of course be noted that REX is the one format you won't be able to include here since Dr.REX does not have a Patch format as such. However, as pointed out earlier you can load REX/RX2 into NN19 or NN-XT, and then save them as .smp or .sxt files, so that's an option.

Tip: You may want to consider creating "audition" Patches for collections of miscellaneous samples that have no "home". For example, in Reason's Factory Sound Bank you'll find folders like "Other Samples" and "xclusive drums – sorted". These homeless samples are something you might miss out on if your habit is to browse Patches to find what you need. One way to get a grip on all these samples is to load the bulk of them into NN-XT (well, in managable numbers at a time of course), and save as new Patches featuring key maps with one sample per key. So whenever you need a bass drum, an effect sample etc, you just load up your custom NN-XT patch "bass drums" or "FX" and presto: you have all samples of a particular category at your fingertips and can instantly try them out one by one in the song you're currently working on. This is much more efficient that scrambling through samples in the browser window.

Suggested procedure:

Click the NN-XT sample browser button and select all samples you want to include in the "audition" patch. In this illustration it's all the bass drum samples in the Factory Soundbank.

Sample Browser: NN-XT 1

Sample Files: .aif .aiff .wav .wave .sf2 .rcy .rx2 .rex

01_BassDrums

Name	Size
BD11 OHSH.aif	45 K
BD1br.aif	13 K
BD1Dub.aif	10 K
Bd2_Abstract.wav	11 K
Bd2_Abused.wav	16 K
Bd2_Anarchy.wav	32 K
Bd2_Artefact.wav	17 K
Bd2_Belbam.wav	11 K
Bd2_Brat.wav	9 K
Bd2_Brickz.wav	12 K
Bd2_Cheese.wav	6 K
Bd2_Chemical.wav	15 K
Bd2_CutCodes.wav	17 K
Bd2_Dentaku.WAV	5 K
Bd2_DubHead.WAV	6 K
Bd2_FatBoy.wav	8 K
Bd2_GiantStep.wav	5 K
Bd2_HardKnox.wav	14 K
Bd2_HotValve.WAV	11 K
Bd2_Jeepkeys.wav	9 K
Bd2_Kru.wav	8 K
Bd2_MarysI.wav	7 K
Bd2_Monk.wav	11 K
Bd2_NeonLights.WAV	20 K
Bd2_NextStep.wav	18 K
Bd2_One2.wav	28 K
Bd2_Optical.wav	15 K
Bd2_Rabotnik.WAV	9 K

Reason Factory Sound Bank

FACTORY SOUNDBANK

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Preview

Play

☑ Autoplay

Open
Cancel
Help

Itsy Bitsy Spiders - part I

DISCOVERING REASON   **The Idle Predators**

Let's face it, the Spider Audio Merger and Splitter and Spider CV Merger and Splitter have very long and interesting names. But you can't say they work hard. They're the simplest of all Reason devices; they're grey, plain, and easily overlooked - all they do is just sit there and wait for other devices to get caught in their webs. But it's often the simplest things that work wonders, and these babies can fundamentally change and improve your workflow. In this month's article we're going spider hunting - bring the magnifying glass and follow us on a field trip to the tangled forest habitat of Spider Audio.

Fields of Interest

Here's a quick rundown of things you can do with Spider Audio:

Creating sub-groups - routing multiple signals to a single mixer channel.
Merging effect input signals - tapping multiple signals into a single effect unit.
Merging carrier or modulator signals - splicing multiple sources for vocoder treatment.
Splitting signals to alternate targets - when you need A/B/C/D alternatives for a signal.
Splitting/merging the same signal - let 'cloned' signals part ways, be processed, then reunited.
Create "pseudo" stereo effects - split a mono signal in two, apply effects, pan left and right

Now for some practical cases.

Effect-ivity
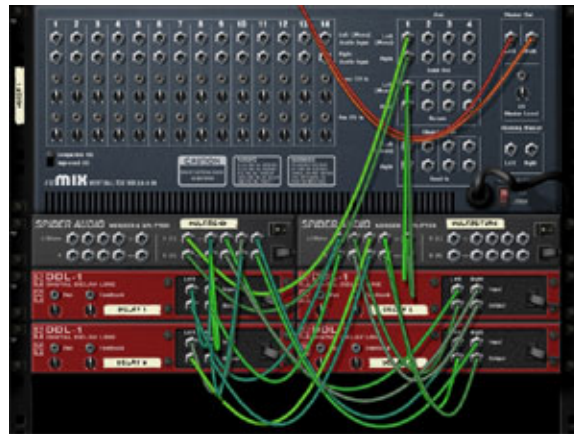
The 14:2 mixer has four auxiliary sends/returns. Depending on the complexity of your setup, this may not be enough. But oftentimes you find yourself applying the same amount of, say, aux 1 and aux 2 to a channel - for instance, if you're using two or more DDL-1 units to create a stereo or multitap delay. Here, Spider Audio comes in handy because you can split the aux send to four separate signal, as well as merge four separate signals and send them to the aux return. In the example shown here we've taken four DDL-1 delays and built a multitap delay - each with an individual sequencer track for automation. And even with four delays hooked up, we still have three aux channels free. This method is useful for any setups where you want two or more effects on one Aux channel. Multiple delays is one; another example is chorus and reverb (as opposed to a chorus+reverb chain), or virtually any signal you want to make sure is sent at equal volume to two separate targets.

Parallel Slalom

If you're prepared to take it to extremes, consider using a whole army of Audio Spiders (the example here shows 14 units) to create one giant patchbay. Yes, there will indeed be lots of tricksy routing involved initially, but if you do it once you won't have to do it again - we encourage you to save a template song once you've emerged safely on the other side.

The idea is to connect your instruments to Spider Audio units rather than a main 14:2 mixer. The Spiders then connect to the mixer, one for each channel - and so far everything will be normal - but the Spiders also break out to a secondary entity. Like what? We present you two scenarios:

1. ReWire

If you've ever found yourself having to unplug all your devices from the mixer and re-routing them to the Hardware Interface so that they show up as channels in the ReWire host application instead, the method described here might be just the ticket. Just connect each Spider to both the main mixer and the Hardware Interface, and you have two parallel routing schemes up and running - and you won't have to disassemble your setup when you want to move over to the ReWire domain. Just mute any channel in the Reason mixer and it will be left out of the main mix; but its signal is still going out to the ReWire host where it can be activated and get its own channel strip. If you want to switch back, unmute the channel on the Reason mixer and mute it in the ReWire host mixer. At no time will you need to mess with cables.

2. Alternate mixes

Another way to employ this "dual lane" method is to route all your devices to two separate 14:2 mixers. That way you can have two alternate mixes going, and switch between them - easily done by pulling the master level on mixer #1 down, and the master on mixer #2 up. There are many applications for this - one is to have a primary mixer where you're doing what you believe is the perfect mix, and a secondary 'scratch' mixer where you try out crazy ideas without running the risk of messing up your delicate settings on the primary mixer. Or, you might want to work on two completely different mixes simultaneously, either because you can't make up your mind which direction to go with the sound, or because you want to challenge yourself to creating a remix while you're still working on the original.

Since each Spider has four split outputs, theoretically you can have both the ReWire routing and the standard routing, and the alternate mixer routing, and yet another one of your choosing, all active at the same time. But be aware that the cable mess will reach the boiling point in no time, so keep track of what you're doing!

In the next article we will go on another field trip to look at Spider Audio's close relative, Spider CV - its web threads are finer, but ever so strong.

Text & Music by Fredrik Hägglund

Itsy Bitsy Spiders - part II

DISCOVERING REASON   **Spidey Senses Tingling**

Right, so... where were we? Oh yes, Spider CV Merger & Splitter, the other eight-legged freak in the Reason terrarium. Now, this is the evil twin of Spider Audio, and this one you have to handle with caution because the webs of cables that can build up around these things can become more or less impenetrable. While it may be tempting to try and be as economical as possible (for example, using both the split and the merge sections on a single unit), we encourage you to spread the burden across as many Spider CV units as possible - they are the most CPU friendly Reason units of all, and it's going to be a lot easier to track down the right cable if they aren't all on top of one another. And now, on with the show.

Matrix Marionettes
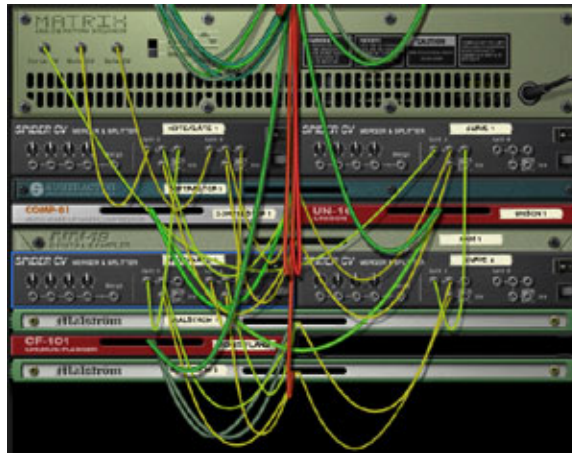
The first (and fairly obvious, but fun nonetheless) setup we'll try is all about stacking. Since a single Matrix unit can now play as many synths as the CPU can handle, you can build up sounds by using any mixture of Subtractor, Malström or NN-sampler sounds. If you run out of split outputs, just add another Spider CV unit to the chain. In the example file stakkabass.rns we've built a massive bass sound with two Malströms, one NN-XT and one Subtractor. For this you actually need four Spider C units; the Matrix has three outputs (Note, Gate, Curve) and since these need to be branched out into a total of twelve cables for four synths, you will need to link a few CV units together in order to serve all four. Check the example file, turn the rack around and see if you can follow that spider...

Side-splitting Automation

Mixer automation is something one will usually let a main sequencer track take care of - and that's great, it offers complete and minute control over all parameters. But sometimes you wish that automation itself could be automated, and thanks to the Spider CV split function this is possible - at least as far as channel Level and Pan is concerned. With the aid of a single Spider CV unit you can control the Level and Pan of three channels at once, and by linking multiple units together it's theoretically possible to control all fourteen channels, either all at once or through some customized sub-grouping scheme.

But let's not go nuts like in the picture here - let's stick with the three-channel idea for now. What we'll do first in this example - splittermix.rns - is create a 14:2 mixer and three Dr. REX. Then we'll create a Spider CV unit and route the split outputs A and B to level and pan, respectively, on the three active mixer channels. And then we'll create... a Matrix? No, been there, done that. Instead we'll use a Malström - not for its sound capacities, but because it features Modulators A and B, two excellent LFO-type modulation sources with a massive amount of crazy curves in store. Using Mod A for level and Mod B, we've got automation of Pan and Level covered, and since the Malström itself can be automated from its own track, the modulation curve types and rates can be changed during the song.

Hit play in the example file and hear this author haphazardly messing around with the settings, and remember that the only automation going on here is on the Malström track. The Malström comes highly recommended as a modulation source if you're looking for things you can't get out of the standard LFO waveforms and the Matrix is too aliased.

Submerged

Another situation where the Merge functionality proves highly useful involves ReDrum. Right, we already know several ways of using CV for rhytmic modulation: You

can use a Dr. REX and let the loop slices send gate signals. You can synchronize a unit's built-in LFO to MIDI. You can "borrow" the LFO from device and modulate another. You can use Matrix Gate and Curve data. So far, so good. But what if you want a Re-Drum to do it? The thing with that is, ReDrum sports per-channel Gate outputs, so you can't have an external target modulated by the entire drum pattern, only by one drum at a time. Here's where Spider CV's merge function shines: You can take the Gate output of up to four ReDrum channels and merge them into one. And of course you can set up multiple Spider CV units to merge the Gate signals from all ten ReDrum channels. So let's try this:

Submerged.rns - In this example file, the idea is to merge two separate groups of Re-Drum Gate signals and have them modulate one instrument each. We've got the drums, a bass and a pad-ish sound.

Now, having both the bass and the pad modulated exactly the same way will sound a little chunky, so here's what we'll do: We merge the bass drum and snare drum Gate signals and route these to the filter on the bass synth. For a secondary modulation source, we merge two of the hi-hat channels - this will provide the modulation pattern for the pad sound. Both the bass and the pad are run through ECF-42 filter units, and these are what the ReDrum will modulate. Having the bass filter retrigged each time a bass drum or snare hit occurs provides a nice tight groove at the bottom, and it gives a nice symmetry to have the high end percussion (the hi-hats) control the high end synths.
In this particular example the ReDrum is doing its normal job as a sound device too, but this isn't a requirement of course - it can be silent and act only as a CV/Gate "puppet-master".

Bottom line

Can't think of one. Enjoy the new site, and Merry Christmas!

Text & Music by Fredrik Hägglund

Filter Up

 This installment of Discovering Reason is adapted from my book, *Power Tools for Reason 2.5*, and describes how to create a high pass filter using the BV512 Digital Vocoder. This is an independent insert effect controlled by cutoff frequency and resonance sliders, and the parameters can be automated with a sequencer track or modulated by CV sources. When filtering a signal though the BV512, the results are not perfect, however in the right situation, the configuration can create some unique effects. The audio and CV modulation principles used in this project demonstrate some alternative ways of using the BV512 CV modulation features.

NN-XT High Pass Filter

There is an alternative method of processing sounds through a high pass filter, and before proceeding with the project, I'll briefly explain how this is achieved. First, you need to solo out the signal that you want to filter. Now, render this signal to an audio file using the export loop or export song feature. Create a NN-XT and load the audio file into a sample zone. Program the sequencer to trigger the sample, and during sample playback, the signal can be processed using the NN-XT LP12 filter. This is a tedious process, but it provides the best results possible without using other software or hardware. If you are willing to sacrifice some audio quality, then the 24dB high pass filter configuration is an interesting alternative.

Example file: tedious_hp_method.rps

How the High Pass Filter Works

The high pass filter configuration starts with raw, unfiltered white noise generated by a Subtractor synthesizer. The Subtractor audio output is connected to the modulator input of a BV512 Vocoder in 32-band mode. White noise is analogous to "white light," the mix of all colors in the visual spectrum. Likewise, combining all frequencies in the audio spectrum creates white noise. When the BV512 receives white noise on the modulator input, all vocoder bands open, because equal amounts of all frequencies are present.

Sweeping the LP24 cutoff frequency closes the vocoder filters from the high frequency bands down to the low frequency bands. In essence, the BV512 filter bands mimic the

behavior of the Subtractor LP24 filter, and carrier signals will be filtered like a low pass filter.

A second BV512 Vocoder, set to FFT mode, is added to the rack, and the Individual Band CV Outputs of the first BV512 are connected in reverse order to the Individual Band CV Inputs on the second BV512 Vocoder. The reverse cabling causes the second vocoder to mirror the LP24 behavior, and the filters close from the low frequency bands up to the high frequency bands. Carrier signals processed through the second BV512 are filtered like a 24dB high pass filter.
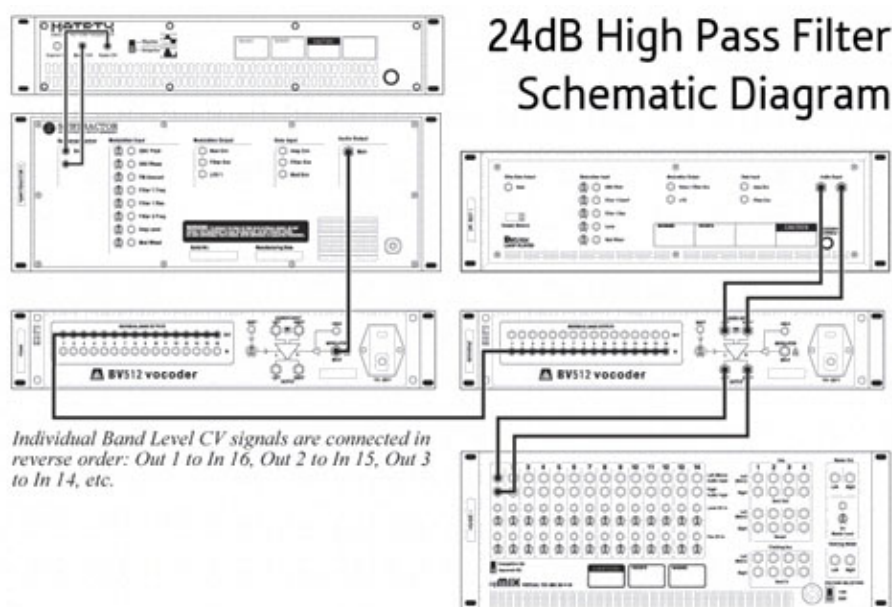
Click image for larger version

Creating the High Pass Filter Effect

The schematic (above) presents a clear overview of the device cabling. The high pass filter is comprised of several different stages, and the directions are grouped to specify the connections and parameter settings of each stage. An example RNS file is provided below, but you should try building the effect yourself.

- Start with an empty rack, and create a Mixer
- Set the Song Tempo to 130 BPM

White Noise Modulator Signal Source

- Bypass Auto-Routing (hold down the shift key) and create a SubTractor Synthesizer
- Set the SubTractor Polyphony to 1
- Enable the Noise Generator
- Set the OSC Mix to 127
- Set the Filter 1 type to LP24
- Set the Amp Envelope attack to 0, decay to 0, sustain to 127, and release to 10
- Set the Velocity to F.Env modulation to 0
- Set the Master Level to 100

- Create a Matrix Pattern Sequencer connected to the SubTractor Sequencer Control Inputs
- Set the Matrix Pattern length to 1 Step, and program a tied gate event on Step 1

Carrier Signal Source The Dr.REX loop provides a source to demonstrate the filtering, and can be replaced with any mono or stereo audio signal source.

- Bypass auto-routing and create a Dr.REX Loop Player
- Load the ReCycle Loop, "130_NuYork_mLp_eLAB.rx2" from the from Reason Factory Sound Bank\Music Loops\Variable (rx2)\Uptempo Loops\ directory
- Set the Dr.REX Master Level to 108
- Copy the REX slice data to the Dr.REX 1 Sequencer Track

Carrier Vocoder Section

- Bypass auto-routing and create a BV512 Digital Vocoder
- Connect the Dr.REX Audio Outputs to the BV512 Carrier inputs
- Connect the BV512 Carrier outputs to the Mixer Channel 1 Inputs
- Rename "Vocoder 1" BV512 to "Reverse"
- Adjust the following parameters on the BV512:

Reverse BV512 Vocoder Settings

Mode	Vocoder
Band Count	FFT(512)
Hold	Off
Attack	0
Decay	0
Shift	0
HF Emphasis	0

Dry/Wet	127
Band Levels	

Modulator Vocoder Section

- Bypass auto-routing and create a second BV512 Digital Vocoder
- Rename the second vocoder to "Main"
- Connect the SubTractor Audio output to the "Main" BV512 Modulator Input
- In reverse order, connect all 16 Individual Band level Outputs from the "Main" BV512 to the Individual Band level Inputs on the "Reverse" BV512. In other words, connect Individual Band Level Out 1 to Individual Band Level In 16; Band Out 2 to Band In 15; Band Out 3 to Band In 14; etc.
- Adjust the following parameters on the "Main" BV512:

Main BV512 Vocoder Settings

Mode	Vocoder
Band Count	32
Hold	Off
Attack	0
Decay	0
Shift	0
HF Emphasis	0
Dry/Wet	127
Band Levels	

Run the sequence.

As the loop plays, adjust the Subtractor Filter 1 cutoff frequency to sweep the high pass filter. Because the modulation is reversed, a filter cutoff setting of 127 is fully open, and decreasing the filter cutoff will sweep the high pass filter up. You will first notice that the audio contains artifacts caused by the chaotic levels of the noise modulator source, and as indicated earlier, this effect may not be suitable when audio quality is important.

Example file: BV512-HP24Filter.rns

BV512 Envelope Settings

The chaotic impulses of white noise cause the BV512 band levels to jump around errati- cally. At the expense of increasing response time, a longer decay time smoothes out the glitchy artifacts. Increasing the "Main" BV512 decay time to 54 or greater will smooth

these out. If you use the high pass filter for fast sweeps or gating, lower decay settings are recommended.

Different Filter Types

Using the spectrum analyzer, compare the Subtractor filter to LP12 response with the LP24. You can see how the LP24 band reject response is sharper than the LP12 filter type. You can also create a 36dB filter with a very steep cut by enabling Subtractor Filter 2 with Link on, and setting the cutoff frequency to 0. Also, the "Main" BV512 Vocoder can be used as a low pass filter to create a merging filter sweep effect where two signals are cut and open simultaneously with one being filtered down and the other filtered up.

Example File: BV512-HP36vLP36.rns

Resonant High Pass

Adding resonance to the high pass filter will sound a bit harsh because the resonance spike will overload the BV512 modulator input. Compensate for the resonance gain by reducing the Subtractor Master Level. For example, set the Master Output Level to 42 and set the Filter 1 Resonance to 100. For other settings, balance the two controls so that the modulator signal peaks do not overload the inputs and vocoder band levels

LFO Modulation

Filter 1 can be modified by any of the Subtractor modulation sources, and the filter modulations subsequently affect the high pass filter. Try the following modification to create a tempo synchronous sweep using LFO1:

- Start with the 24dB High Pass Filter Configuration from above
- Modify the following Subtractor Parameters:
    - Filter 1 Cutoff Frequency to 63
    - Select the ramp wave form (3)
    - Enable LFO 1 Sync
    - Set the Rate to 4/4
    - Set the Amount to 120

Run the sequence

Example File: BV512-HP24_LFO.rns

Envelope Controlled High Pass Filter

Using a gate CV from a Matrix connected to the Subtractor Filter Envelope Gate Input, you can create a pattern controlled high pass filter. Rhythmic patterns programmed on the Matrix will trigger the Filter Envelope without interrupting the white noise signal.

- Start with the 24dB High Pass Filter Configuration from above
- Modify the following Subtractor Parameters:
    - Filter 1 Cutoff Frequency to 127
    - Filter Envelope A: 16, D: 85, S: 105, R: 41
    - Filter Envelope Invert: On
    - Filter Envelope Amount to 127
- Bypass auto-routing and create a second Matrix Pattern Sequencer
- Rename the Matrix to "PCF Gate"
- Connect the "PCF Gate" Matrix Gate CV output to the Subtractor Filter Env. Gate Input
- Program tied gate events on steps 1, 2, 5 and 6
- Program normal gate events on steps 3, 8, 9, 11, 13, 14, 15, and 16

Run the Sequence

The high pass filter adds a new twist to pattern controlled filtering because it uses inverse envelope modulation. The envelope parameters on the "Main" BV512 can be used to further shape the response of the PCF. Zero attack and zero decay times lend to very sharp patterns, but a decay setting of 30 or greater adds an interesting lag to the effect.

Example File: BV512-HP24_PCF.rns

Text & Music by Kurt Kurasaki

Although this article is released exclusively for the Discovering Series, the 24dB high pass filter project reflects the general style and content in Power Tools for Reason 2.5. Written for experienced Reason users, this book contains similar example projects that illustrate how various audio and modulation principles are applied in Reason. The book is structured on modular synthesizer and audio engineering principles, and the topics include control voltage routing, audio routing, compression techniques, vocoding tricks,

digital signal processing effects, rhythm programming, synthesizer patch programming, and a list of timesaving shortcuts. For more information, please visit www.peff.com

Power Tools for Reason 2.5 is available for online purchase directly from the publisher, Backbeat Books, and will be available at book and music stores in February 2004. For special sales and educational orders, contact Kevin Becketti at Backbeat Books (866) 412-6657 or email info@backbeatbooks.com

Go With the Workflow

 Alright! A brand new version is out and it's time for another round of Reason discoveries. Version 3.0 introduces enough possibility multipliers to keep us going at full steam for a long time to come. But before we sink our teeth into the headlining stars - the Combinator and the MClass Mastering Suite - we're going to start out light with a little warm-up routine and explore some of the workflow enhancements introduced in version 3.0, as well as some golden oldies that haven't been universally absorbed yet. Streamlining the workflow saves time, minimizes frustration and helps keep the creative juices flowing without interruption when the inspiration kicks in. Learn to use Reason 3.0 efficiently and save hours, bucks - and your hair.

Powerbrowsing

Reason's proprietary file browser has been given an extreme make-over and now sports a search engine, favorites and advanced preview functionality. The most fundamental change is that the program now offers a patch centric approach as an alternative to the device centric one; you no longer have to think in terms of "should I go for a Malström bass or a Subtractor bass?" - instead, you can simply browse any and all bass sounds and preview them directly without leaving the browser. Not only can you stop worrying about which device a patch happens to belong to, you can even replace a rack device with a different one without losing routing, mixer settings or sequencer track assignment and data.
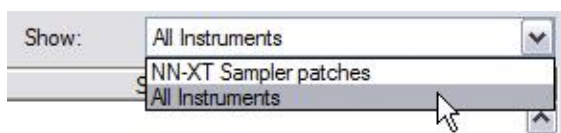
 The key to patch centric browsing is the Show menu, where you can select the view option "All Instruments" (and/or "All Effects", depending on situation). By selecting this view the browser is no longer constrained to patches for the device you opened it from. In fact you don't even have to create a device before you load a sound - you can simply select "Create Device by Browsing Patches..." from the Create menu and have the device created for you once you've found the right patch.

The Factory Sound Bank offers two special folders entitled "ALL Instrument Patches" and "ALL Effect Patches for the purpose of browsing by sound or effect category rather than device model. These two folders contain duplicates of all the patches found in the device categorized folders that make up the rest of the sound bank.

Locating sounds this way is very simple:

- Select "Create Device by Browsing Patches..." on the Create menu (you can also right-click in an empty part of the rack and select it from the context menu)
- Expand the ALL Instrument Patches folder.
- Expand the sound category subfolder of your choice.
- Now you can see all instrument patches pertaining to that category.
- Click on any patch and a temporary rack device will be created, allowing you to preview the patch by playing your keyboard.

Now, let's try replacing one device with another:

- Start with an empty rack.
- Create three devices in the following order: Mixer 14:2, Malström, Matrix.
- Program a Matrix sequence and leave it playing.
- Open the Patch Browser from the Malström.
- Select "All Instruments" on the Show menu.
- Browse to the Subtractor Patches folder and click on any patch.
- Notice how the Malström is temporarily replaced with a Subtractor.
- To make the change permanent, load a patch by double-clicking on it.
- Hit Tab to flip the rack around.

As you can see, the CV/Gate cables as well as the audio cables are now connected to the new device. This works with all patch based devices, though there are obvious exceptions - for instance, if you have connected multiple outs from an NN-XT and replace it with an NN19, you will lose all connections except the main stereo out.

Management by Search

Have you ever felt that urge to organize your sound library once and for all, but you can't come up with a failsafe way to categorize them? "Do I put this Choir patch in Pads, Vocals or Atmosphere?" With the new Search function in the browser, you now have the possibility to forget about the categorization dilemma and instead let the browser do the thinking. The new browser lets you search across multiple ReFills, directories or even all local media. However, it won't help you find files that don't want to be found - file names like "My Patch 34" are bound to fall outside of any search results. So what you need to do is name all your files in accordance with a "virtual index". For example, REX

loops have various attributes such as BPM, music genre, time signature, length, instrument and so on. Which of these attributes to include in file names is up to you, but the more detailed the file names the more flexible your virtual index will be. The Factory Sound Bank uses a file name syntax that can serve as a good starting point:

GenreNN_Name_BPM_Manufacturer.rx2 (*example: Dub01_RoomRoots_060_eLab.rx2*)

Genre, a descriptive name, and original BPM are all useful for searching. The manufacturer name may be of less interest, and could be replaced with the name of the collection the loop belongs to, or perhaps any of a list of adjectives you often attribute to your sounds - "hard", "heavy", "lo-fi", "ambient"...

My Favorites Thing

Having unlimited choices in a creative situation can be frustrating, and one good way to narrow down the number of choices is to handle the hunt for good sounds as a separate process. One way to do this is to make a temporary Favorite list for the song you're currently working on. There you can gather all the best candidates for the arrangement you've pictured in your mind, and this way you won't have to stop and plod through thousands of sounds later on when you're in the middle of recording.

Suggested procedure:

- Start with an empty rack and create a mixer.
- Right-click in the empty area below the mixer and select "Create Device by Browsing Patches..."
- In the browser window, click the New Favorite List button.
- Double-click on the new list to rename it (a relevant name might be the working title of your song, or simply "Candidates")
- Make sure "All Instruments" is selected on the Show drop-down menu.
- Locate the "All Instrument Patches" folder in the Factory Sound Bank.
- Now you can begin looking for patches and REX files that might be suited for this particular project. Each time you click on a patch, a temporary device will be created so that you can play it from your MIDI keyboard without leaving the browser window.
- Whenever you strike gold and find a patch that might fit the bill, simply drag it from the main browser area and drop it on your Favorite list. You can of course go

outside the Factory Sound Bank and add sounds from any ReFill or disk location, the Favorite list is location independent.

Once you have built up a decent collection of candidates, you can close the browser and get busy. When you want to bring in one of the sounds, simply right-click in the rack and select "Create Device by Browsing Patches..." again. In the browser, click on the Favorite list icon and the browser will show all the sounds you hand picked earlier.

The Little Things

Know thine keyboard shortcuts!

Some functionality that users request on a regular basis has been around since version 1.0. Manual reading is not for everyone, but the Keyboard Key Commands document is only four pages long and holds many secrets and you will thank yourself for reading it. Tip of the day: Holding the Shift key while drawing velocity levels in the Edit grid will exclude all notes except the selected ones. This lets you edit the velocity of a single note buried inside a chord.

Wondering where the browser's "Up One Level" button went?

Fear not, the Backspace key on your computer keyboard now substitutes for this button.

Having trouble designing your ideal default song?

Well, why have one when you can have many. Make shortcuts/aliases for several different default songs and accustom yourself to starting up Reason by using these rather than the standard shortcut that launches the main program. Instead of "Reason", you can have "Reason - Empty Rack", "Reason - Basic setup", etc.

Frequently accessed locations?

You can drag any folders, ReFills or drives to the Locations list in the Browser. You can also drag folders from inside of ReFills to the Locations list. If you're working on a project where you often find yourself navigating to the same location, inside or outside a ReFill, seize the opportunity to place this location in the list and it will be a single mouse click away from then on.

And that concludes this month's Discovering Reason. Stay tuned for the the upcoming articles which will dive deep into the heart of the Combinator and dig for gold in MClass territory.

Text & Music by Fredrik Hägglund

The Hitchhiker's Guide to the Combinator - Part I

DISCOVERING REASON It was a long and winding road, but Reason has finally been armed with a tool that lets you build your own Frankenstein's monster out of the existing devices. The Combinator unleashes the full power of Reason and fulfills the promise set forth by the "workstation" label. For it has long been a staple of the hardware workstation paradigm that you can combine single patches into superpatches tailored for different needs such as live performance splits or giant stacked pad sounds. But in the hardware world, combining sounds usually entails some nagging limitations - the most obvious of these being that the integrity of a single sound is compromised once you move into the multi-timbral domain, where the fixed set of onboard effects suddenly goes from local to global. This invariably leads to those single sounds losing some of their power, all depending on how heavily they relied on the effects. Not so with the Combinator, where each combined sound lives in its own microcosm and one sound doesn't need share any resources with another. Well... apart from the CPU. You can't have it all!

Combination of One

Before we lose ourselves in outrageously elaborate power-user routing exercises, we're going to focus on a simple theme: How can the Combinator bring something new out of one particular device? In a two-part article we will build four fairly simple Combinators dedicated to four different Reason devices, and see how the Combi can improve upon these devices by extending their functionality, both through the Combinator itself and through whatever supporting devices can make the device more powerful. Deluxe versions of the original devices, if you will; useful allround-Combinators which are task based rather than sound based.

One of the fun things about the Combinator is that it lets you redirect CV signals to parameters that are normally not accessible via CV. This lets you control virtually any parameter via CV - to name but one example, the waveform selection on the Subtractor (we will get to that in part II) and the Step parameter of the DDL-1 delay. And of course, CV/Gate has always been a favorite here at the Discovery camp - it's one of those things that make Reason special, something you can tinker with in your proverbial garage. And now with the Combinator you can even do custom paint jobs. Without a safety mask, even.

OK. Less talk, more action.

Dr. REX / Extended

Since Dr.REX is a device which doesn't save and load patches, it will always default to certain settings when created - settings which you may or may not agree with. After creating a Dr. REX you may often find yourself routinely activating high quality interpolation and LFO sync, adjusting the volume, adjusting the velocity amp setting, the pitch bend range, filter and amp envelope, LFO settings and so on. By simply placing an empty Dr.REX inside a Combinator and saving that Combinator patch, you will have quick access to a Dr. REX with all the default settings just how you like them...!

In this Combinator we will also supply the Dr. REX with a couple of useful assistants, namely two Matrix sequencers: One for sequencing the pitch, the other for modulating the parameter of your choice. We will also add an MClass Maximizer as a quick remedy for loops that are too low or uneven in volume and you don't have time to fiddle with levels.

The pitch sequencer is based on a method described in the very first Discovering Reason article, entitled Ask Dr. REX!. The basic idea is to use a Matrix sequencer to control the pitch. This eliminates the process of editing the pitch of each individual slice, which can be a tedious task. The alternative method used here allows you to simply draw in notes in the Pitch sequencer Matrix. This is useful not only for instrument and vocal loops, but for drum loops as well. For example, this simple sequence will pitch up the snare drum by an octave on any loop that features snare hits on the 2 and the 4.
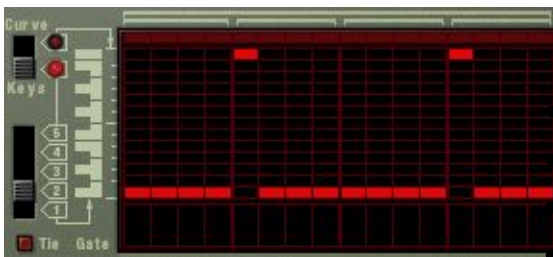
Here's what we need for this Combinator:

1 Line Mixer 6:2
1 Dr. REX
1 MClass Maximizer
2 Matrix sequencers

Audio routings

The Dr. REX stereo output goes to the first channel on the Line Mixer. The main out from the mixer is connected to the input on the MClass Maximizer, and the Maximizer's stereo output is routed to "From Devices" on the Combinator. Simple as that.

CV/Gate routings

Connect the Note CV output of the first Matrix sequencer (the one for controlling the pitch) to the OSC Pitch input on the Dr. REX, and turn the CV adjustment knob (next to the OSC Pitch input) full right. Connect the Curve CV output of the second Matrix sequencer to any of the modulation inputs on the Dr. REX - in the following demonstrations we will use both the Level and the Filter 1 Cutoff inputs.

Rotaries

Nothing fancy here, we will simply use the rotary controls to substitute for a selection of controls on the Dr. REX front panel. The first will control Filter frequency, the second will be assigned to Filter envelope amount. The third will be a Transpose control and the fourth will control LFO amount.

Buttons

The first button will toggle the Maximizer between On and Bypass mode. The second will act as an on/off switch for the Filter. The third will enable/disable pattern playback on the Matrix sequencer controlling the pitch, and it will simultaneously shift the Dr. REX pitch by two octaves in order to compensate for the Note CV signal offset (for more on this, see the "Ask Dr. REX!" article). The fourth button will enable/disable the Matrix sequencer responsible for additional modulation.

Combinator Programming

Device: 3	Dr REX Ext		Min:	Max:
Source:	Target:		Min:	Max:
Rotary 1	Filter Freq	▾	0	127
Rotary 2	Filter Env Amount	▾	0	127
Rotary 3	Transpose	▾	-12	12
Rotary 4	LFO1 Amount	▾	0	127
Button 1		▾		
Button 2	Filter On/Off	▾	0	1
Button 3	Osc Octave	▾	4	0
Button 4		▾		
		▾		
		▾		

Most of the programming required here deals with the Dr. REX, of course.

For Rotary 1, select Filter Freq from the menu and leave the values at Min: 0, Max: 127. Select Filter Env Amount for Rotary 2 and use the default values. Rotary 3 is the Transpose control and should be set to Min: -12, Max: 12. Rotary 4 is the LFO Amount control - again, use the default values.

Button 2 is the Filter on/off switch, and values should be Min: 0, Max: 1. Finally, Button 3 is the one that toggles the whole Pitch Sequencer thing on and off, so it will control both a Matrix Sequencer and the 'Osc Octave' parameter on a Dr. REX. It should be set to Min: 4, Max: 0. Why? Because when the Pitch Sequencer kicks in, the modulation CV signal will offset the pitch by 4 octaves so the Dr. REX must compensate for this.

OK, that's the Dr. REX, but there's a couple more things to do in the programmer.

Bring up the Modulation Routing for the MClass Maximizer and assign the parameter 'Enabled' to Button 1. Values should be Min: 2, Max: 1. This means that when the button is in the Off (Min) state, the Maximizer will be in Bypass mode, and the On (Max) state will put the Maximizer in On mode.

Finally, there's the two Matrix sequencers. For the first one, assign the parameter Pattern Enable to Button 3 (Min: 0, Max: 1). For the second Matrix, do the same for Button 4.

The finished Combinator should look like this: dr_rex_extended.cmb (custom skin included). Now let's look at a couple of examples of the Dr.REX/Extended in action.

rex_pitchdemo.rns - Here we've taken three different guitar loops (one acoustic, two electric) from the Factory Sound Bank. All three involve strumming a single chord (major) and they have a similar rhythmic feel. Now, with the pitch sequencer onboard, changing the pitch of the chords is a walk in the park - simply draw notes on the pitch sequencer Matrix display, and you have a different guitar loop. Since the Factory Soundbank guitar loops are available in Major, Minor, Cmaj7 and so forth, you can patch together a guitar track for a whole song using 2-3 Dr.REX/Extended and alternating between them - or you can build upon this Combinator even further by adding another Dr. REX and then reprogram a Combinator button so that it mutes one Dr. REX and unmutes the other, allowing you to alternate between two loops with the click of a

button.

rex_moddemo.rns - Starting with the above example, we put the secondary Matrix sequencers to use by having each one modulate its Dr. REX in its own way. On the acoustic and one of the electric guitars it modulates the level, resulting in a gate/tremolo-like effect. On the second electric guitar it modulates a resonant bandpass filter, approximating a wah-wah effect. The idea with the Modulator sequencer in this Combinator patch is to have it standing by in case you want to modulate something real quick - level, filter frequency, resonance...

The best way to use these kinds of task-based "template" Combinators is to put them in a separate Favorites list and call it Create, Templates or something to that effect. Then, whenever you want to call up one of them you simply use the 'Create Device By Browsing Patches' command, which brings up the browser. There you click on the Favorites list where you keep your template Combinators, and presto - the machine is inserted into the rack.

Multi DDL-1

Another device which, through the Combinator, is open for CV control of parameters previously off limits to CV/Gate is the DDL-1. Toying around with the DDL-1 in the past, you may have noticed - by accident or other - that changing the number of steps on the fly while the delay is processing a signal, causes a fun side effect reminiscent of pitch bend. What we're going to do here is build a multitap delay comprised of three DDL-1 units, and featuring the following additional goodies:

- - Optional Matrix control of the Steps parameter on each of the three delays
- - One ECF-42 filter for each of the delays, optionally controlled by a Matrix
- - Chorus
- - Phaser

This Combinator patch lets you do a number of nifty things. Its most basic function is to serve as a straightforward multitap delay with the option of adding phaser and/or cho-

rus as icing on the cake. The filters let you give the different delay signals different flavors, and the filter sequencer lets you do cool stuff like gradual HF damping.

OK, here's our shopping list for this baby:

- 3 DDL-1 delays
- 3 ECF-42 filters
- 1 CF-101 chorus/flanger
- 1 Spider Audio
- 1 Spider CV
- 4 Matrix sequencers

Audio routings

No mixer is required here. Instead we route the signal from the Combinator's 'To Devices' to the Splitter input on the Spider Audio. From there we split the signal to each of the three DDL-1 units. We then route the outputs from first DDL-1 to the inputs of first ECF-42 and repeat this for all three pairs of delays and filters. We then route the outputs of the three filters to the Merger input on the Spider Audio. Finally, the merged output from the Spider Audio is routed to the Chorus input, the Chorus output to the Phaser input, and the Phaser output to 'From Devices' on the Combinator.

CV/Gate routings

We've got no less than four Matrix units here. One of them will be used to control the cutoff parameter on the three filters. The other three will be used to control the Steps parameter on each DDL-1, respectively. So: Route Curve Out from Matrix #1, 2 and 3 to Rotary 1, 2 and 3 on the Combinator. Route Curve Out from Matrix #4 to Split A In on the Spider CV. Route Split A Outputs 1, 2 and 3 to the Freq CV In on ECF-42 #1, #2 and #3.

Rotaries

The three first rotary knobs on the Combinator will be devoted to the Steps parameter on the three delays. When the "Step sequencer" is active, the three knobs are overridden by the three Matrix sequencers doing the automation; however when when the "Step sequencer" is inactive, they can be used for manual control. Incidentally, the dotted scale on the rotary dials corresponds to the number of steps (1-16). The fourth rotary knob will control the global Feedback.

Buttons

The first button will be assigned to Pattern Enable on the "Filter sequencer". The second button will enable/disable the three Matrix units that control the Step parameters. The third and fourth button will enable/disable the Chorus and the Phaser, respectively.

Combinator programming

For the first DDL-1, assign Rotary 1 to the "DelayTime (steps)" parameter and leave it at Min: 1, Max: 16. Repeat this for the second and third DDL-1, but use Rotary 2 and 3, respectively.

For all three delays, assign Rotary 4 to the "Feedback" parameter and leave it at Min: 0, Max: 127.

Each of the three Matrix units that control the DelayTime (steps) parameter should have the following modulation routing on the Combinator: Button 2 - Pattern Enable - Min: 0, Max: 1.

The Matrix that controls the Filter Sequencer should be routed in a similar way: Button 1 - Pattern Enable - Min: 0, Max: 1.

The three filters should each be routed as follows: Button 1 - Enabled - Min: 2, Max: 1. Meaning that the button which toggles the Step sequencer on and off will also toggle the three filters between On and Bypass mode.

The Chorus: Button 3 - Enabled - Min: 2, Max: 1.

The Phaser: Button 4 - Enabled - Min 2, Max: 1.

The finished Combinator should look like this: multi_ddl-1.cmb. Now let's see some examples of how to use it.

ddl_stepdemo1.rns - A simple REX drum loop showing off the crazy delay effects you can achieve from the step time automation. The delay is bouncing around all over the place, at times sounding like vinyl scratching. The patterns controlling the delay times are entirely randomized.

ddl_stepdemo2.rns - This one uses programmed patterns that are a little saner (but only a little). It also uses different filter settings for the delays, which makes for some interesting variation.

ddl_filterdemo.rns - The filter sequencer in action, here shown gradually filtering out the high end over the course of two bars.

ddl_pandemo.rns - Here we have simply re-routed some CV signals - by going into the Combinator and changing the target parameter from DelayTime (steps) to Pan, we now have a random(ish) panning for each delay. Chorus and Phaser added for extra sweetening.

That's all for this issue of Discovering. In part II of the article we will continue with more customized rides, including one entitled "SuperSub". Have fun!

Text & Music by Fredrik Hägglund

The Hitchhiker's Guide to the Combinator - Part II

As an aspiring Combinator aficionado you will proba-
bly go through these two phases:

As an aspiring Combinator aficionado you will probably go through these two phases:

Phase 1: Awe. "*Woah, I'm gonna build like the biggest, baddest, meanest synth evverr!*".

Phase 2: Sanity. "*OK, that was fun... but now I want to create something truly useful*".

Combinators don't have to be gargantuan, complex CPU hogs - they can be lean and ef-
ficient too. We're going to continue with the theme we started in part one of this article
and build a few more "replacement" Combinators; the term 'replacement' signifying that
these Combinators can do everything that the original devices do, and then some - so
why not let them replace the standard devices in your compositions?

The SuperSub

Here's a Combinator based on dual Subtractors. We'll call it SuperSub in honor of the
legendary Roland Super JX, which was essentially two JX-8P synths in one machine. (In-
cidentally, the Super JX was one of Douglas Adams' favorite synths, which ties in with
the title of this article.). The JX-8P was perhaps not the most exciting of synths, as it had
lower quality oscillators than predecessors such as the Juno-60, but once they stacked
two of them together inside the JX-10 it transformed into a whole other animal.

The "Discovering twist" here is that the Combinator allows you to control the waveform
selection on the Subtractors with the rotary controls, and the rotary controls can be con-
trolled by CV. This means you can change waveforms using for example a Matrix se-
quencer, allowing you to build rhythmic Subtractor textures by sequencing waveform
changes. To top it off, we'll also add some instant fattening functionality: Unison, macro
detune and stereo spread.

The SuperSub recipe:

- 1 Line Mixer 6:2
- 2 Subtractors
- 1 UN-16 Unison
- 2 Matrix sequencers
- 1 Spider CV

Audio routings

The line outs from the two Subtractors connect to channels 1 and 2 on the Line Mixer. The UN-16 Unison will be used as a send effect (not an insert as one might think), so we'll just connect it to the Aux sends and returns on the Line Mixer.

CV/Gate routings

We've got a Spider CV here and we'll be using it to split the the Note and Gate CV coming from the Matrix unit responsible for sequencing the notes. Connect Note CV on the Matrix to Split A - In, and Gate CV to Split B - In on the Spider CV. Then connect the Split A and B outputs to the Note and Gate CV inputs on the two Subtractors. Finally there's the second Matrix which controls the Waveform selection on the Subtractors: Connect Curve CV to Rotary 3 on the Combinator. If you want to change waveforms on one of the Subtractors only, simply unplug this cable or disable its routing in the programmer.

Rotaries

Rotaries 1 and 2 will be controlling the Filter Frequency and Resonance parameters on both Subtractors simultaneously. Rotary 3 is for selecting waveforms - this will normally be done by the Matrix Sequencer, but it can be done manually too of course. Rotary 4 detunes the oscillators using different scales - detuning Osc 1 up and Osc 2 down, spreading evenly so that you'll get max detune values of -25 and +25 on Subtractor A and -50/+50 on Subtractor B.

Buttons

Button 1 toggles the mode on the UN-16 Unison between On and Off (not Bypass, as this would amplify the signal). Button 2 will pan the first Subtractor full left and the second Subtractor full right - this is handy if you're using the same patch on both Subtractors to create a stereo sound - with the detuning, the Unison and stereo spread you can get a really wide sound quickly. Button 3 switches the Wave sequencer On/Off, and Button 4 toggles the Note sequencer On/Off.

Device: 6	SuperSub A		
Source:	Target:	Min:	Max:
Rotary 1	Filter Freq	0	127
Rotary 2	Filter Res	0	127
Rotary 3	Osc1 Wave	0	31
Rotary 4	Osc1 Fine Tune	0	25
Button 1			
Button 2			
Button 3			
Button 4			
Rotary 3	Osc2 Wave	0	31
Rotary 4	Osc2 Fine Tune	0	-25

Combinator Programming

Most of the programming here deals with the two Subtractors. Rotaries 1, 2 and 3 should be programmed identically for both synths. Rotary 1 controls Filter Freq, Rotary 2 controls Resonance - pretty straightforward stuff.

Rotary 3 is controlled by a Matrix unit and its job is basically to convert Curve CV data to Waveform selection. It should therefore be assigned to the parameters "Osc1 Wave" and "Osc 2 Wave" with the values Min 0 / Max 31 (there are 32 waveforms in total).

Rotary 4 will control the parameters "Osc1 Fine Tune" and "Osc2 Fine Tune".

We'll use four different sets of values here in order to detune each of the four oscillators differently. Subtractor A - Osc1 Max 25, Osc2 Max -25. Subtractor B - Osc1 Max 50, Osc2 Max -50. The Min value should be 0 for all four. That takes care of the Subtractors, but there are a few items left - the buttons.

In the Unison modulation routing panel, create a simple On/Off function by routing Button 1 to "Enabled", Min 0 / Max 1.

For the 6:2 Mixer, there should be two entries for Button 2: "Channel 1 Pan" Min 0 / -64 and "Channel 2 Pan" Min 0 / Max 63. This is the "spread" function which is obtained by simply panning the two Subs hard left and hard right. For the Matrix that selects waveforms, route Button 3 to "Pattern Enable", Min 0 / Max 1.

Finally, on the note sequencing Matrix unit: Button 4, "Pattern Enable", Min 0 / Max 1.

This is the SuperSub, skinned, sealed & delivered: subersub.cmb. Now let's see what it

can do.

moving_mongolia.rns - We loaded the factory patches "Outer Mongolia" and "Cloud Chamber" into the Subtractors and enabled all the SuperSub extras; heavy detuning, unison and spread. On the first 8 bars, the Waveform Sequencer is disabled. It kicks in at bar 9, after which you can hear a major difference in the way the texture moves and evolves. Exercise: Try loading other pad sounds from the factory bank into the Subtractors. The waveform sequencer will keep going. You can stumble upon great sounds by simply doing this.

80s_super_dance_lead.rns - a heavily detuned, old school techno lead patch. Exercise: Remove all the extras progressively. Switch off the spread and the unison, and set detuning to zero. Finally, Mute one of the Subtractors. Notice what a day-and-night difference a little layering and FX magic can make.

annoying_man.rns - a buzzy, glitchy sound with a dash of distortion. The waveform sequencer is running a randomized pattern. Exercise: With the sequencer playing, open the curve display on the "Wave Seq" Matrix and experiment with drawing different curves by just click-dragging arbitrarily across the display. Endless variations there.

squaretooth_bass.rns - this is a bass sound that alternates between sawtooth and square wave on every 1/8 note, to showcase a slightly more organized way of using the waveform sequencer. Exercise: Experiment with the setting on the 3rd Combinator knob "Waveform 1-32" while the pattern is playing. Notice how the waveforms change, but the sequencer keeps alternating between two adjacent waveforms.

ReDrum D / A / G

ReDrum D, A and G are three different Combinators based around Matrix sequenced ReDrum units. D is for Dual, A for Analog and G for Graintable. ReDrum A emulates an analog drum machine, with one Subtractor for each ReDrum channel. ReDrum G is similar, but uses Malströms instead of Subtractors. We will go through the D model in detail; the other two are bonus patches that you can download and explore on your own.

Playing ReDrum with Matrix sequencers is great because you get one independent pattern sequencer for each drum channel, meaning you can have individual pattern length and time resolution for each drum. You also get a finer velocity resolution (0-127) than with the internal ReDrum sequencer (0-3) - and last but not least, it's much quicker to draw Matrix patterns than to enter them in the ReDrum sequencer.

You can have one Matrix unit for each of the ten ReDrum channels, of course, but this is likely overkill. It's more practical to use the internal ReDrum sequencer for those sounds which have little or no variation, while using separate Matrix sequencers for the snare and bass drum, for example. In our example Combinators we've put three Matrix sequencers (for bass drum, snare drum and closed hi-hat), but you can add more if you like.

The ReDrum D (Dual) uses two ReDrum units that are linked via Gate, so that channel 1 on ReDrum B is triggered by channel 1 on ReDrum A. This allows you not only to layer two samples per channel, but also to make velocity gradients between the two samples.

The ReDrum D shopping list:

- 1 Mixer 14:2
- 2 Spider Audio
- 2 Redrum
- 3-10 Matrix Sequencers

Audio routings

Since we're dealing with 10+10 ReDrum channels here, we ought to need 20 mixer channels. Of course we could have two 14:2 mixers, or add a 6:2 mixer and get a total of 20, but we want to keep this one lean. So let's see. It makes sense to merge ReDrum channels 8 and 9 into one mixer channel, as they're intended for closed and open hihat. Then we could also merge channels 3-5, the tom channels, into one. That leaves a total of 7 outputs per ReDrum unit, 14 channels, precisely what we have.

Now, the routing isn't very complicated but it's a loooot of cables back there so we need some guidance through the jungle.

First, channel 1 (BD) of the "Master" ReDrum goes to mixer channel 1. Channel 1 of the "Slave" ReDrum goes to mixer channel 2 (it's more practical to have these 'pairs' next to each other on the mixer). Then there's channels 3-5, which we were going to merge. Since the Main Output on a ReDrum gets all the leftover channels that aren't routed via individual outputs, and all ReDrum channels except 3-5 will indeed be routed via individual outputs, this means ReDrum is already merging channels 3-5 for us on the Main output.

So, connect Main Out on the Master ReDrum to mixer channel 3, and Main Out on the Slave ReDrum to channel 4. Now onto ReDrum channels 6 and 7. These will be routed to the mixer as follows: ReDrum Master 6 to Mixer 7. ReDrum Slave 6 to Mixer 8. ReDrum Master 7 to Mixer 9. ReDrum Slave 7 to Mixer 10. Now for the hihat channels 8+9. These will need to be merged with Spider Audio. Take channels 8+9 on the Master ReDrum and route them to Spider Audio #1. Do the same for the Slave ReDrum and Spider Audio #2. Then take the merged output from Spider #1 and route it to mixer channel 11, and from Spider #2 to mixer channel 12. Finally, connect Master ReDrum channel 10 to Mixer channel 13 and Slave ReDrum channel 10 to Mixer channel 14. Phew. What do we have on the mixer now?

Channels 1-2: Bass Drum A+B.
Channels 3-4: Snare Drum A+B.
Channels 5-6: Lo/Mid/Hi Toms A+B.
Channels 7-8: Perc 1 A+B.
Channels 9-10: Perc 2 A+B.
Channels 11-12: Hihat Open/Closed A+B.

Channels 13-14: Cymbal A+B.

CV/Gate routings

Two issues to deal with here: 1) The Master+Slave routing between the two ReDrums, 2) Matrix routing. First, you need to connect Gate Out on each of the 10 channels on the Master ReDrum to Gate In on each of the channels on the Slave ReDrum. Many cables, but easy. Then there's the Matrix routing. As mentioned earlier, we only have 3 Matrix units in this example, but you can fill up with more if you need'em. Either way, the routing is simple: Just connect Gate Out on the Matrix to Gate In on the Master ReDrum channel you want to control.

Rotaries/Buttons

Simple stuff. The Combinator panel buttons 1-4 and rotaries 1-4 will act as four pairs, with the button starting and stopping a sequencer and the rotary selecting patterns on that sequencer.

Combinator programming

Not much! Each Rotary will be assigned to "Pattern Select", Min 0 / Max 31. Each Button will be assigned to "Pattern Enable", Min 0 / Max 1. You will need to enter this four times, for the Master ReDrum (button 4/rotary 4) and the three Matrix sequencers (buttons 1-3/rotary 1-3).

That's all there is to it.

The finished ReDrum D Combinator: redrumd.cmb

And, as promised, ReDrum A (Analog): redruma.cmb and ReDrum G (Graintable): redrumg.cmb. Play around with these if you ever get tired of sampled drums! And enjoy

what little is left of the summer!

Text & Music by Fredrik Hägglund

One Hand in the Mix - Building Crossfaders using the Combinator

 With the invention of Propellerhead Software's Remote technology, Reason is, more than ever, a powerful live performance tool, especially when used in conjunction with a control surface. One of the basic elements for many live applications is the ability to segue between two audio sources in real-time using a crossfader. The principle is simple where the level of two faders simultaneously change when receiving a single controller event. The fader levels are inversely proportional so as one level increases, the other decreases. This allows convenient blending or transitions between audio sources without having to use two controllers or separate mouse parameter changes. This installment of Discovering Reason describes two methods of creating crossfaders using the Combinator in Reason 3.0, and when used with an external control surface, present a powerful tool for live performance.

The principle of a crossfader is very straightforward, and most Reason users can probably already figure out how to create a crossfader patch using the Combinator and the Micro Mix Line Mixer. For those who have yet to explore this type of control, the following project is adapted from Chapter 5 of Power Tools for Reason 3.0, which describes using multiple modulation routings from a single Combinator Rotary control.

Basic Crossfader

In Chapter 4 of Power Tools for Reason 3.0, an example demonstrated a technique of using a CV signal and an inverted CV signal to modulate two fader levels simultaneously in order to create a crossfader. A more elegant solution to this effect uses the multiple modulation routing features of the Combinator.

1. In an empty rack, create a reMix mixer.
2. Create a Combinator.
3. In the Combinator sub-rack, create a microMix Line mixer.
4. Click on the Show Programmer button, and click on Line Mixer 1 in the device list.
5. In the modulation routing list, set the rotary 1 target to Channel 1 Level. Set the rotary 1 Min to 127 and Max to 0.
6. Beneath the button 4 source field, set the empty source field to Rotary 1. Set the target to Channel 2 Level, in 0,Max 127.

7. Double-click on the rotary 1 label and change the name to "X-Fade."
8. Click on the disk icon in the Combinator and save the patch as "Basic Crossfader.cmb."

This configuration fades between the microMix channel 1 and 2 inputs, and now that it is saved as a patch, it can be recalled anytime in the future for use in other projects. The next section illustrates how to implement the crossfader with sound sources.

1. Click on the empty rack area beneath the Combinator.
2. Bypass auto-routing (Hold the Shift Key and select an item from the create menu) and create a Dr.REX Loop Player.
3. On the Dr.REX, load the loop, "Hhp11_Chronic_093_Chrnc.rx2" from the Factory Sound Bank\Dr Rex Drum Loops\Hip Hop directory.
4. Copy the REX data to the Dr.REX 1 sequencer track.
5. Connect the Dr.REX 1 audio outputs to the microMix channel 1 inputs.
6. Click on the empty rack area beneath the Dr.REX.
7. Bypass auto-routing and create another Dr.REX Loop Player.
8. On the second Dr.REX, load the ReCycle file "Hhp18_Furious_093_Chrnc.rx2" from the Factory Sound Bank\Dr Rex Drum Loops\Hip Hop directory.
9. Copy the REX data to the Dr.REX 2 sequencer track.
10. Connect the Dr.REX 2 audio outputs to the MicroMix Channel 2 inputs.

Example File: Basic Crossfader.rns

Cables from outside of the Combinator can connect directly into devices in the sub-rack, but these connections are not saved with the patches. The Combinator will indicate "external routing" when such connections are present. The two Dr.REX loops provide two different sound sources. Run the sequence to play the loops, then adjust the Combinator's X-Fade rotary control to crossfade between the two loops.
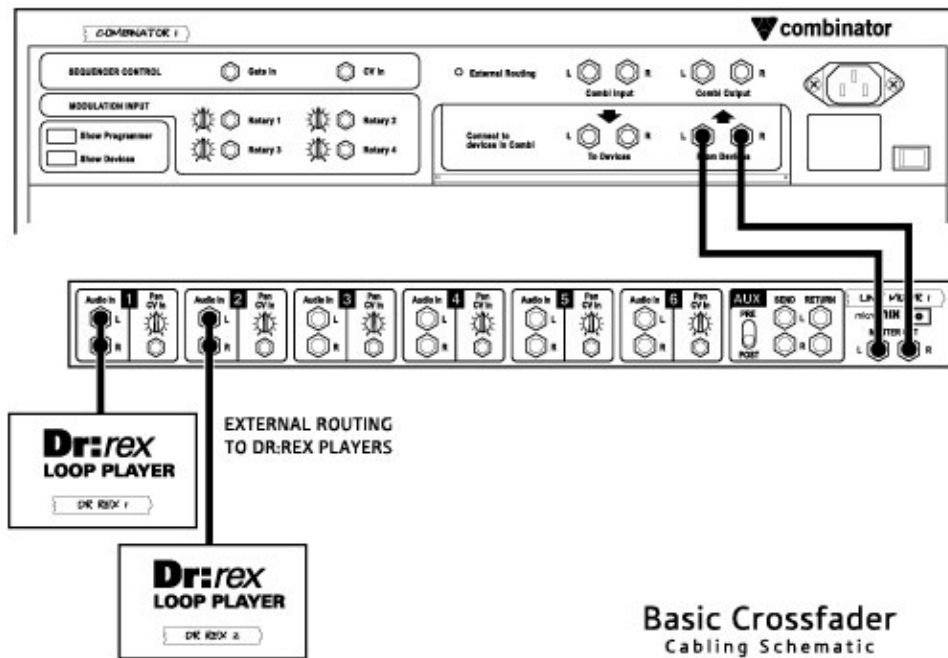
Figure 1. *The basic crossfader only uses a single Micro Mix Line Mixer. The sound sources are two Dr.Rex players externally routed into the Line Mixer inputs. The Rex Players are not included in the Combinator rack.*

There is an inherent problem with this configuration, heard as you fade between the two sources. As you run the sequence, slowly turn the X-Fade Rotary from left to right and back. As the control nears the mid point, the output level dips down. The linear fading characteristic of this configuration causes this decrease in level. The effect of the dip can be reduced by decreasing the range of modulation, but cannot effectively be overcome for this configuration. While this works fine for most applications, for live performance applications a more desirable effect is an equal-power crossfade.

Equal Power Crossfader

The attenuation characteristics of both the ReMix and MicroMix mixer pan pots is not linear like the crossfade example above, instead the pan pots have a scaled attenuation rate called constant-power or equal-power. The output level of the two panned signals is constant between the left and right channels. The scaled attenuation of the pan pots can be used to create an Equal Power Crossfader.

Equal Power Crossfader
Combinator Cabling Schematic
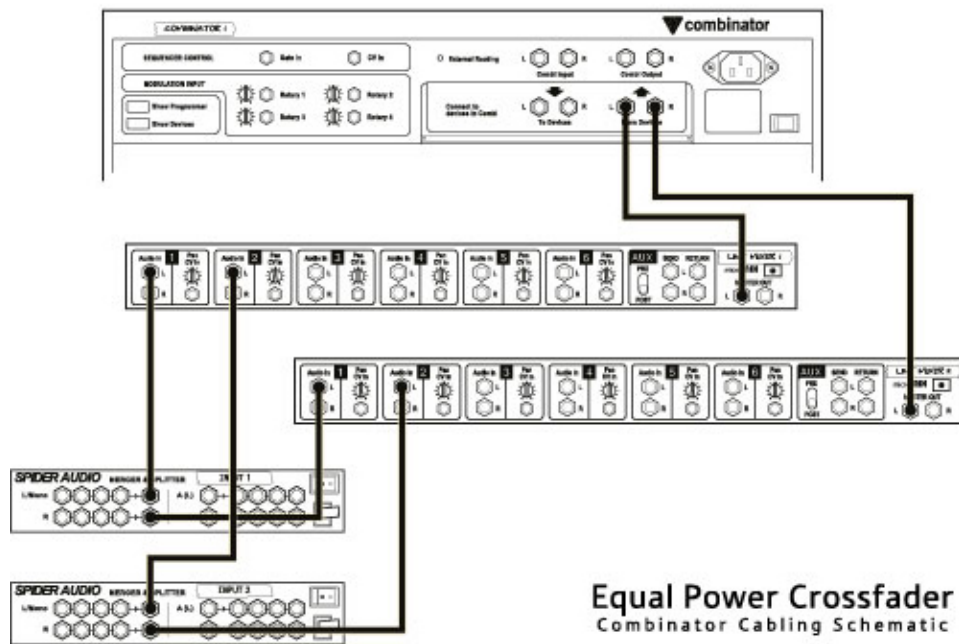
Figure 2. *Two Line Mixers are used for the Equal Power Crossfader patch. Each Mixer handles a single channel, so two are required for a stereo crossfader. The Spider Audio Merger/Splitters are used as patchbays so that external stereo connections automatically route to the proper input channels.*

The following is referenced on the Power Tools for Reason 3.0 CD-ROM in the Combinator patches folder. For the readers of Discovering Reason, the project illustrates how to create this effect on your own. An example Reason song file is included at the end of the project.

Equal Power Crossfader Combinator Patch

1. In an empty rack, create a reMix mixer.
2. Create a Combinator, and verify that it is cabled to the remix Channel 1 inputs.
3. In the Combinator sub-rack, bypass auto-routing (hold down the shift key) and create two microMix Line mixers.
4. Bypass auto-routing and connect the Line Mixer 1 Left output to the Combinator 'From Devices' Left input.
5. Connect the Line Mixer 2 Left output to the Combinator 'From Devices' Right input.
6. Create a Spider Audio Merger/Splitter, and Rename it to 'Input 1'
7. Bypass Auto-routing and connect the 'Input 1' Merger Output Left to the Line Mix-

er 1 Channel 1 Left input.

8. Connect the 'Input 1' Merger Output Right to the Line Mixer 2 Channel 1 Left input.
9. Create another Spider Audio Merger/Splitter, and Rename it to 'Input 2'.
10. Bypass Auto-routing and connect the 'Input 2' Merger Output Left to the Line Mixer 1 Channel 2 Left input.
11. Connect the 'Input 2' Merger Output Right to the Line Mixer 2 Channel 2 Left input.
12. Click on the Show Programmer button, and program the following modulation routings. For the second Rotary 1 source, use the Assignable Source set to Rotary 1:

Device	Source	Target	Min	Max
Line Mixer 1	Rotary 1	Channel 1 Pan	-64	63
Line Mixer 1	Rotary 1	Channel 2 Pan	63	-64
Line Mixer 2	Rotary 1	Channel 1 Pan	-64	63
Line Mixer 2	Rotary 1	Channel 2 Pan	63	-64

13. Double-click on the rotary 1 label and change the name to "X-Fade."
14. Click on the disk icon in the Combinator, and save the patch as "Equal Power Crossfader.cmb."

The Spider Audio Mergers are not a necessary part of the effect, but they provide an easy way to connect sources into the crossfader. The merger inputs take advantage of the auto-routing rules so that stereo sources are properly cabled in a single connection. One of these can be routed to the combinator's inputs instead. Since four inputs are available, multiple input sources can be configured to mix through a single crossfader.

Sound Source

15. Click on the empty rack area beneath the Combinator.
16. Bypass auto-routing (Hold the Shift Key and select an item from the create menu) and create a Dr.REX Loop Player.
17. On the Dr.REX, load the loop, "Hhp11_Chronic_093_Chrnc.rx2" from the Factory Sound Bank\Dr Rex Drum Loops\Hip Hop directory.
18. Copy the REX data to the Dr.REX 1 sequencer track.
19. Connect the Dr.REX 1 audio outputs to 'Input 1' Merge Input 1 inputs.
20. Bypass auto-routing and create another Dr.REX Loop Player.
21. On the second Dr.REX, load the ReCycle file "Hhp18_Furious_093_Chrnc.rx2" from the Factory Sound Bank\Dr Rex Drum Loops\Hip Hop directory.

22. Copy the REX data to the Dr.REX 2 sequencer track.
23. Connect the Dr.REX 2 audio outputs to the 'Input 2' Merge Input 1 inputs.

Run the sequence and adjust the X-Fade (Rotary 1) on the Combinator.

Example File: Equal Power Xfade.rns

The louder crossfade action is noticeable immediately in the Equal Power Xfade example. The mid point between the two sources does not diminish as much as the basic crossfader example making it better suited for applications like mixing loops where even dynamics are desired during the transition.

The example song file features extra controls for input level trims to balance the incoming signals as well as a master level control. A few other features you might wish to add are kill switches on the button controls that modulate input channel muting. All of this can be assigned to a Remote control surface for use in a live performance or other real-time mixing applications, and of course, the crossfades can be recorded and edited on the Combinator 1 sequencer track.

Wet/Dry Balance Crossfader

Crossfader configurations effect can also be implemented as a Wet/Dry Balance control as well for real-time switching between a direct signal and a processed signal. Again, this can be especially useful for drum loops in a live situation where you want to create the energy of a performance rather than a preprogrammed sequence.

The following example file demonstrates using a ReCycle Loop playing from a Dr.REX Loop player being processed with a custom delay effect based on the "Beat Juggler" Project in Power Tools for Reason 3.0. The Loop audio is split into parallel signals using a Spider Audio Splitter. The dry signal passes through the splitter into the Equal Power Crossfader Input 1, and a split signal, processed through the delay effect, is connected to the Crossfader Input 2. The demonstration sequence is recorded in real-time with various control parameters for the crossfader, delay time, and delay feedback being controlled by a Remote Control surface.

Example File: Equal Power Xfade DelayFX.rns

A single fader is used to mix between the direct loop level and the delayed loop level via the crossfader, liberating a hand to manipulate the delay and feedback effects. Obvi-

ously this quite useful for live performances, but it's also an exciting way to create tracks in the studio environment as well. You can quickly create rhythmic variation on the fly rather than tediously programming REX slices. This type of effect crossfader can be implemented with any variety of effects like the Scream 4 distortion, the ECF-42 Filter, or the Reverb units for even more variations.

Text & Music by Kurt Kurasaki

The "Basic Crossfader" example in this issue of *Discovering Reason* can be found in **Power Tools for Reason 3.0** by Kurt "Peff" Kurasaki (although the REX Files used here are created especially for www.propellerheads.se).

The "Equal Power Crossfader" example is completely new, written exclusively for www.propellerheads.se, although the combi patch is included on the *Power Tools for Reason 3.0* CD-Rom. Both schematic illustrations were created exclusively for this article.
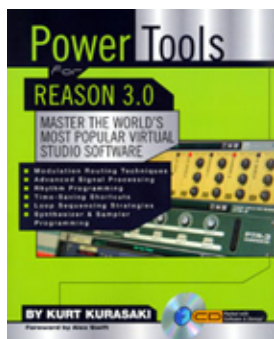
Power Tools for Reason 3.0 by Kurt "Peff" Kurasaki is written for experienced Reason users. It is not recommended for the complete beginner! This work covers various audio engineering and music production, including control voltage routing, audio effects, Combinator patches, rhythm programming, and sound design principles. Each section includes projects with step-by-step directions and schematics that clearly illustration how you can apply these principles. The book includes techniques that explain the new Mastering effects, as well as revised Synthesizer and Sampler Programming chapters.

Book Website: http://www.pt-reason.com
Author's Website: http://www.peff.com
Publisher's Website: Backbeat Books

Let's RPG-8!

DISCOVERING REASON With Reason version 4 came the RPG-8 Monophonic Arpeggiator. As most of you may already know, an arpeggiator can be used to generate rhythmic monophonic melody lines out of input notes or chords. This is exactly what the RPG-8 does - and much more if you dig a little under the surface. In this article we're going to have a look at how you can record and edit individual arpeggiated notes in the main sequencer to manually "fine tune" your arpeggio lines. We'll also show some other cool stuff you could do with the RPG-8 - things you maybe wouldn't consider using the arpeggiator for at first thought.

Rendering Individual MIDI Notes From an Arpeggio

A very nice feature with the RPG-8 is that it allows you to render the arpeggiated notes as individual MIDI notes. These notes can then be edited and treated just like any other recorded MIDI notes in the sequencer. Rendering individual MIDI notes from an arpeggio is made in two steps. First you have to record the notes that will generate the arpeggio, i.e. single notes and/or chords. Then, you can render separate MIDI notes from the recorded arpeggio. The following example shows how to go about it:

We have set MIDI input to the "Arp" track in the sequencer and connected the RPG-8 to an NN-19 device loaded with a percussive patch tuned to a fifth. We record eight bars with the RPG-8 in Manual Mode and 3 Octaves range.

RPG8example1.rns |

Now, we decide to switch Mode in the middle of the sequence to change the direction of the arpeggio pattern. We record a change of the Mode parameter from Manual to Down starting at bar 5.

RPG8example2.rns |

Now, we're

happy with the result so far and are ready to enter step 2 - the actual rendering of individual MIDI notes from the arpeggiated notes. We make sure the left and right locators are set to span eight bars to cover the entire length of the original recording. Then, we change from the "Arp" track to the "Arp Sound" track in the sequencer. We select the RPG-8 device in the rack and use the Edit menu, or bring up the device context menu from which we select "Arpeggio Notes to Track". Now, notes will be created on the target device track between the left and right locators.

Here comes an important thing to remember. At this point, we have two arpeggios, one coming from the "Arp" track, created by the RPG-8 and one being played as notes from the "Arp Sound" track. Before we proceed we therefore need to mute the "Arp" track. We will still keep the "Arp" track in case we change our minds later on and want to use other arpeggio patterns, ranges or directions for example.
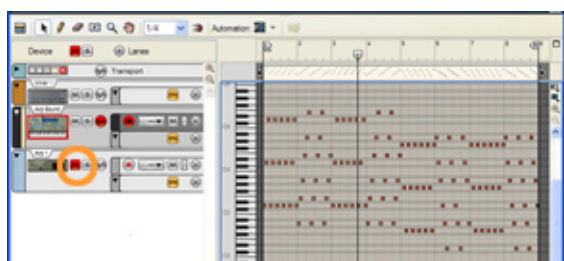
We decide to modify some of the rendered notes to change the arpeggio "melody" a bit. We enter Edit Mode on the "Arp Sound" track and re-arrange some of the notes.
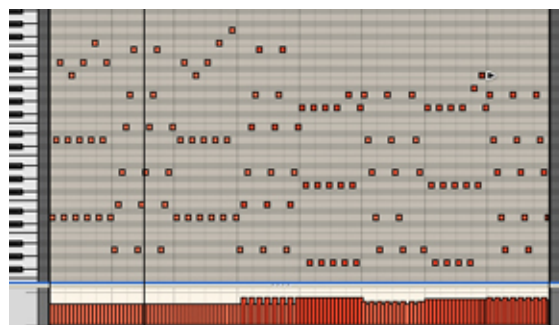
RPG8example3.rns |

Finally, to put the arpeggio line in a context, we add on a pad and a bass sound on two additional sequencer tracks.

RPG8example4.rns |

Filter Modulation

Besides using the RPG-8 for generating melody lines, we could also use it as modulation source for modulating various device parameters. The following example shows how the RPG-8 can be used for modulating the filter frequencies of a Subtractor synth.

Here, we're using an RPG-8 connected to the Subtractor via a Spider CV device. The Spider device is necessary because we need several CV signal outputs when we want to control the filters. We create the Spider CV device by selecting the RPG-8 in the rack and choosing Create -> Spider CV from the menu. The Spider CV device appears and is automatically connected to the RPG-8. Then, we select the "Arp" track on the sequencer and record a couple of simple chords.
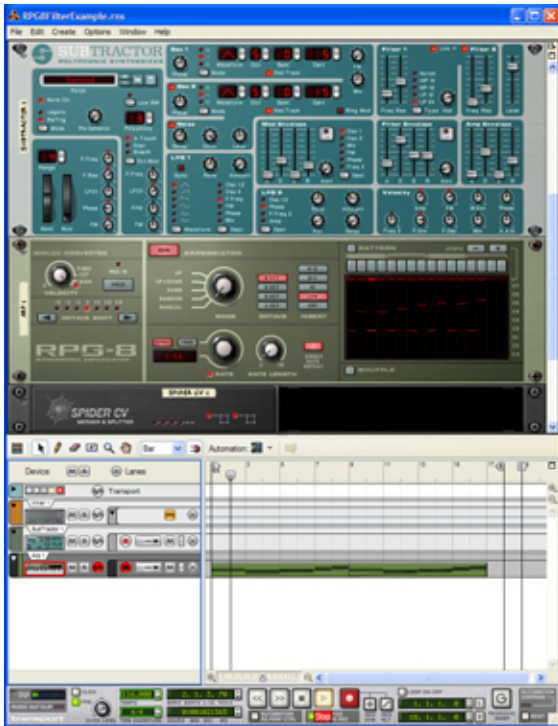
Now, we want to modulate the Subtractor Filter frequencies from the Note CV signal of the RPG-8. We flip the rack around to make some additional cable connections.

Since the Note CV Out of the RPG-8 is connected to the Split A input of the Spider CV device, we connect cables from two Split A outputs to the Filter 1 Freq and Filter 2 Freq inputs of the Subtractor device. Then, we crank up the modulation knobs next to the inputs.

Don't let the word Note CV limit your experimentation - it only implies that the output control signal level depends on current note value. The Note CV output signal can be used for controlling almost any type of parameter in Reason - not only notes and oscillator pitches.

Now, when we play back the sequence, the two Subtractor filters will open up according to the output Note CV signal from the RPG-8. The higher the Note CV out value, the higher the filter cutoff frequencies. The effect will be more significant if you choose a wide Octave range in the RPG-8 - in this example we've chosen 4 octaves range.

RPG8FilterExample.rns |

Sample Modulation

Another fun application is to use the RPG-8 for modulating sample start position. In this example we have connected an RPG-8 to an NN-19 Sampler via a Combinator device. In the NN-19 we have loaded a vocal sample which changes characteristics over time. By modulating the start position of the sample from the RPG-8 we can get really interesting rhythmic effects.

In the Combinator we have assigned Rotary 1 to the Sample Start parameter in the NN-19 Oscillator section.
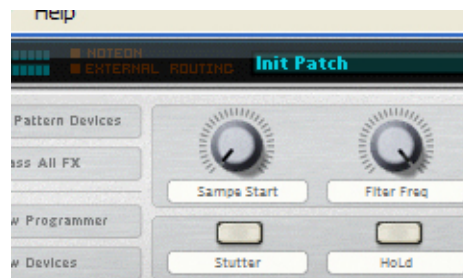
If we flip the rack around, you can see that we have connected the RPG-8 Gate CV Out to the NN-19 Gate input to force it to continuously re-trig the sample. We have also connected the RPG-8 Note CV Out to the Combinator Rotary 1 modulation input and turned the modulation knob. This way we will get the Note CV signal from The RPG-8 to change the sample start position for every step of the arpeggio.

The original vocal sample sounds like this:

RPG8SampleStartModulation.rns |

Like in the filter modulation example above, the modulation effect will be more pronounced if we choose a wide Octave range in the RPG-8.

Text & Music by Fredrik Hylvander
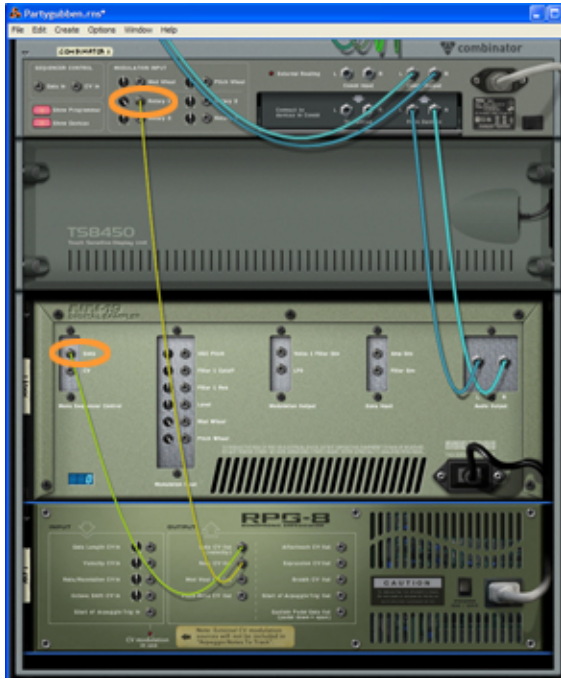
Making friends with clips

DISCOVERING REASON The main sequencer in Reason 4 has undergone some serious improvement since version 3. One improvement is the introduction of so called clips. In this article, we'll try to describe the philosophy behind clips and give an example of a basic music production process using the sequencer in Reason 4.

What's a clip?

A clip is essentially a take. Everything you record in the Reason 4 sequencer will end up in a clip. A clip can contain different types of information; note events, device parameter automation, pattern automation, tempo automation etc. Personally, I like to picture a clip as a dynamic piece of transparent tape containing a short piece of music information. A selected clip in the Reason 4 sequencer is indicated by a black frame with handles on either side:
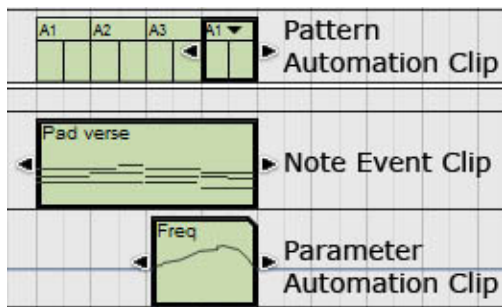
When recording, a note clip will always snap to the closest bar to make it easy to arrange afterwards. A clip is not fixed but can be modified in various ways; extended, shortened, joined with other clips etc. etc. Everything in a clip can also easily be modified by entering Edit Mode from the sequencer toolbar or by simply double-clicking the clip.

Why clips?

The reasons for choosing the clips concept are many but first and foremost to make for creativity and ease when creating and arranging music. Everything you record in the Reason 4 sequencer ends up in clips. This way you can rest assured that no information or data will be lost by mistake during editing and arrangement.

Some introductory clips tips

Too many clips on the lane?

Sometimes, if you do several overdubs on the same region of a lane, you could end up with a lot of small clips which eventually could be difficult to overview and to manage.

If this happens, just select all clips in that region of the lane with the Arrow tool and choose Join from the Edit menu (or Ctrl [PC]/Option [Mac]+J). Now, everything will end up in just a single, easily manageable, clip.

Drawing notes in clips

If you want to draw note events using the Pencil tool it's important to remember that you have to do this in a clip. If there is no clip present in the lane you have to create one first. The nice thing is that you create a new clip with the Pencil tool as well. So, in Edit Mode, select the Pencil tool and draw the empty clip to cover the number of bars you like. Then, right after and without switching tool, begin to draw your notes in the clip.
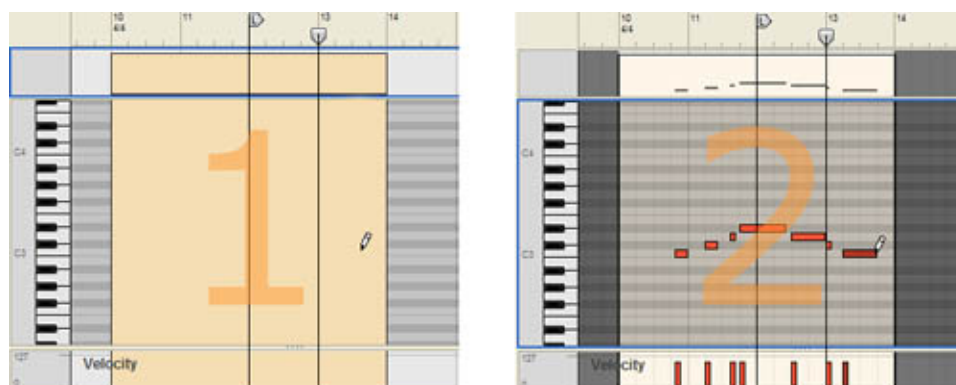
Joining masked clips

If you decide to join clips that contain masked notes, be aware that any masked notes in the "joint section" will be deleted. The reason for this is that it's not possible to mask notes in the middle of a clip. And if you join two clips, the "joint section" will be in the middle of the new clip. However, a rule of thumb is that what you hear before joining clips is exactly what you're going to hear after joining.

The recording process

The following is a basic example of how a recording process could evolve using the Reason 4 sequencer. Let's start from scratch with a new document.

Drums

First we create a Redrum device. We record a couple of patterns in the internal Redrum pattern sequencer. Then, we click the Create Pattern Lane button on the main sequencer toolbar, hit Rec and record some of the Redrum patterns into the main sequencer.
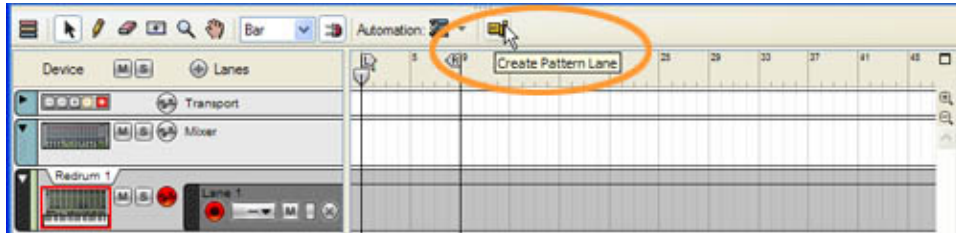
By clicking the rightmost clip handle and dragging to the right we can now expand the patterns to cover as many bars as we like. By clicking the small pull-down triangle next to the pattern name in the clip we can easily change to another pattern without re-recording. We can also move the clips if we like.

We add some more bars manually with the Pencil tool until we have filled up 8 bars. Finally, we select Pattern A1 for the last two bars by clicking the pull-down triangle next to the pattern name.

Bass

Next, we create a bass instrument and record a 2-bar bass line. We resize the clip to exactly two bars with the rightmost handle before we copy and paste the clip to fill up all 8 bars. When you copy and paste a clip, the copy automatically ends up after the last clip on the lane - you don't have to manually define any insertion point.

Now, we want to mute the notes of bar 4 to get some variation in our bass line. We se-

lect the second clip and place the cursor on the right handle and drag to the left across the last notes. The notes will still be there, only you won't hear them since they are now masked. If we change your minds later on we'll just expand the clip to make the masked notes sound again.

If you know you're never going to use these notes again you could permanently delete them by selecting the clip and choosing Crop Events to Clips from the Edit menu. This command will delete all masked events in a clip.

Melody

Now, let's move on to the melody line. We create a new instrument and choose a nice piano-ish sound. We record in a single clip throughout the entire 8 bars and then stop the sequencer. We're not entirely happy with the result but most parts of the melody line sounds OK so we'll keep it. We want to record an alternative take to see if we could nail it this time. We click the New Alt button on the transport bar to automatically create a new lane and mute the previous lane.

We record the second take on the new lane and stop the sequencer. Now, we want to record an alternative ending of the verse. Again we click the New Alt button to create a third lane and mute the other two. We place the ruler at bar 7 and record the last two bars.

Now, we have three clips on three separate lanes for the melody. We decide to make one single melody clip out of the three separate ones - i.e. take the "goodies" from the three clips and merge into a single clip.

We use the Razor tool to cut up the three clips in smaller "good" and "bad" clips. Then we mute the bad clips using the Mute Clips command on the Edit menu (or by pressing

M on the computer keyboard) to verify that the melody sounds the way we want. We also click the M buttons to un-mute lanes 1 and 2 to be able to hear the clips on all three lanes.

The good parts turned out to be bar 1, 2, 5 and 6 on lane 1, bar 3 and 4 on lane 2 and bar 7 and 8 on lane 3. Now, let's merge all the clips on the three note lanes to one single lane. Select Merge Note Lanes on Tracks from the Edit menu and the clips end up on a single lane. The muted clips are automatically deleted after the merging.

Finally, we want to join the separate clips on the lane to have the entire verse in just one clip. We select all clips on the lane and choose Join Clips from the Edit menu

Pad with parameter automation

We now have the backbone of our song; the drums, bass and melody. Let's add a pad so the melody has something to "float" on. We create a new instrument and choose an "airy" pad sound. We record the pad chords in a single 8-bar clip. In bar 3 we accidentally hit the wrong notes. We enter Edit Mode by selecting the clip and pressing the Enter key (new in V4.0.1). We correct the wrong notes and exit to Arrange Mode by pressing the Esc key (new in V4.0.1).

Let's introduce a slow filter sweep to the chord in bar 5. We place the song position marker at bar 5 and hit Rec. A new clip is automatically created on the note lane since the sequencer doesn't know what data we're going to record yet. When we start changing the Filter Frequency parameter on the SubTractor device, the parameter automatically ends up in a new clip on a separate parameter lane and we can see the value changes appear as a grey line in the new clip.
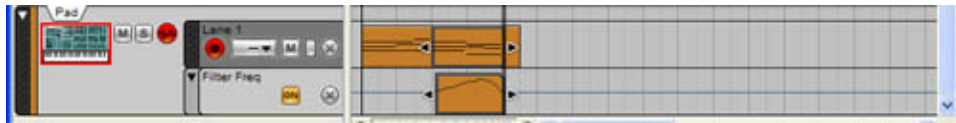
As soon as we record a parameter change in the sequencer, the parameter has been auto-mated. By automated we mean that this particular parameter will always have defined values throughout the entire song - also before and after the actual clip.

Let's take a closer look at the Filter Frequency clip by double-clicking it. We can see that in the beginning and end of the clip, dashed lines expand on either side. These dashed lines indicate the first and last values of the Filter Frequency parameter in the clip.
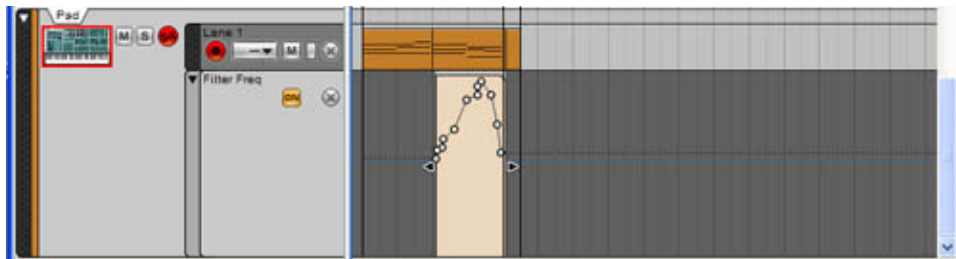
The first Filter Frequency parameter value in the clip is called the static value. The static value, indicated by a blue line, is the value to which the parameter will default to when outside the clip. The static value will follow the parameter until any new changes of the Filter Frequency parameter are recorded anywhere in the song - before or after the clip. This way, you'll never have to bother about any "send controllers" command at the be-ginning of your song - or any "chase controllers" functionality. Everything is automati-cally and neatly set up for total recall functionality - wherever you are in your song.

If you like, you could also change the static value for an automated parameter after-wards. Do this in Edit Mode either by dragging the Static Value handle up or down or by entering another numeric value in the Static Value handle. Parameter changes in clips will not be affected by this.

If we would have wanted our automated parameter to be embedded in the note clip on the note lane instead of in a separate clip on the parameter lane, we would have clicked the Automation As Perf Ctrl button on the transport bar before recording the parameter changes. This is a nice feature if you want to make sure the parameter changes are kept

together with the notes at all times. For example, Modulation Wheel, Pitch Bend and Sustain Pedal controllers are by default recorded as Performance Controllers in the note clip.

Song build-up and arrangement

When we're finished with the verse, we'll continue with the chorus, bridge, intro and anything else we want to put in our song. We'll build up the other parts the same way we did with the verse; clip by clip, lane by lane and track by track, until we're happy with the result. To make it easier to visually identify the different parts, we rename and color the clips by selecting them and choosing Add Labels To Clips and Clip Color from the Edit menu. We also choose to join the separate clips on the Bass and Pad note lanes respectively to keep down the number of clips per lane and make it easier to manage. We copy and paste our verse so that we now have an intro, two verses, a bridge and a chorus.
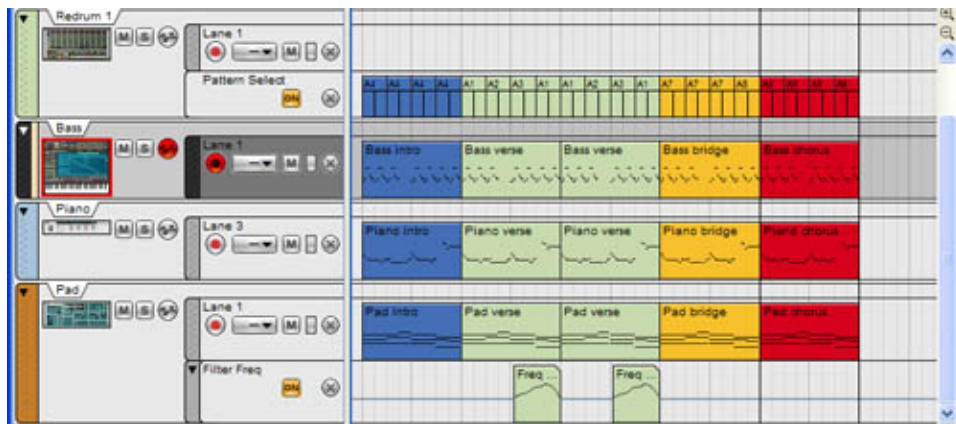
Now, we want to copy only the last four bars of the chorus and repeat a couple of times in the end of the song. A very handy method for doing this is using the Razor tool in the sequencer toolbar. We place the Razor on the track background just above the topmost track, click and draw a rectangle starting in the middle of the chorus part and covering the last four bars of the chorus
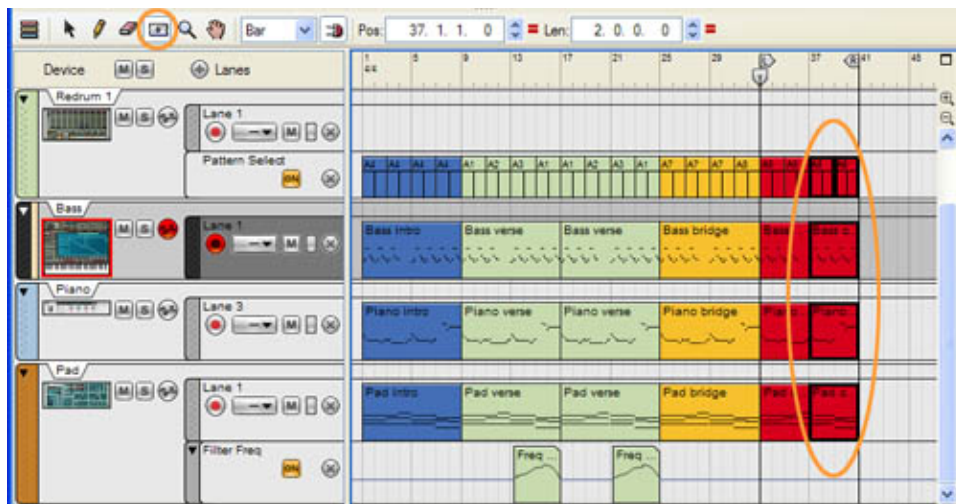
Then, we switch to the Arrow tool and press the Ctrl [PC]/Option [Mac] key and drag away two copies of selected clips to the end of the song.

We continue to build up our song until we have a rough complete arrangement. Finally, we'll do some fine adjustments and editing in the different clips to complete our song.

More about clips and sequencing

If you haven't already, be sure to check out the Reason version 4 video tutorials here!

Text & Music by Fredrik Hylvander

Propellerhead Software

DISCOVERING REASON **Thor demystified 1: The Analogue Oscillator**

In principle, Thor can be a daunting synthesiser, capable of generating a huge range of sounds and effects. But if you break it down into its components parts and seek to understand each in isolation, you'll soon find that you can understand and put sounds together like a professional sound designer.

In this series of tutorials, we're going to look at the fundamental building blocks of the audio synthesiser; the oscillators that provide the initial waveforms that we can later shape with filters, envelope generators, modulators, effects and so on. There are six oscillator types and we shall address each in turn, seeing how they differ from one another, and how you can build distinctive sounds using them.

The Analogue Oscillator

Figure 1: The basic setup

Start by invoking an instance of Thor and patching it directly to a line mixer with no effects in the signal chain. Having done so, create the basic setup shown in figure 1. As you can see, this has a single 'analogue' oscillator, no filters, no modulation, no overdrive, no effects, nothing... except for the VCA opening and closing instantaneously

when you press or release a key.

You are now in a position to inspect each of the four types of waveform generated by Thor's emulation of a traditional analogue oscillator. If it is not already selected, click on the sawtooth waveform, and press a single key on your controller - somewhere around middle 'C' would be appropriate. The result is a bright, buzzy sound that, as you become more experienced, you will instantly recognise for what it is. Now select the square wave, ensure that the PW (pulse width) knob is set to 64, and press the same key as before. The sound is still full, but has a slightly hollow quality that is often compared to that of a clarinet. Next, select the triangle wave and press the key. The sound is significantly quieter and duller. While the character is somewhat similar to that of the square wave, it's clear that something has been removed from it. Finally, select the sine wave (the bottommost of the four waveform options) and press the key. The sound is now very simple - it's what is called a 'pure' tone.

The four waveforms sound different because each has a different harmonic structure - both in terms of which harmonics are present, and at what amplitude. There is a vast amount of information explaining harmonics on the web, but much of it is inexact, and some is simply wrong. So let's summarise these four waveforms here, as follows:

What is a harmonic?

A harmonic is a frequency at which an object known as a "harmonic oscillator" will naturally vibrate. Examples of harmonic oscillators include stretched strings (guitars, violins, 'cellos, pianos, and so on) and the air in hollow, cylindrical pipes (flutes, pipe organs and so on) and their harmonic frequencies are related by simple integer numbers.

The lowest frequency of vibration for any given harmonic oscillator is called the "fundamental", and we call this 'f'. The next lowest frequency at which the object will vibrate is exactly double that of the fundamental, and we express this as '2f'. The next lowest is exactly three times the fundamental, so we express this as '3f'... and so on.

There are many other types of objects that can vibrate, such as bells, gongs, cymbals, and all manner of other percussion. The relationships between the vibration frequencies for these are often not harmonic, and may be described by complex mathematical formulae, so we will not discuss them here.

Sawtooth wave

If an object vibrates in such a way that all the harmonics are present and in phase with each other, and the amplitude of each harmonic is "the amplitude of the fundamental divided by the harmonic number", the resulting waveform is a sawtooth wave. To clarify:

- if the fundamental 'f' is present with an amplitude of '1'
- the second harmonic '2f' is present with an amplitude of 1/2
- the third harmonic '3f' is present with an amplitude of 1/3

... and so on.

The bright, buzzy nature of the sawtooth wave is, therefore, a consequence of the presence of all the harmonics. To a reasonable approximation, this wave is generated naturally by plucked and bowed strings, and by flared pipes, as used for brass instruments.

Square wave

The square wave is a special case of the family known as Pulse Waves. It has a 'duty cycle' (the amount of time the pulse remains at its upper level divided by the length of one complete cycle) of 1/2 or - as it is more usually expressed - 50%.

If an object vibrates in such a way that only the odd harmonics are present and in phase with each other, and the amplitude of each is still "the amplitude of the fundamental divided by the harmonic number", the resulting waveform is a square wave. In this case:

- if the fundamental 'f' is present with an amplitude of '1'
- the second harmonic '2f' is not present
- the third harmonic '3f' is present with an amplitude of 1/3
- the fourth harmonic '4f' is not present

... and so on.

To a reasonable approximation, the air in cylindrical pipes closed (or blown) at one end does not vibrate at frequencies of 2f, 4f, 6f... so this is why a square wave has much of the character of a clarinet.

Triangle wave

If an object vibrates in such a way that only the odd harmonics are present and in phase with each other, and the amplitude of each is "the amplitude of the fundamental divided by the square of the harmonic number", the resulting waveform is a triangle wave. In this case:

- if the fundamental 'f' is present with an amplitude of '1'
- the second harmonic '2f' is not present
- the third harmonic '3f' is present with an amplitude of 1/9
- the fourth harmonic '4f' is not present
- the fifth harmonic '5f' is present with an amplitude of 1/25

... and so on.

As you can see, the underlying nature of the triangle wave is similar to that of the square wave, but the higher harmonics are more quickly attenuated, which is why this waveform sounds duller than either the sawtooth of the square wave.

Sine Wave

This is called a pure tone because, assuming there is no distortion in the signal path, it contains nothing other than the fundamental.

Building Sounds Using Nothing But Simple Waveforms

Perhaps because most synthesisers have pre-patched signal paths, it has become common for players to create sounds using dramatic filter effects and contouring without paying enough attention to the sound generated by the oscillators themselves. However, it's remarkable how many useful sounds can be created without recourse to filters and such-like.

Let's start with a single sawtooth wave with the following parameters: octave=4, semitone tuning=0, fine tuning=0. Played as a single note   it's not particularly interesting, but some chords   demonstrate the richness of the tone. You may have noticed the slight movement in the sound in the second example. This is a consequence of the 12-note scale used in western music, which requires notes to be slightly 'out of tune' with one another so that pieces of music can be played in any key.

Let's now make things a bit more interesting by adding a second analogue oscillator set

to the sawtooth waveform (see figure 2). If this has the same parameter values as the first, you might expect the sound to be the same as before, but twice as loud. This is not the case, because Propellerhead have implemented Thor's analogue oscillators so that they are initiated with random phase. This means that the tone will differ noticeably and the sound may even disappear totally (!) when you switch on a second oscillator with the same parameters as the first.   demonstrates this as oscillator 2 is switched on and off during the course of a note. Of course, this is not something that you would often choose to do, but it demonstrates that digital oscillators do not sound natural unless you detune them or let them drift against one another.

Figure 2: Adding a second analogue oscillator

Understanding this, let's now set the fine tuning of Osc1 to +3 and the fine tuning of Osc2 to -3. That's starting to sound more interesting! The single note   has a greater depth to it, and the chords   are reminiscent of all manner of classic analogue poly-synths. An even more extreme effect can be obtain by increasing the detune to +10 and -10 or further, producing a pleasing "chorused" sound. But to maximize the quality of this, let's now add a third analogue oscillator, again producing the sawtooth waveform at the same octave and semitone as the others, but with a fine tuning value of zero. If you increase the contribute of Osc3 in Thor's mixer (see figure 3) you now have three oscillators in the sound, each slightly detuned with respect to the others, and the result is a lovely 'polysynth' sound that would grace any recording.

Figure 3: A three-oscillator 'chorused' sound

Now that you have experimented with fine tuning, we're going to create a very different type of sound by setting the fine tuning to zero for all three oscillators, but adjusting the semitone tuning.

Select the sine wave for each of the three oscillators, and set the octave and semitone tuning as follows:

Osc1: octave=4 semitone=0
Osc2: octave=5 semitone=0
Osc3: octave=5 semitone=7

This patch (shown in figure 4) generates the first three harmonics of a harmonic series. It is also one of the most common registrations used on a Hammond Organ, on which the pitches are called 16', 8' and 52/3', because these are the lengths of pipe that produce them when you play a bottom 'C'. You can hear this patch in   and it can be used as the basis of all manner of organ sounds.

Note: If you look at the Amp Env you will see that I have increased the Attack time by a tiny amount (to 2.8ms, to be precise). This is not to alter the sound, but to reduce the click that would otherwise occur at the start of the notes due to the super-fast envelopes in Thor.

An even more interesting effect is obtained if you now modify the patch to detune the oscillators very slightly against one another. Try setting the fine tuning of Osc2 to +3 and the fine tuning of Osc3 to +6. (There's a good reason why one value should be double the other. Can you work out what it is?) The result is, as you would expect, a slight chorusing of the sound, but in such as way that it creates an impression of a rotating speaker set to its low speed.   Increase the fine tuning (but always with the value for Osc3 set to double that of Osc2) and the speed of the effect will increase.

Figure 4: Constructing an organ sound

Finally, we're going to create one of the classic synthesiser sounds of the late 1970s and early 1980s; the so-called "Bass-Fifth". Return to a dual-oscillator configuration and select the sawtooth waves for each. With the octave=4 for both, set the semitone tuning for Osc1=0 and Osc2=7. Set the fine tuning to zero for both. It doesn't sound particularly interesting, does it? But now reduce the octave to '2' for both oscillators and press a note. You'll recognise the timbre of this sound instantly.

Finally (you knew that I was going to say this, didn't you...) fine tune Osc2 to +16. As before, this produces a slight chorusing effect but, because the pitch of the note is much lower, the effect is rather different from the "polysynth" timbre demonstrated in sound 6. The result � which takes us deep into the bass territory of early modular synthesisers - is shown in figure 5 and can be heard in
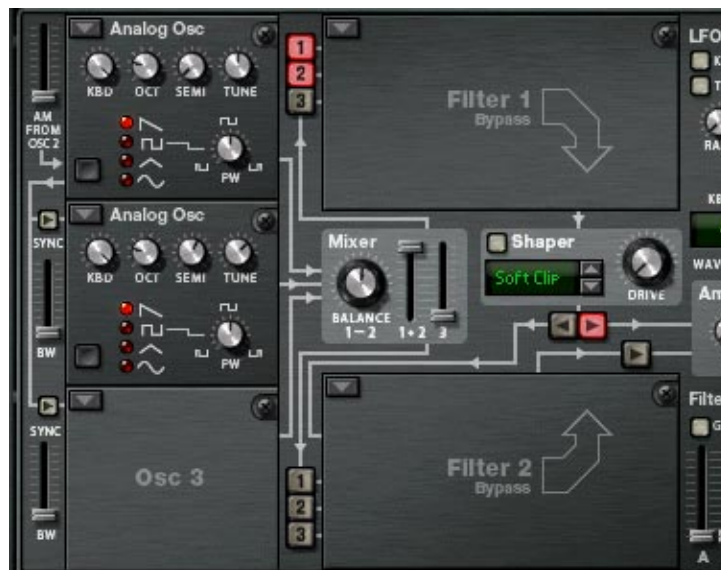
Figure 5: Deep bass

So that's it for now. We've created some distinctive and classic synth sounds without touching a single filter, modulator or (with one tiny exception) envelope generator. There's a significant lesson there.

In the next tutorial, we'll look at how you can extended your palette of sounds using two more of the sophisticated facilities built into the oscillator bank itself: amplitude modulation, and sync.

Text & Music by Gordon Reid

Thor demystified 2: Amplitude Modulation and Sync

Last month's article in our series of Thor-centric Discovering Reason articles generated a lot of positive feedback from our readers. For those interested in vintage synths and synthesis, be sure to check out article author *Gordon Reid's homepage* for more goodies. When he is not explaining Thor, Gordon writes the Vintage Synth column in Sound On Sound magazine.

In the first tutorial of this series, I showed you how to create a range of important sounds using nothing more than Thor's analogue oscillators. You might think, therefore, that the next step would be to invoke things such as filters and modulators. But even without invoking Thor's more esoteric oscillator types, there's still much more that we can do with the oscillators; things such as using amplitude modulation (AM), oscillator synchronisation (sync), and pulse width modulation (PWM). In this tutorial I'm going to introduce two of these - AM and sync - and show you how you can use each of them to create one of the 'classic' families of synth sounds: the so-called 'analogue piano' patches.

What is Amplitude Modulation?

We'll start with Amplitude Modulation but, before creating a sound using it, perhaps we should ask, "what is it?".

To answer this, imagine that you take a continuous 'pure' tone (a sine wave) and affect its loudness (or, to use the technical terminology, modulate its amplitude) using a second sine wave with a frequency of, say, 1Hz. If you do this, you will hear the previously continuous tone get louder, then quieter, drop to silence, get louder, then quieter, drop to silence... and so on, with two silences per second. This is a very common effect, and it's called tremolo.

Let's now ask what happens if you increase the frequency of the modulation? At first, the result sounds like a faster tremolo, but as the modulator moves into the audio range, a strange thing happens... you begin to hear two distinct tones, one moving up in pitch, and the other moving down. The reason for this is straightforward to explain using a bit of high-school trigonometry, but don't worry... we'll not do the maths here. Instead, I'll just ask you to accept that, if you modulate the amplitude of one oscillator (the carrier) of frequency X using another oscillator (the modulator) of frequency Y, the result is two new signals with frequencies X+Y (the sum) and X-Y (the difference). So, if you increase

Y while X remains constant, the frequency of the sum increases, while the frequency of the difference decreases. Those are the two tones that you hear.

While this may seem a bit arcane, it suggests a powerful way to obtain complex new tones using just two simple oscillators. Imagine that you have two sine wave oscillators: one with a frequency of 300Hz and the other with a frequency of 50Hz. If you combine them in an audio mixer you will obtain a sound with two frequencies present - 50Hz and 300Hz - which can be described as a fundamental and its sixth harmonic. But if you treat the first oscillator as the carrier and the second as its modulator you obtain two new frequencies, 250Hz and 350Hz, which are not harmonically related in a simple fashion. In other words, you have created a new, enharmonic, timbre.

This is not an earth-shattering result, but now imagine that you change the waveform of the carrier from a sine wave to a sawtooth wave which, as I explained in the first tutorial, contains harmonics at 300Hz, 600Hz, 900Hz... and so on. Applying the modulator to this generates partials at 250Hz and 350Hz, 550Hz and 650Hz, 850Hz and 950Hz... and up through the spectrum. This is a much more complex waveform, but it will still sound musical because all the frequencies are integer multiples of 50Hz. Next, let's change the carrier frequency from 300Hz to, say, 299Hz. This time, there are no simple relationships between the resulting partials (249Hz, 349Hz, 548Hz, 648Hz... and so on) and the result is a harsh, dissonant sound that you can not easily generate by any other means. Finally, let's change the modulator to a sawtooth wave, too... I'll leave you to work out all the frequencies present (it's a huge number) and to imagine how complex the timbre has become.

A simple example of AM

Amplitude Modulation is most commonly implemented in a device called a Ring Modulator, which you can use to create harsh sound effects, complex frequency sweeps and, when applied to a vocal signal, Dalek voices. These are all perfectly valid applications of ring modulation, but amplitude modulation can also synthesise musical tones, so I'm going to demonstrate how it can generate the waveform for an electric piano patch that will have more character than a patch based upon the basic waveforms.

Figure 1: The basic setup

We'll start with the basic patch from the previous tutorial, as shown in figure 1, and shape the output appropriately. We do this by reducing the sustain level in the Amplifier Envelope (Amp Env) to zero, and programming a decay of around 4s, which is a nice, natural decay for the type of sound that we want. If I play three notes using this (figure 2) it sounds like a typical single-oscillator polysynth patch.

Figure 2: Buzzy, shaped sawtooth patch

Next, we're going to make things a bit more interesting by adding a second analogue oscillator as shown in figure 3. The first remains a sawtooth with its pitch defined by Oct=3, Semi=0 and Tune=0, while the second is also a sawtooth, but with tuning of Oct=5, Semi=0 and Tune=0. If you now increase the AM slider to maximum, you'll modulate every harmonic of the carrier (at octave 3) with every harmonic of the modulator (at octave 5). But remember, you don't want to listen to the output from Osc2... you're using it only as a modulator, so you must make sure that the "2" button alongside the empty Filter 1 panel is switched off. If I play the same three notes using the modified patch, the result   has a very different tone.

Figure 3: Adding a second oscillator and applying amplitude modulation

At this point, the sound is still too buzzy, so we're going to place a low-pass filter into the Filter 1 panel. You don't need to do anything too clever with this because you simply want to attenuate the higher harmonics, so set the cut-off frequency knob to around 12 o'clock, add a little resonance, and set the KBD (keyboard tracking) to around 12 o'clock. (See figure 4.) Playing the same three notes now produces sound 4 ▶, which is starting to sound a little like an electric piano, especially if I play something suitable from the 1970s to demonstrate this

To illustrate the effect of the Amplitude Modulation, you should experiment with various positions for the AM FROM OSC2 slider. As you will hear, moving it the bottom to eliminate the AM removes the 'tine-y' timbre, and the result sounds much more like a basic synthesiser patch.

You can of course refine the patch in figure 4 still further, filtering the sound more carefully, using additional envelopes to shape the results more accurately, and even adding various forms of low-frequency modulation and effects to create something that could be very close to the output from a Rhodes or Wurlitzer electric piano. I've gone a little way toward this with Sound 7 ▶, but there's no room to discuss this further. Instead, I'll leave you to experiment.

Figure 4: Static filtering of the sound in figure 3

Building a similar (perhaps better) sound using Oscillator Sync

Oscillator sync is very different from Amplitude Modulation. The easiest way to describe it is to say that when one oscillator (in this case, Osc1, the master) completes one cycle of its waveform, it resets the waveform of the "synchronised" oscillators (in this

case, Osc2 and/or Osc3, the slaves). This means that the pitch of the master determines the pitch of the sound, while the pitch of the slave, if at a lower frequency, determines its timbre. Although sync can be hard to envisage, there's a good diagram in the Thor manual, so you should refer to this.

Oscillator sync is often used to create a distinctive type of sound, sometimes referred to as 'tearing' because it feels as if the sound is being 'torn apart'. Sound 8   (generated by the patch in figure 5) is an example of this, obtained by sync'ing Osc2 to Osc1 and then sweeping the frequency of Osc2 using one of Thor's envelope generators.

Figure 5: A typical 'sync sweep' patch

You may have noticed in the figure that the BW (bandwidth) fader beneath the Osc2 sync button is set to its maximum. Unlike the AM slider (which controls the depth of the modulation) this controls how 'hard' the synchronisation is. Some early analogue synths offered 'hard sync', which meant that the slave was reset almost instantly (i.e. with a hard edge to the resulting waveform) each time that the master completed a cycle. Others offered 'soft sync', in which the reset was less abrupt and the resulting waveform had a softer edge. Thor offers you both, and everything between, allowing you to choose how dramatic the result will be.

The example in Sound 8 is hard sync and, as you can hear, its tone changes radically as time progresses, which means that the harmonic content of the sound is different at every stage of the note. This suggests yet another type of synthesis; using sync to generate interesting new waveforms. In other words, instead of sweeping the sync'd oscillator in an arbitrary fashion, you can select the difference between the master and oscillator frequencies carefully, and use envelopes and modulators (and so on) to shape specific sounds. Again, I'll demonstrate this by synthesising an analogue piano.

We'll start with the patch in figure 4, but reduce the AM slider to zero. Now, leaving Osc1 at its previous settings (Sawtooth, Oct=3, Semi=0 and Tune=0) we'll set Osc2 to the following: Sawtooth wave, Oct=4, Semi=11 and Tune=-50. (See figure 6.) If you switch on both oscillators and play a few notes, this patch emits a rather discordant sound with a slightly metallic flavour caused by the detuned relationship between the two oscillators.

Figure 6: Tuning the oscillators for a 'sync' piano

If you now eliminate the audio signal from Osc1, switch on the SYNC button for Osc2 and push the BW slider to its maximum, you obtain a sound that is in tune, but with a more electric-piano-like timbre. (Indeed, sound 10   is not dissimilar to the sound that we obtained using Amplitude Modulation.) But here's a neat trick: if you switch the audio from Osc1 back on again, you'll find that Osc1 contributes 'body' to the sound, while the sync'd Osc2 provides the overtones.   Nevertheless, the sound is still not particularly like an electric piano, and it needs a little additional filtering to shape the way that the brightness of each note changes in time.

If you look at figure 7, you'll see that I have reduced the filter's initial cut-off frequency to zero and the resonance to zero. If I play the controller keyboard with these values, I obtain silence because the filter is completely closed, so I'll make it open when I hit a note and then close over the course of a few seconds by setting the ENV amount to around 100 and setting up the Filter Envelope as shown. I'm also going to set the VEL amount to 100 or so, which means that notes played with greater velocity are brighter than those played more gently. Suddenly... it's an electric piano!

Figure 7: A 'sync' piano

Of course, this just a single example of a 'sync' sound, and there are myriad other patches that take you far beyond the simple "zeeoooowww" in sound 8. Experiment with this patch, changing the oscillators' waveforms and pitch relationships and, even with such a (relatively simple) sound, you'll be surprised at how many variations you can develop. What's more, there's nothing stopping you from using amplitude modulation and sync with other oscillator types, or even at the same time! In some cases, making a change will have no audible effect - for example, it is only the frequency of the master oscillator that matters when patching with hard sync, not its waveform - but if you change the slave to another type, the effects can be radical. Unfortunately, that's a story for another day...

Text & Music by Gordon Reid

Thor demystified 3: Pulse Width Modulation

DISCOVERING REASON In the early days of analogue synthesis, electronic components and designs were far more expensive than they are now, so manufacturers such as Korg entered the market with simple, single-oscillator monosynths. Yet these instruments could, if programmed correctly, sound surprisingly rich and full. Much of the appeal of the Korg 700 (for example) was a result of its Chorus I and Chorus II waveforms, but while the word 'chorus' was a good description of how these waves sounded, it didn't tell you what they were. In retrospect, a more accurate name would have been "pulse waves whose duty cycles (or 'pulse widths') are being modulated by a low frequency oscillator", but how many players would have understood that in 1974?

Nowadays, this "pulse width modulation" (PWM) is a standard facility on all analogue and 'virtual analogue' synths, and it remains important because of this ability to create rich, chorused sounds using a single oscillator. But before showing you some examples of how you might like to use PWM in Thor, let's begin by asking...

What is Pulse Width Modulation?

To explain this, let's start with the square wave in figure 1. As you can see, this is a special case of the family of pulse waves, defined by the fact that the wave spends exactly the same amount of time at the top of the waveform as it does at the bottom. A slightly more technical way of describing this is to say that it is has a duty cycle of 50% (because the wave is at the top of its cycle for exactly half of the time).
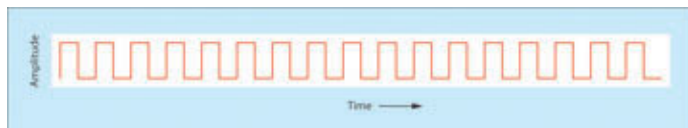
(Click to enlarge)

Figure 1: A pulse wave with a duty cycle of 50%

Of course, there's nothing stopping us from generating pulse waves with any other duty cycle we please, from 0% (where the wave is permanently rooted to the bottom of the cycle, and is therefore silent) to 100% (where it is rooted to the top of the cycle, and again silent). For example, figure 2 shows a pulse wave with a duty cycle of 25%.
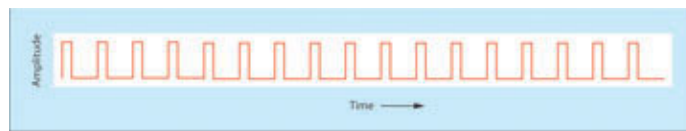
(Click to enlarge)

Figure 2: A pulse wave with a duty cycle of 25%

On some analogue synthesisers (such as the Minimoog) you are merely offered a selection of pulse widths from which to build sounds, but this is a shame because it's not hard to design the electronics so that a modulator can affect the duty cycle. For example, you could start with a square wave and use an LFO to sweep the duty cycle repeatedly from 0% to 100% and back again. I've illustrated this in figure 3, which shows the initial square wave (the red line) the triangle wave LFO modulating the duty cycle (the green line) and the resulting waveform (the blue line).
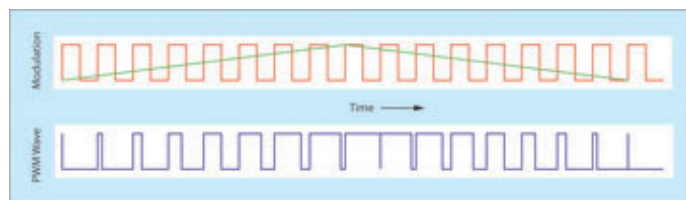
(Click to enlarge)

Figure 3: A triangle wave modulating the pulse width

Without going into the maths of PWM, it should be intuitively obvious that the harmonic content (and therefore the tone) of the sound is changing from moment to moment as the shape of the waveform changes. However, this still doesn't explain why sweeping the duty cycle makes a single oscillator sound 'chorused'. To cut a long story short, it's because PWM using a triangle wave as the modulator splits the single pitch of the initial oscillation into two frequencies such as those shown in figure 4. This is the same as having two oscillators, with one being frequency modulated with respect to the other. If you think back to what we achieved by detuning two oscillators in the first of these tutorials, this is a wonderful result, and it suggests all manner of synthesis possibilities.
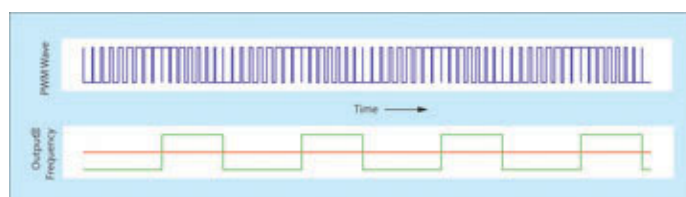
(Click to enlarge)

Figure 4: The frequencies of the two components that comprise the PWM waveform

PWM Example 1 - the PWM bass sound

One classic application for PWM is the creation of thick 'synthy' bass lines, so this is where we'll start. Set up a simple, single-oscillator bass sound. Insert an analogue oscillator in Osc1, set it to Octave 2, and select Mono Retrig as the keyboard mode. Now insert a low-pass filter as shown, with moderate drive and full keyboard tracking. There's no need for a filter envelope or anything else more sophisticated because the filter will be used simply to attenuate the high frequencies in the sound. Finally, set the Amp Env Sustain to maximum to create a 'square' shape for the sound, and you're done. (See figure 5.) Sound #1   shows that this has a suitably low pitch, but that it is featureless and rather boring.

(Click to enlarge)

Figure 5: A very simple bass patch

Experimenting with the other waveforms in Osc1 doesn't help, but don't despair... we can improve this sound without invoking extras oscillators or complicated filtering. Set the oscillator to be a pulse wave, and set its duty cycle to 24, which is approximately 20%.   That's no better, you might say, and you would be right. But now we'll invoke PWM. Choose a slot in the modulation matrix and select LFO1 as the modulation

source, with the pulse width of Osc1 as its destination. Now we have to choose suitable parameters for the LFO and the modulation depth. Experience shows that PWM works well at bass frequencies if the modulation speed is quite slow - around 1Hz - and at high frequencies when it is somewhat faster - around 5Hz. We can make this happen by setting the LFO rate to around 1.5Hz and having it track the keyboard at around 50%, which is a value of 63 or thereabouts. Once you have set this, a modulation depth of around 40 creates a nice effect; any less would be too little for my taste, any more would be too much. (See figure 6.) The result is contained in Sound #3.

(Click to enlarge)

Figure 6: A much better PWM bass patch

I discovered this sound on a Korg 700 in the early 1970s, and I still like it a lot. It is rich and involving, but not so complex that it makes the mix too thick or muddy. What's more, it is ideal for further filtering and shaping, and it requires just a single oscillator and only the simplest synthesiser architecture to obtain it, which is just as well, because the little Korg had just a single oscillator and only the simplest synthesis architecture!

Thor demystified 4: The Multi Oscillator

 In the first of these tutorials, I demonstrated how you can use detuning to create complex 'chorused' sounds. In the third, I showed how Pulse Width Modulation of a pulse wave (PWM) splits a single frequency into two components, thus emulating the sound of two detuned oscillators. Both of these are powerful synthesis techniques, but they are limited (in the first case) to the number of oscillators in the synthesiser, and (in the second) by the fundamental nature of PWM and what it can achieve.

Happily, unlike true analogue synths, Thor incorporates an oscillator type specifically designed to provide an extensive range of rich, chorused sounds, and it can do so using just a single oscillator slot. The beastie in question is the Multi Oscillator, and in this tutorial I am going to use a single instance of this to create one of the most challenging of all synthesised sounds: the 'analogue' choir.

What is a Multi Oscillator?

The Multi Oscillator is quite a complex module, but if you break its operation down into individual concepts, all becomes clear.

Let's start by reloading the basic patch from the start of tutorial #1, but instead of an Analogue Oscillator, we'll insert a Multi Osc in the Osc 1 slot. You can now see that determining the waveform, the pitch (octave, semitone and fine tuning) and the keyboard tracking is the same as before. Sure, the set of waveforms offered is slightly different from the Analogue Oscillator, but this shouldn't obscure the fact that – at this level – you control the Multi Osc in exactly the same way as the Analogue Osc. So let's now look at the bits that are different: the Detune Mode and the AMT (amount) control...

With the AMT knob set to zero, the Detune Mode is irrelevant, and a Multi Oscillator generates a single frequency just like an Analogue Oscillator, so you can use it in the same way. Hmm... this is not strictly true, because the Fifth and Octave modes produce two pitches when the AMT is zero, as described below. But the key here is that none of the modes produces a detuned sound when the AMT is zero. So, let's move on...

When you increase the AMT, the Multi Osc will generate eight pitches simultaneously. This means that, for example, if you select the sawtooth wave and (say) the Random 1 mode, and then increase the AMT above zero, eight separate sawtooth waves are creat-

ed, each slightly detuned with respect to the others. This is equivalent to having up to eight sawtooth waveform generators in a massive oscillator bank, but much more economical in form and use. What's more, there are things that you can do with the Multi Osc that would not be practical with eight individual oscillators. (We'll touch upon this later in this tutorial.)

The final element in the Multi Osc is the Detune Mode itself, which determines the way in which the frequencies of the eight waves are distributed. At low values of AMT, these can produce quite subtle results, but at high values of AMT, they can be wildly different, and you will be able to find many way to use them to create new sounds. The eight modes are:

- **Random 1**

  At AMT=zero there is no detune. As AMT is increased, the pitches of the frequencies are detuned according to a quasi-random distribution that was selected by the developers to have particularly pleasing characteristics.

- **Random 2**

  At AMT=zero there is no detune. As AMT is increased, the pitches of the frequencies are distributed according to a second quasi-random distribution. The character of Random 2 is slightly different from Random 1, and this is particularly noticeable when the AMT is high.

- **Interval**

  At AMT=zero there is no detune. As AMT is increased, the eight oscillators are detuned equally in cents.

- **Linear**

  At AMT=zero there is no detune. As AMT is increased, the eight oscillators are detuned equally in Hz. This means that this mode is a good starting point for synthesising enharmonic sounds.

- **Fifth Up**

At AMT=zero there is no detune. At maximum AMT, four of the oscillators are tuned a fifth (seven semitones) above the played pitch, with both sets of four oscillators slightly detuned.

- **OctUpDn (Octave Up and Down)**

  At AMT=zero there is no detune. At maximum AMT, half of the oscillators are detuned down one octave while the others are tuned up one octave, again with both sets of four oscillators slightly detuned.

- **Fifth**

  At AMT=zero, half of the oscillators are tuned up by approximately a fifth (seven semitones). As AMT is increased, each of the two groups of four oscillators is detuned quasi-randomly.

- **Octave**

  At AMT=zero, half of oscillators are tuned up by approximately an octave. As AMT is increased, each of the two groups of four oscillators is detuned quasi-randomly.

Having digested all of this, we're now ready to create sounds with the Multi Oscillator.

A Classic Analogue Patch...

There are many sounds that you can synthesise using the Multi Oscillator, from conventional things such as ensemble strings and brass to enharmonic sounds, percussion and extreme 'off-the-wall' effects. However, there's one sound to which it's better suited than any other form of analogue-style oscillator, and that is the 'human' choir. The reason for this should be obvious: a choir is an ensemble of people who are singing at almost the same pitch, with quasi-random deviations between each person's voice and that of the others in the ensemble.

So let's start by invoking a Multi Oscillator in the minimal configuration used as the starting point for previous tutorials. (See figure 1.) I have chosen the sawtooth waveform in the oscillator, and applied a gentle Attack and a moderate Release in the Amp Env. It's a pretty boring timbre, and there's not much you would normally choose to do

with it. Sound #1

Figure 1: Setting up a Multi Oscillator

We can improve this considerably by invoking the 'multi' aspect of the oscillator, select-ing one of the Detune modes, and dialling in an appropriate AMT. In this case, I'm go-ing to choose Random 1 mode and apply a detune of 28, as shown in figure 2. Sound #2 demonstrates that this is much more promising. Indeed, without any further synthesis – no filters, no modulation, nothing... – this is an excellent sound that would grace many tracks.

Figure 2: Creating a rich, detuned sound

In a moment, we'll move on, invoking other modules to modify the sound. But before doing so, I'm going to introduce some useful tricks that demonstrate how subtle the Multi Oscillator can be. Let's turn to the Modulation Matrix and link the amount of oscillator detune to the pitch of the note so that, at low pitches, there will be less detune than at higher pitches. This will increase the depth of the ensemble effect as you play up the keyboard, simulating what tends to happen in a real, human choir. Having done so, let's also invoke a 6Hz triangle wave LFO and direct this to the amount of detune. This is a wonderful facility, because it means that we can increase and decrease the amount of detune of the eight oscillators to create a wonderfully rich, chorused sound. (See figure 3.)

Sound #3 is a simple chord sequence using played using the unmodulated Multi Oscillator. Sound #4 has both of these types of modulation present and, as you can hear, it is more heavily chorused, with a complex, tremulant quality that will add character when we turn this 'stringy' sound into a choral one.

Figure 3: Adding detune modulation

Now we're going to introduce something a bit special. This is Thor's Formant Filter, which synthesises the sound of the human vocal tract. Describing the Formant Filter in detail is beyond the scope of these tutorials, but if you experiment with it, you will find that the position of the dot in the middle of the module's X/Y display enables you to create a wide range of vowel sounds. Mind you, this is not as simple as it seems. Listen to Sound #5. This is the Multi Oscillator sawtooth wave with no detune (AMT=zero) played through the Formant Filter set up to create a deep, basso voice. It has a vocal quality, but it's nothing special. Now listen to Sound #6, which is exactly the same patch but with detune applied. Wow... this is a remarkable result, and you may recognise it as an imitation of the revered Roland VP-330 Vocoder Plus, the instrument that provided much of the character of Vangelis' soundtracks for Chariots Of Fire and Blade Runner, and which was used by numerous players to replace their Mellotrons!

Figure 4: Creating a vowel sound

We can now play the chord sequence again, but through the Formant Filter. Sound #7 This is a better choral sound than you'll obtain from almost any analogue synthesiser, and it would be a fantastic result if it created the choral sound that I want across the whole range of the keyboard. Unfortunately, it doesn't. This is because, if I want to simulate male voices in the bass and female voices in the treble, the Formant Filter needs to change its characteristics as the pitch of the sound changes.

We can make this happen in two ways. Firstly, I'll make the Formant Filter track the keyboard to a certain extent (the value I have chosen is 69). Secondly, I'm going to return to the Modulation Matrix and make the Gender parameter track the keyboard. The improvement is far from subtle. Sound #8

Figure 5: Making the Formant Filter track the keyboard

Our sound is now all but complete, and perfectly useable. However, I am going to refine it a little further to obtain exactly the result I want. To do so, I'm going to add a low-pass filter in the Filter 3 slot. This will remove the higher frequencies and give the patch a band-limited quality that you will recognise instantly if you're a Mellotron aficionado. I'm also going to add a little delay to spread the result across the stereo field. (See figure 6.)

Figure 6: Adding a low-pass filter and delay

Playing the chord sequence again reveals that the female character of the sound has been retained at higher pitches (Sound #9) so I'm now going to add a few deeper notes to reintroduce the bass and tenor (male) singers. The result is Sound #10, which I think is gorgeous. What's more, I've created it without using a single chorus unit or any external effects!

So, could you emulate the Multi Oscillator's sound using Analogue Oscillators? To be honest, neither in Thor, nor in anything short of a room-sized polyphonic modular synth that offers eight oscillators and eight dedicated LFOs for each note that you play. Not allowing for the overlap in notes caused by the contour generator's Release, the chord sequence in these examples uses six notes at a time, which means that it would require 48 analogue oscillators and 48 dedicated LFOs to begin to recreate Sound #10. As you can now appreciate, the Multi Oscillator is a very powerful and desirable element within Thor.

Text & Music by Gordon Reid

Thor demystified 5: The Noise Oscillator

Noise is all around us. But what is it? Some people might find the scream of an internal combustion engine revving at 19,000 rpm to be a deafening noise, but it is like music to a fan of Formula 1 motor racing. Conversely, much of modern dance music is an incomprehensible noise to anybody over 50 years old, and my mother thought that even Mahler's symphonies were a horrible noise. In all of these cases, the word "noise" is simply being used to mean "a disliked sound", but in synthesis it has a much more precise set of meanings.

What is noise?

All electronically produced sounds are subject to noise. In the audio world, a simple view of a cyclic waveform might suggest that its peak amplitudes are +1 and –1 on some arbitrary scale, but the moment that you turn the perfect representation into something tangible, the thermal noise in the electronics adds or subtracts a small amount from the signal at every point in time. This means that the peaks may lie at 1.0003, or -1.0001, or +0.9992 or -0.9985, or whatever. What's more, every point between the peak values will also be subject to a small deviation from the ideal.

If these deviations are in some way systematic – say, caused by the superimposition of some form of a hum or buzz – you will hear a sound defined by each of the components, and the result will not be noise because it is not random. But if the deviations are truly random, you will hear broadband (i.e. covering a range of frequencies) noise which, on a simple level, can be described as the adding or subtracting of a random amplitude from the ideal signal at every frequency and at every point in time.

There are many sounds that you can consider to be valid examples of this type of signal, but they can be very different from one another. For example, the noise of tape hiss is very different from the noise generated by cheap air conditioning units, and both of these are very different from the sound of a truck rumbling down the street just outside your studio. So, what – apart from their loudness – is the principle difference between these signals? The answer is that they contain different amounts of noise at different frequencies. Tape hiss seems to contain predominantly higher frequencies, while the truck seems to generate predominantly lower frequencies, which we call rumble. Consequently, we must define broadband noise not only by its loudness, but also by the relative amounts of signal present at each of the frequencies in the audio spectrum.

The colour of noise

Because noise is random, we cannot say that a given frequency is present at any given moment or, if it is, at what power. We can only say that there is a probability of this frequency being present at that power at any given moment. If we then make this statement for every possible frequency, we create something called a power spectral density ("PSD").

If the PSD is constant across the audio spectrum, we can say that "averaged over a reasonable period, the amplitude of the noise at any frequency will be equal to a constant value". We call the resulting signal white noise because, when our eyes encounter visible light with all the frequencies present in equal amounts, it looks white to us.

You might think that, because all the frequencies in white noise are present in equal amounts, we would hear this noise as neutral, spread evenly across the whole audio spectrum. However, this proves not to

be the case. Our ears and brain are such that, when presented with white noise, the higher frequencies seem to dominate, and we hear something that sounds predominantly hissy.

A PSD that sounds less 'coloured' to the human ear is one in which the upper frequencies contain less power that the lower frequencies. If we define the PSD of the noise such that the power is inversely proportional to the frequency, we have described a type of broadband noise that rolls off at approximately 3dB/octave across the audible spectrum. This means that, instead of obtaining equal powers in bands of constant width, the noise has equal power per octave. This complements the response of the human ear in such a way that the noise now sounds evenly distributed. Extending the previously analogy with visible light, this PSD would look pink so, in the audio world, we refer to this noise as pink noise.

White and pink noise are not the only power spectral density functions, and other types of noise are also important, not just in audio, but in fields such as communications theory, image processing, and even chaos theory. If the noise rolls off inversely to the square of the frequency (which is what you obtain if you apply a 6dB/octave low-pass filter to white noise) you obtain red noise, so-called because light with this distribution would look red.

Alternatively, you could have a spectrum in which the noise power increases rather than decreases with frequency. The power spectral density of so-called blue noise is proportional – rather than inversely proportional – to the frequency, and the noise power increases by +3dB/octave. Likewise, the PSD of violet noise increases according to the square of the frequency (i.e. at 6dB/octave).

If the concepts presented here are a bit daunting, don't worry about it. I have shown all five of these noise types in figure 1, which should help to make everything clearer.
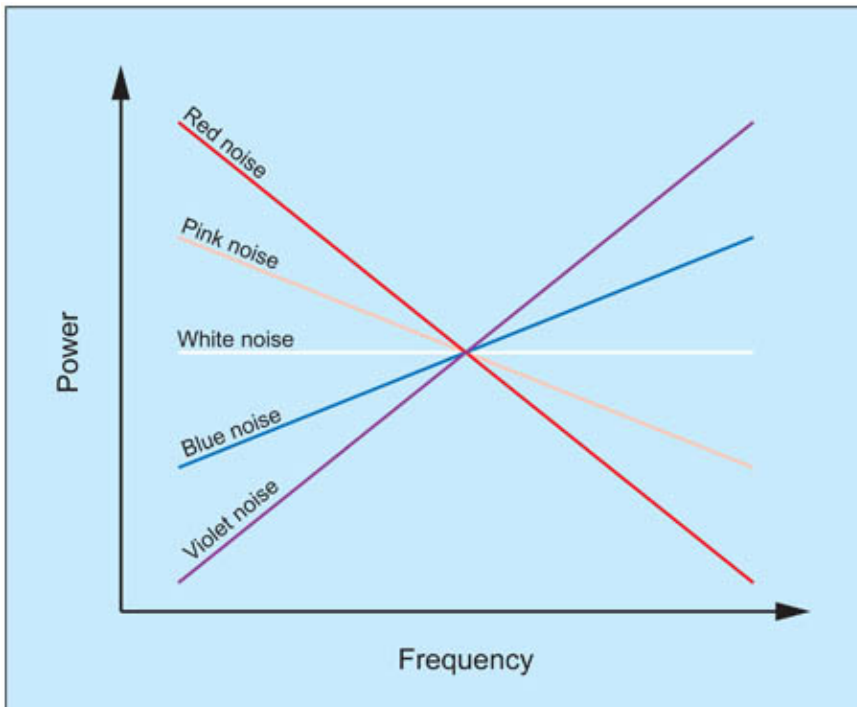
Figure 1: Types of noise

More types of noise

There are many types of noise that do not conform to this description of broadband noise. For example,

some audio exhibits artefacts such as clicks and crackle. Clicks are individual impulses whose durations, in the digital domain, can vary from single samples to tens or even hundreds of samples. Crackle is generated by a high density of smaller impulses distributed randomly in time. (If these small impulses are regularly spaced, you hear a buzz rather than crackle, but we need not worry about that, here.) So... can we call clicks and crackle "noise"? Of course we can; they are simply forms of impulsive noise rather than broadband noise.

Now, having classified these sounds as noise, and having defined some of the colours of broadband noise in a mathematical sense, I have to confuse the issue further because there are many signals that – in isolation – may be considered to be noise, but which are nonetheless important and wanted components in musical sounds. The most obvious examples of this are percussion instruments. Their sounds are based in very large part on impulses plus shaped broadband noise yet, played appropriately, they sound musical rather than noisy. Even instruments that generate primarily harmonic sounds generate signals that are "noisy". Examples include the breathiness of a flute or panpipe, the plucking sound of a guitar, the hammer noise of a piano, the sibilants of human voices... and many, many more. These are all noise, and yet they are also hugely important part of the genuine sounds, and therefore an important part of synthesis.

Yet another type of noise: Tuned broadband noise

So far, I have only discussed noise as contained within or superimposed upon a wanted signal. But ask yourself what happens when the noise is generated when there would otherwise be silence. The answer, of course, is that the noise is the totality of the signal. In other words, the sound you hear is only noise, and most synthesisers offer a means to generate broadband noise in isolation, at the same level as the cyclic waveforms that form the basis of harmonic sounds.

Imagine that you have a white noise generator on your synthesiser, and that you filter it in some fashion. I have already explained that, if you apply a 3dB/oct low-pass filter you will obtain pink noise, and a 6dB/oct LPF will give you red noise, so it's simple to extrapolate that the common 12dB/oct and 24dB/oct LPFs will create what we might call "infra-red" noise spectra. Conversely, if you apply 3dB/oct high-pass filtering to the white noise signal you will obtain blue noise, a 6dB/oct HPF will create violet noise and – by extension –steeper HPFs will create various forms of "ultra-violet" spectra. In the next tutorial, I'll demonstrate how you can use these types of noise to generate a range of sounds but, before that, I want to ask what you'll obtain if you apply band-pass filtering...

As figure 2 shows, you can constrain the bandwidth of white noise more and more tightly until, at the extreme, only a tiny band of frequencies can pass, and at this point the sound becomes less like noise and more like a somewhat "noisy" oscillator. The sound thus produced has a unique character that can form the basis for many of the so-called "spectral" patches that became popular after the launch of the Roland D50. Happily, Thor is more than capable of recreating these, as I will now show...
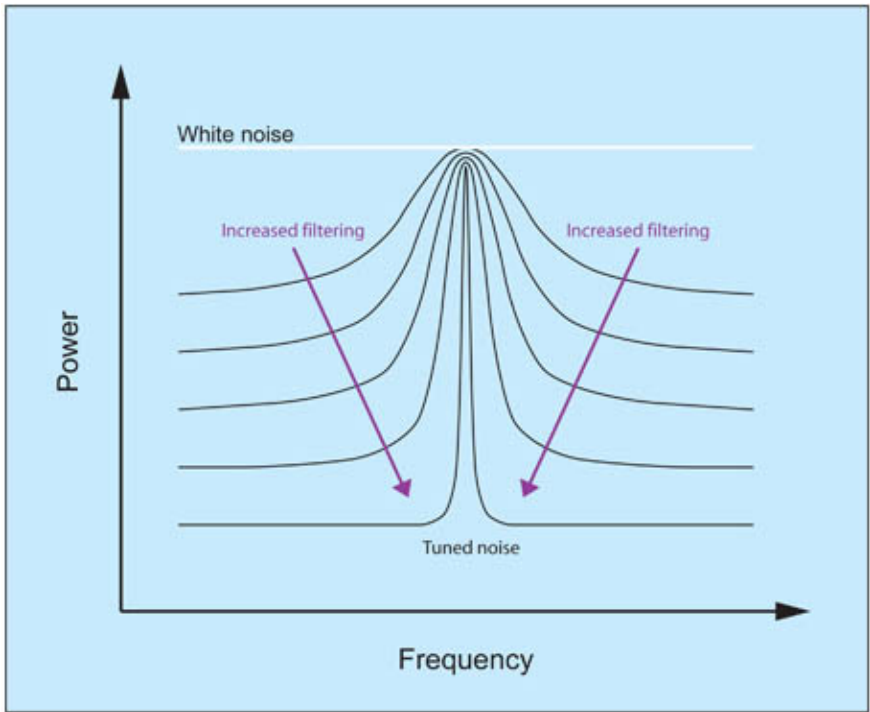
Figure 2: Filtering noise to create tones

Figure 3 shows Thor's Noise Oscillator, with its band-pass option ("BAND") selected, and the noise parameter knob (the one in the lower right-hand corner) turned to its maximum value, which causes the oscillator to produce an approximation to white noise.

Figure 3: Thor's Noise Oscillator in BAND mode

If you invoke the starting patch from previous tutorials and insert the noise oscillator into the oscillator 1 slot, you will obtain the patch shown in figure 4. Set it up as shown, with KYBD=127, Oct=5, Semi=0, and Tune=0. If you now press any key, you will hear white noise. Furthermore, this sound is invariant no matter where you play on the keyboard. If you think about it, this is correct... if there is no tonal content or variation in the noise spectrum, the very definition of the sound precludes you from hearing higher or lower notes.

(In truth, there is some variation to the character of the noise in Thor's noise oscillator, and you will hear that it becomes 'grainy' when you play low notes. This is a consequence of the coding within Thor, and not something that you would hear in a mathematically perfect "noisy" world.)

Figure 4: A simple white noise patch

If you now listen to sounds #1 [▶] , #2 [▶] and #3 [▶] , you can hear how this patch sounds. This is the same note played firstly with the noise parameter at 127 (white noise), secondly with it set to 64 (band limited noise), and thirdly with it set to zero (tightly tuned noise).

I'm now going to create a simple envelope to soften the effect of sound #3, with an Attack of approximately 0.5s, and a release of around 2s. Since I find the fully tuned sound in #3 a little too pure, I am also going to broaden the spectrum a touch by increasing the noise parameter value to 6. This results in a haunting sound that lies somewhere between the sound of a blown bottle, a finger rubbed around the rim of a wineglass, and a siren (the mythological and rather alluring Greek female, not the klaxon that you find on top of a police car). To turn this patch into something that sounds like it may have come from a D50, I need only add reverb. To demonstrate this, I selected the RV7000 Advanced Reverb module within Reason and loaded the factory preset "EFX Scary Verb", and then played a simple chord sequence to create sound #4 [▶] .

Figure 5: "Bottled Sirens"

I love this sound, but it lacks the depth that I want today, so I'm now going to add a second oscillator, set up in exactly the same way, but tuned an octave lower than the first. (See figure 5.) If I now play the same chords, I obtain sound #5 . You have to admit, this is a pretty amazing sound given that there no filters involved, no LFOs, no modulation of any other sort, no chorus, no delay, the simplest of AR envelopes, and just reverb that adds ambience to the output from Thor itself. What's more, you can't easily obtain this type of sound in any other way.

Finally, to demonstrate how powerful tuned noise can be as a generator of superb synthesised sounds, I added a single module within Thor to the patch in figure 5, and then recorded sound #6 . However, I'm not going to tell you what that module was. Can you work it out? Answers on a postcard...

Text & Music by Gordon Reid

Thor demystified 6: Standing on Alien Shorelines

In the last tutorial, I promised to show you some of the ways in which you can use the Noise Oscillator in Thor to generate a range of sound effects, and that's what we're going to discuss in a moment. But before that, I have to fulfil a promise to the chaps at Propellerhead to explain how I generated the final sound in the last tutorial. (It's reproduced here as sound #1 .)

The answer is remarkably simple: I added a Comb Filter to the Sirens patch, as shown in figure 1. The "notes" that you hear in the sample are not generated by pressing different keys on a MIDI controller keyboard; they are determined by the position of a knob: in this case, the cut-off frequency of the comb filter. When the peaks of the comb filter are in the right positions, they accentuate the harmonics of the tuned noise generators in the Sirens sound so you hear musically related notes and, with just a little practice (very little, in fact it took me just 10 minutes or so), I found that I could "play" melodies and create all manner of effects by holding down a couple of notes and then sweeping the FREQ to the appropriate positions.

Mind you, if you try to recreate this on a PC or Mac alone, you'll find that it's quite hard. I was unable to master the technique using a mouse or track-pad, but when I mapped one of the knobs on my Korg MS20iC USB controller to Thor's Filter 1 cut-off frequency, I was able to create some excellent effects. Of course, you don't need the Korg keyboard to do this; any physical controller with a suitable knob will allow you to do the same.

Figure 1: Tuned noise patch with comb filtering
(Click to enlarge)

Back to the early 70s

OK, having sorted that out, let's now turn our attention to the broad and intriguing area of noise-based sound effects. If you jump into the Reason time machine (it's the same model as I used for many years at Sound On Sound) and travel back to the early 1970s, you'll find that – even in the early days of synths

– the method of using filtered noise to generate a range of 'natural' sounds was well established.

Tom Rhea's highly respected book of Minimoog sound charts offered patches with self-explanatory names such as Jet Plane, Surf, Thunder, and Wind, while the ARP2600's patch book offered more complex delights including Clapping Thunder, Mother Whistler, Water Drops, and the long-forgotten Stereo Chickadee Conversation! All of these patches shared one thing in common: they were based on nothing more than the output from the synth's noise generator, shaped using a filter, some envelopes, and (in the case of the ARP2600 patches) the occasional application of some of the more esoteric voltage control modules.

Unfortunately, we don't have space here to investigate all of the ideas contained within the Moog and ARP patches, but I'm going to show you how we can create two of the most common noise effects – wind, and waves – and then combine them in Thor to create a sound that, for me, evokes images of strange, alien shorelines. (Cue 'The Hitchhikers' Guide To The Galaxy' "That sunset! I've never seen anything like it in my wildest dreams. The two suns! It's like mountains of fire boiling into space" and so on.)

Hmm before I get too carried away, let's return to Earth, and start by invoking the Sirens patch from tutorial #5, with the RV7000 reverb still in place, but without the comb filter. (See figure 2.)

Figure 2: Bottled Sirens
(Click to enlarge)

Generating broadband noise

As you may remember, this uses tuned noise to create an aetherial vocal timbre. But for wind and waves, we do not want tuned (i.e. pitched) sounds; broadband noise* is much more suitable as the starting point for these. Thor's Noise Oscillator provides many ways to obtain this type of noise, and all five Noise Waves (Band, S&H, Static, Color and White) can be used, but I'm going to stick with the Band option, and reduce the amount of tuning to create a sound that is much closer to white noise*. I do this by increasing the Noise Mod parameter contained within each of the oscillators. In the case of Osc1, I'm going to increase it to a value of 99, which generates broadband noise with a slight tonal quality while, for Osc2, I'm going to turn the knob fully clockwise, thus increasing the Mod all the way to 127 so that its output is, essentially, white. Figure 3 shows the modifications I've made, and the sound that this

generates is captured in sound #2 .

* For an explanation of terms such as broadband noise and white noise, please refer back to tutorial #5.

Figure 3: Generating broadband noise

Shaping waves

I now want to shape this to simulate the sound of waves breaking upon a shore, and I'm going to do so by adding a traditional 24dB/octave low-pass filter into the signal path. I'm going to give it a cut-off frequency of around 200Hz, lots of input ("drive") level, and a slowly opening and closing envelope defined by the A and D settings in the Filter Env panel. The result of this is that, at the start of the sound, only low frequencies are allowed through. Then, as the filter opens, more and more high frequencies are passed. Finally, as the filter closes, only low frequencies are passed again. While hard to describe in words, the effect of the AD contour is immediately apparent when you listen to sound #3 . (The eagle-eyed among you will have noticed that I have also extended the Amp Env to create a long, slow amplitude envelope. The reason for this is, I hope, self-evident.)

Figure 4: A single wave
(Click to enlarge)

Breaking waves

I find this sound to be very intense and, due to the slightly grainy nature of the Noise Osc, it is already starting to assume some of the character of a wave breaking on a seashore. However, there's a huge amount of low frequency energy present and, if you have large speakers, you'll hear a deep rumble within the sound. You may like this (it would certainly have the desired effect in a science-fiction movie seen in a large cinema) but I think that it is a bit excessive, so I've decided to tame it by inserting a high-pass filter into the Filter 3 slot. The HPF in Thor's State Variable Filter is not particularly steep (it has slope of 12dB/octave) so I have chosen a relatively high cut-off frequency of 500Hz or thereabouts. The effect of this is obvious, and you can hear the result in sound #4 ▶ . If you feel that this is a little extreme, you can adjust the cut-off frequency downward to attenuate the low frequencies to the degree you feel appropriate.

To complete the first part of the sound, we now need to make sound #4 repeat in a sensible fashion, so that the synthesised waves continue to crash against the shore. There are many ways to achieve this, but I'm going to do so by invoking Thor's step sequencer. It's not apparent from the screen shot (see figure 5) how this is set up, so here's the explanation:

The sequence has two steps and is running at 0.1Hz, so each step is in principle 10 seconds long. However, I have set the duration of the second step to 1/2, so this lasts just 5 seconds, and I have also set it up so that it does not send a trigger to the rest of Thor. I therefore have a sequence that last 15 seconds, the first 10 with the sound triggered (with a Gate duration of 75%), and the next 5 with only the delay and reverberation generated by the RV7000 present. (Note: If you play this patch without the reverb switched on, you will hear that the sound stops dead when the Gate is closed by the sequencer. The RV7000 is therefore an important component of the program, not just a reverb in the conventional sense.)

You can, of course, adjust all of these timings and parameters to taste, but I find that this to be a pleasing result, so let's view now hear the waves. ▶ You can almost hear the ocean draining back across the shingles!

Figure 5: Waves breaking
(Click to enlarge)

Adding more elements

Interesting though this sound is, it's not very realistic, even for an alien shoreline lying somewhere in the unfashionable arm of the Milky Way. (Oops... there I go again.) If you live in England, you'll know that there's no such thing as summer, and any experience of the seaside is accompanied by a biting wind that seeks to strip the flesh from your bones unless you wear a thick vest and a ski jacket. So the next stage is to generate a wind effect using the second signal path that Thor provides.

Let's switch off the signal path from the oscillators to Filter 1 by clicking on the "1" and "2" buttons within the filter itself, and then launch a second 24dB/octave low-pass filter in the Filter 2 slot. I have done this as shown in figure 6, with the filter's cut-off frequency set to around 600Hz, its resonance set to 100 or thereabouts, and a fairly low drive level. Routing the oscillators to this filter results in sound #6 ▶ , which is another example of tuned noise, not dissimilar to the timbres that I created in the previous tutorial.

Figure 6: Tuning the wind sound

Although the underlying timbre is appropriate, sound #6 is static (which wind is not) so I now need to modify the patch to create some movement. This is easy. Firstly, I use the modulation matrix to direct the output of LFO1 to the Filter 2 cut-off frequency. I don't want a cyclic sweep (which would be boring) so I choose one of the quasi-random waveforms. This stops the wind effect from cycling in synchronisation with the waves, so the composite sound doesn't repeat in an obvious fashion. Secondly, I use the matrix to direct the output of Filter 2 directly to Audio Out 1 so that the wind sound can be heard without interruption, no matter what else might be happening. The resulting patch is shown in figure 7, and the sound that it generates ▶ is now much more evocative.

(It's worth noting that we could generate this sound using a third noise oscillator in Band mode, with the tuning increased, and that we could then modulate this directly without invoking a second filter to create the wind effect. The benefit of my approach is that it leaves an oscillator slot free so that I could, if I wished, insert a different type of oscillator to increase the complexity of the sound or even to change its nature altogether. There are numerous ways to create this type of patch, and no 'wrong' ones.)

Figure 7: Adding a wind effect
(Click to enlarge)

So now we have waves crashing on a shoreline (courtesy of the signal routed through Filter 1) and wind whistling around us (courtesy of the signal routed through Filter 2). To create our alien soundscape we need only add these two elements together by reactivating the audio inputs to Filter 1. (See figure 8.) If you ever wanted to become a sound designer for the next incarnation of Star Trek, you could do worse than to start with sound #8 , which is just a short snippet of the endless crashing of the waves and howling of the wind that this patch generates.

Figure 8: Standing on alien shores
(Click to enlarge)

The possibilities are endless

Of course, there are many changes and embellishments that you make to this sound. For example, the rigorous tempo of the wave effect becomes tiring if you listen to it for too long, so you could add a degree of randomness to this. In addition, you could modulate the loudness. You could also connect the cut-off frequency and resonance of Filter 2 to a pair of knobs so that you could adjust the wind effect in interesting ways. And, as you would expect, there's a huge range of alternative timbres that you could produce by adjusting the other parameters of the sound. You could spend time making it more realistic (in the sense of standing on a weather-beaten cliff in Scotland) or develop it further in more experimental 'alien' ways. The possibilities are endless.

Text & Music by Gordon Reid

Thor demystified 7: The Phase Modulation Oscillator

DISCOVERING REASON Phase Modulation is a brilliant method of synthesis championed by Casio in the 1980s. It enables a digital (or hybrid analogue/digital) synth to generate an enormous diversity of sounds – ranging from traditional analogue to FM in character – using a minuscule amount of wave memory and, by modern standards, minimal processing power. This made it possible for the company to develop a family of polysynths that were cheaper than almost anything that had come before, but were more flexible than pure analogue synths costing many times more. For people who thought of the CZ series as the "poor man's DX7" and didn't rate them very highly, they proved to be surprisingly successful. For people who understood them fully and knew how good they could sound, they proved to be surprisingly unsuccessful. But what, exactly, was Phase Modulation? To answer this, we need to delve briefly into the realms of geometry. (I could explain it with pure maths, but you'll prefer the geometry, believe me.)

Bending a sine wave
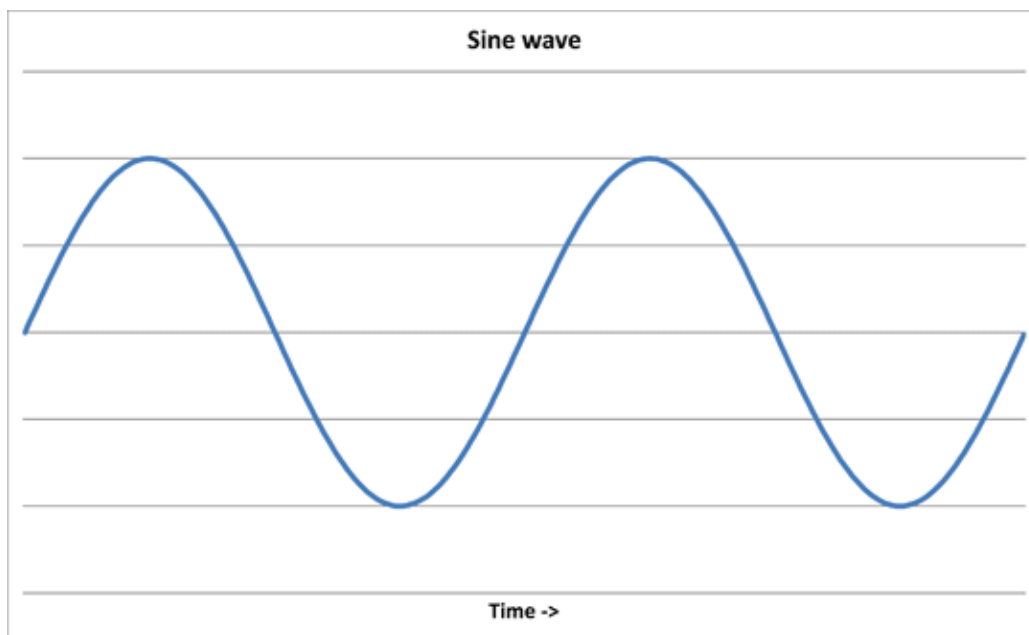
Figure 1: A sine wave

Take a look at the sine wave shown in figure 1. If you imagine sampling this at regular intervals along its length, it seems intuitively obvious that you could store these samples in a small ROM and later read them out in a steady stream to recreate the wave-

form. (Please, let's not get embroiled in silly arguments about digital waves looking and sounding like staircases, and stuff like that. They are all rubbish.) I have illustrated the sampled wave in figure 2.
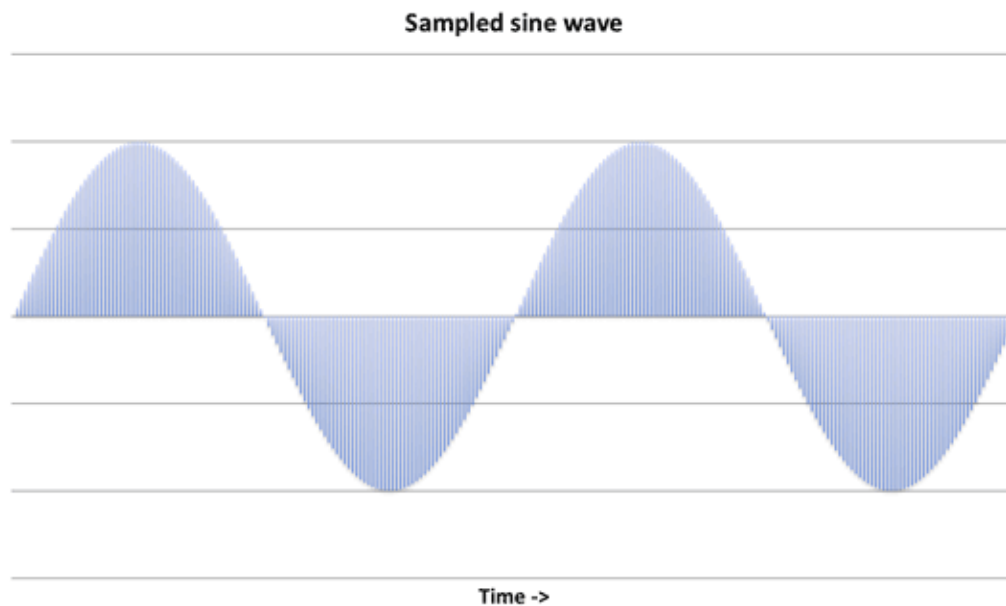
Figure 2: A sampled sine wave

Now let's imagine the situation where the samples are not read from the ROM at a constant rate. Imagine what would happen if the first batch of samples (shown in green in figure 3) were read back very quickly, and the highest value sample (shown in red) was then held for a long period. The black line in the figure shows the result of this.

Now I'll do the same for the next group of samples, reading these out very quickly until I reach the one with the lowest value, and I'll then hold this. (Figure 4) I'm sure that you can see where I'm going with this, but I'll complete the description. We just keep reading out the green samples very quickly, while pausing for an appropriate length of time on the samples with the maximum and minimum values. The result is a surprisingly accurate representation of a square wave (figure 5) that I created simply by altering the rate at which the system read the sine wave's samples. This is Phase Modulation!
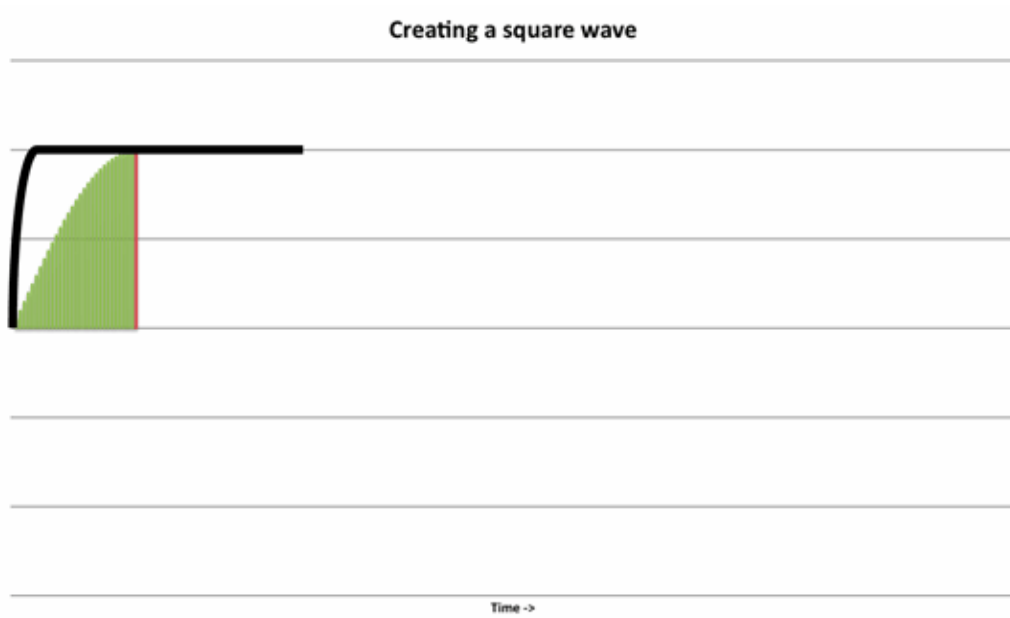
Figure 3: Reading out the first few samples at different rates.
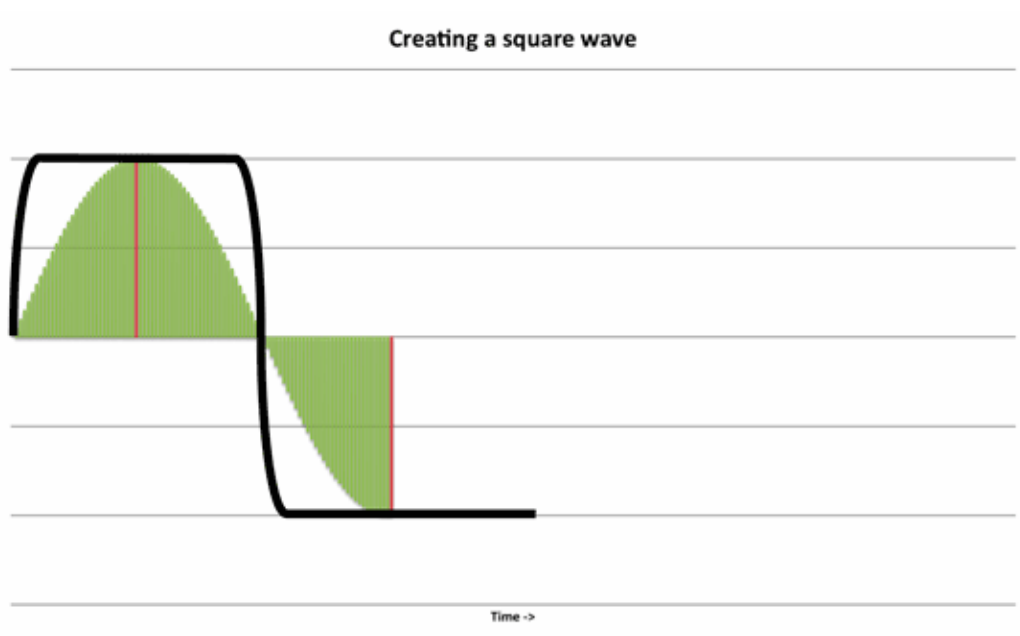
Figure 4: Reading out some more of the samples
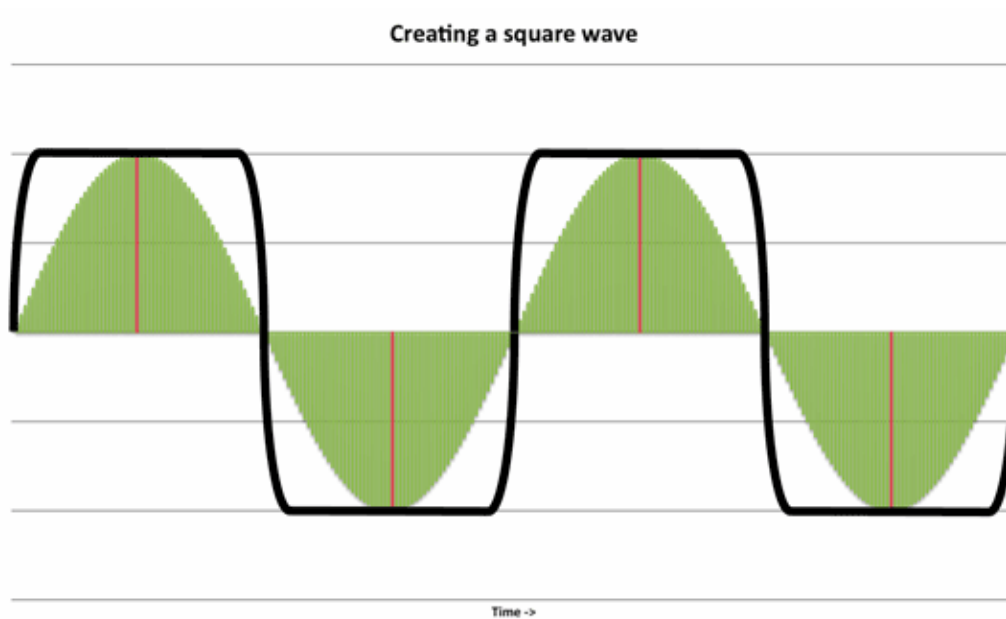
Figure 5: A Phase Modulation square wave

Creating some other waveforms

The black line in figure 5 is the waveform that you hear, not the one that describes the changes (or 'distortion') in the clock rate. But working this out is not difficult. When the clock rate is high, the samples are read out very quickly (which creates the almost vertical lines in the figure) and when the clock rate is almost zero, the same value is held for a long time (which is what happens at the top and bottom of the square waveform). So we can draw the changes in clock rate that generate an audio square wave as shown in figure 6.
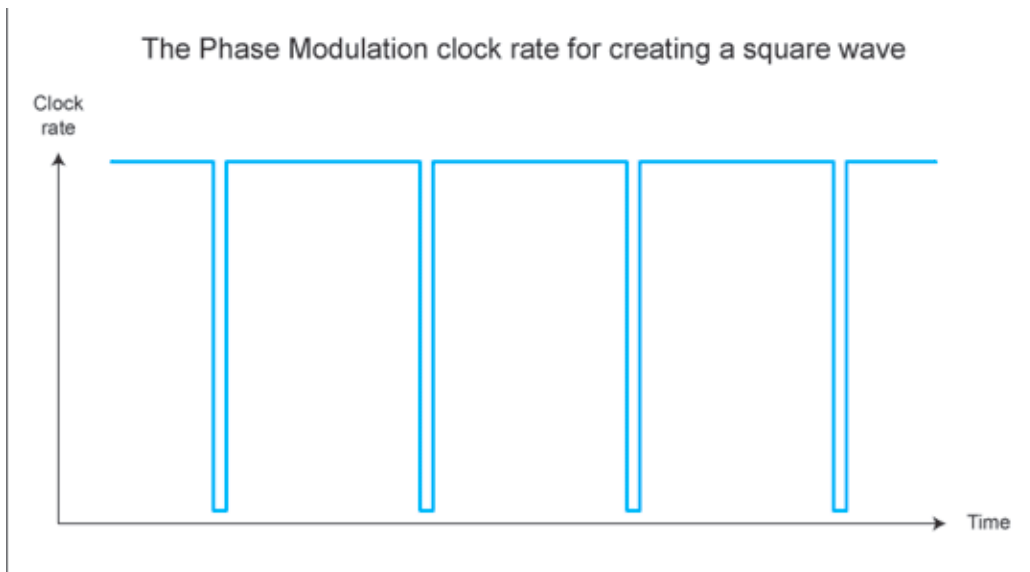
Figure 6: The clock speed changes that generate the square wave in figure 5

Of course, there are many other clock rate distortions, and Casio chose eight shapes that created the eight waveforms available on the CZ synthesisers. I have shown another in figure 7, which would generate something akin to a sawtooth wave.
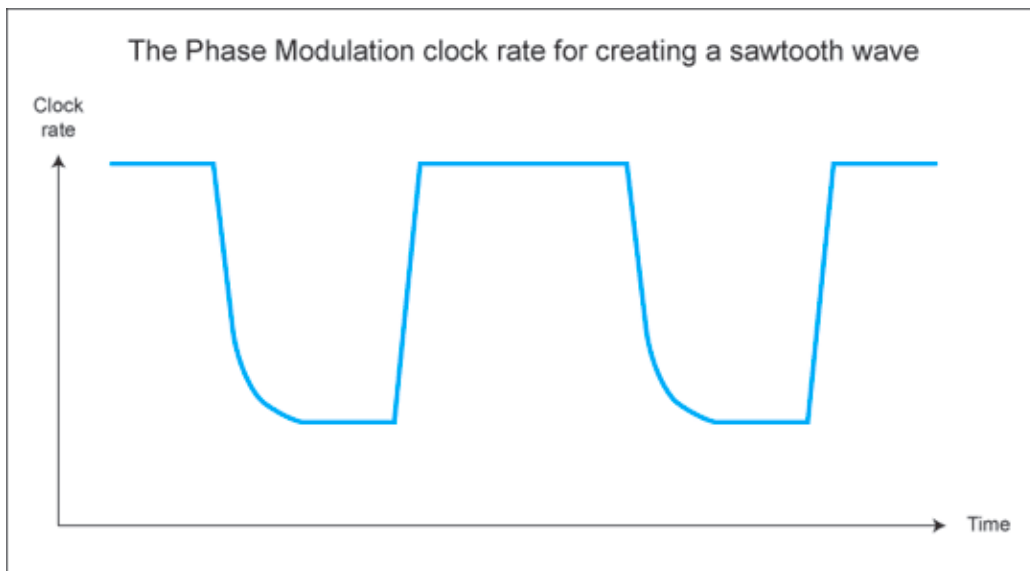
Figure 7: The clock speed changes that would generate a sawtooth wave

Getting Down and Dirty with Delay

In our tireless efforts to build new and exciting Reason devices from principal building blocks – because A) it's fun, B) it showcases the modularity and flexibility of Reason, and C) we played with Lego as kids, and never really grew up – in this month's Discovering we venture into the realm of complex delay effects. This is a category of FX for which there is no ready-made device in Reason's toolbox; there's the DDL-1 for your most basic delay needs, and there's the RV7000 with its multitap and echo algorithms, but there is no Ultimate MegaDelay™ with six dozen knobs and a coffee machine.

Our mission today, then, will be to build our own in the garage. We'll be using the trusty old DDL-1 digital delay as a starting point for these patches, and on that foundation we'll add some flavor and spice to give the delay a unique texture. We're presenting two patches; the first is a lo-fi delay from the olden days, and the second is a more complex and spaced out creation from an unspecified time in the future.

Lo-Fi Delay

As the name may suggest, this patch attempts to emulate a vintage tape delay sound with a bit of compression, distortion and degradation thrown in. To achieve the kind of squelchy effect we need, we turn to the Scream 4 and its Tape algorithm – Reason's go-to tool for all things warm and fuzzy.

At first glance, the front panel of this setup looks fairly straightforward, and you'll probably assume that it's a stereo delay with two pairs of delay and Scream units, one for each channel. But it's not a parallel setup, it's a serial one. Since we're going for a tape-style delay here, the signal needs to deteriorate

slightly for each cycle, and we achieve this by doing the following: The signal is passed through the first DDL-1 and the first Scream 4, and comes out slightly squelched. The processed signal is then passed through the second DDL-1, which is set to a 50/50 mix of dry and wet signal. Dry, so that the signal from the first delay is heard, and wet, so that the signal – which has already been processed with the Tape algorithm once – is "tape-ified" again by the second Scream 4. On top of that, we use some naughty wiring to create a feedback loop where a wee bit of signal is fed back to the first DDL-1 unit, adding a phased quality to the delay as it fades and dissolves. Confusing? It will make a whole lot more sense when you hear the result.
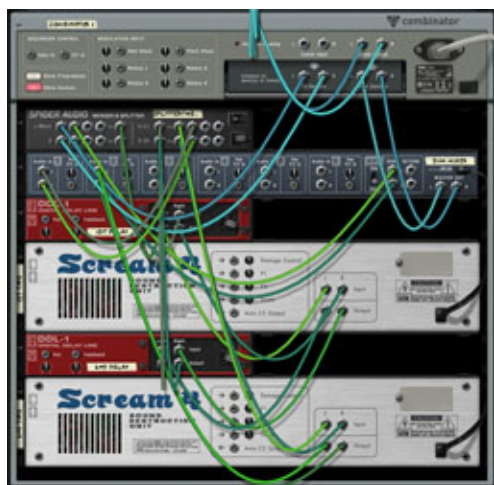
Shopping list

1x Combinator
1x 6:2 Line Mixer
1x Spider Audio
2x DDL-1 Delay
2x Scream 4

Routing

The Combinator's To Devices output is connected to the Spider Audio's Merge Input, rather than straight to the FX device chain. This is so that the external signal and the processed signal that comes back from the FX chain can be merged and fed into the chain simultaneously (the feedback loop we mentioned above). The Spider Audio's Merge Output is routed to the first DDL-1 unit, from which the signal is routed to the first Scream 4. From there, it continues to the second DDL-1 unit's input. Here, it's time to split the signal in two – which is why we connect it to the Spider Audio's Split Input – because it needs to go both to the second Scream 4 (for another round of squelching) and the 6:2 Line Mixer (so that it can be sent back into the chain, creating a feedback loop). And finally, from the second Scream 4's output to the mixer. Done.

Controls

Any delay unit will of course need time and feedback controls, so we'll use two of the Combinator's rotary controls for that. We'll also add a third rotary control called "Extra Squelch", which warrants an explanation: What this knob will do is control the Aux

Send parameter on channel 2 of the Line Mixer. This determines the amount of signal that's fed into the feedback loop. We've set the range to 0-21, this is about as far as you can go before the feedback loop turns into unstoppable mayhem.

Finally, we thought it might be a good idea to have a switch for tempo sync. Since it's not particularly natural for an analog tape delay to have rock solid tempo sync, it makes sense to have the option to switch to millisecond mode.

Here's the patch itself, and here's a couple of sound examples: The first one illustrates the classic tape delay effect put to somewhat exaggerated use in a reggae context, on the snare drum and the voice sample: Elevator reggae   .

The second example goes bonkers with the feedback loop and uses heavy automation on the delay time parameter to create the impression of an FM radio on a sugar rush: Radiohyperactive

Complex Delay

Our second tinkertoy is a stereo delay with an optional reverse effect, courtesy of the RV7000's Reverse algorithm. Also featured are LFO-controlled band shifters created using a couple of BV512 Vocoders in EQ mode. The result is a trippy, ethereal delay effect that pulls ghostly shapes of the original sound backwards into the mix, where they bounce around in the clouds and leave jet vapor trails that slowly dissolve. This effect can fill out a mix nicely and create a dreamy atmosphere; just throw it a few sparse notes and let the ripple effect take it from there.

Shopping list

1x Combinator
1x 6:2 Line Mixer
2x Spider Audio
2x DDL-1 Delay
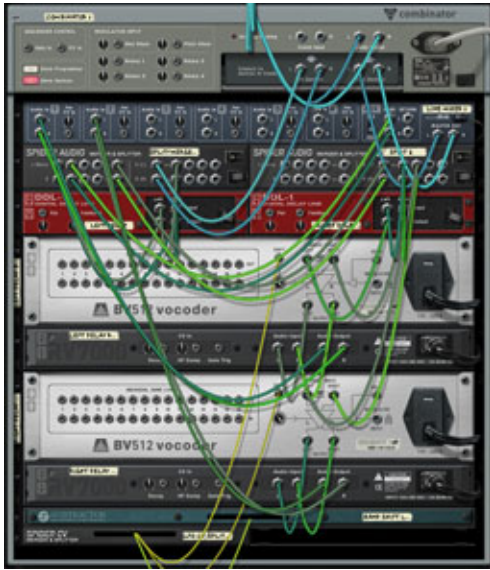
2x RV7000 Reverb
2x BV512 Vocoder
1x Subtractor
1x Spider CV

Routing

Ooookay, brace yourselves as we attempt to follow the signal path here. The To Devices output goes straight to the first DDL-1 delay (Left). From there, it goes to the Merge Input on Spider Audio 1 (along with another signal, but more on that later). The Merge Output on Spider Audio 1 is then connected to the Split Input on Spider Audio 2. Here, the path breaks in two. One leads to the second DDL-1 delay (Right), and from there to Vocoder 1, and on to RV7000 1 (so far, disregarding the Spiders, the path is Delay L > Delay R > Vocoder 1 > Reverse Delay L). The other leads to Vocoder 2, to RV7000 2 (the path here is Delay L > Vocoder 2 > Reverse Delay R). The signal from RV7000 1 is routed to channel 1 on the Line Mixer, the signal from RV7000 2 is routed to channel 2, and the Line Mixer's Master Out goes to the From Devices input on the Combinator. Finally, the Aux Send on the mixer is connected to the Merge Input on Spider Audio 1, where it is merged with the signal from the first DDL-1 delay (the first stop for the original signal) and tossed right back into the FX chain one more time. One last thing: We want to use LFO1 on the Subtractor to control the Shift parameter on both the Vocoders, so we add a Spider CV for splitting the signal. And that's a wrap.

Controls

We want to be able to control delay time and feedback, as usual, but since this is a stereo delay we'll want to dedicate one rotary control for each side. We set Rotary 1 to control the DelayTime parameter on the first DDL-1 (Left), and Rotary 2 to control the same parameter on the second DDL-1 (Right). Rotary 3 controls the Feedback parameter on both DDL-1 units. That leaves us with Rotary 4, which we'll use to control the LFO1 speed on the Subtractor. It'd be a shame to leave the four buttons on the Combinator unused, so we'll use those to toggle the Reverse delay effect and the Band shifter effect on/off. This is done by linking each of the buttons to the "Enabled" parameter on each of the devices (the two RV7000's and the two Vocoders) and setting the Min value to 2 (Bypass) and the Max value to 1 (On).

You can download the patch here. And here's a quick demo of the Complex Delay in action: Floatation.   Listen to it once, then disable the Complex Delay by switching it off or silencing Aux Return 2 on the main mixer, and listen again. You'll notice that pretty much all sense of depth and width in the arrangement disappeared once the delay was removed from the equation.

Text by Fredrik Hägglund, patches by James Bernard

Thor demystified 8: More on Phase Modulation Synthesis

In this tutorial, I'm going to show you how you can imitate analogue filter resonance using Phase Modulation synthesis, and offer two examples – a bass patch and a lead synth sound – that illustrate how you can use this.

Here's a patch that I call "Grod's Reso-Bass". I'm sure that you'll recognise this type of sound, which was inspired by some old – but wonderful – analogue bass pedals that I used in the 1970s. Given that this sample was generated by Thor, it isn't analogue of course, but it's a great 'virtual analogue' sound, isn't it? Don't you just love that deep, resonant filter sweep? Except… this isn't a virtual analogue patch, and that isn't a filter sweep. Four modules generated this, and not one of them is a filter. The patch comprises just two oscillators and two envelope generators, and it's another example of the brilliant Phase Modulation system developed by Casio in the 1980s, which has been almost universally (and unfairly) derided ever since.

We'll start recreating this patch by making sure that everything except the Gain and Master Volume in Thor is set to 'off' or to zero (as appropriate), by setting the keyboard mode to Mono Retrig, and then by inserting a Phase Mod Osc in the Osc1 position. When you have done this, select the sawtooth waveform from the eight possible waves in the First slot within the oscillator, tune the pitch down to Oct=3, and make sure that the PM knob is set to zero. Next, switch on the Amp Env and push its Sustain level to maximum. If you now play a few notes you'll hear (as I explained in tutorial #7) a sine wave, not a sawtooth wave. What's more, you'll obtain some nasty clicks and thunks at the start and end of each note. These are not a bug… they're generated by the superfast envelopes within Thor. Shown in figure 1, this is not a pleasant patch.

Figure 1: A rather unpleasant bass sound

We can refine this by modifying the Amp Env contour to eliminate the clicks, to generate a pleasing overall shape for the sound, and to control the amount of phase modulation as each note progresses. We do this by selecting suitable ADSR values and routing the envelope to the Osc1 PM Amt parameter in the modulation matrix. I find that a modulation Amount of 75 works quite nicely. (See figure 2.) This is a much nicer patch.

Figure 2: Adding some interest to the basic bass sound

If you now step through the waves offered by the First slot, you'll find that the last three are rather unusual waveforms described in the Thor manual as…

- Saw x Sine
- Sine x Sine
- Sine x Pulse

… and by Casio as …

- Resonance I (Saw-tooth)
- Resonance II (Triangular)
- Resonance III (Trapezoid)

These were described by Casio as "resonant" because, when the PM amount is swept, each of these waveforms simulates the sound of a filter with the resonance (or "emphasis") turned up to a high value. Some people feel that they don't do this very convincingly but, to some extent, that's analogue snobbery. The effect can be quite realistic and, even when it's not, it's very useful because it generates a new family of analogue-like sounds that are subtly different from anything you can obtain from a traditional analogue synth.

Let's investigate these waves by increasing the Amount in the modulation matrix to 100 and placing the Resonance I waveform in the First slot. If you now play, you'll hear sound #3 , which exhibits the distinctive character of the swept, resonant filter. Sounds #4 and #5 demonstrate the same patch with the Resonance II and Resonance III waveforms inserted.

Lost and found: Hidden gems in Reason 4

Have you ever found yourself using a recent version of a software product like it was the same old version you used ten years ago? When you work so much with an application that the workflow migrates from your conscious mind to your muscle memory, it automatically becomes more difficult to pick up new tricks, and instead you will follow the path of least resistance and use the old method of doing things.

Over the years, Reason has seen many additions of new features, big and small. In this article we're going to take a closer look at some of the small and sometimes overlooked items that were overshadowed by new instruments and other major features that stole the spotlight.

The Inspector strip

The context-sensitive inspector strip on the sequencer's toolbar is great for both micro- and macro-editing recorded events. In Arrange mode, selecting a single clip will bring up two fields in the inspector: Position and Length. You can then enter numerical values or use the up/down buttons to change the position or length of the clip.

If you select multiple clips, however, two additional items will appear: The Match Value buttons (represented by red equal signs to the right of their respective parameter fields), which allow you to match the position or length of the first selected clip in the timeline.

The inspector serves a similar purpose when you edit automation data; by double-clicking on an Automation clip and selecting one or multiple automation points, you'll tell the inspector to bring up the position value and the automation value of these events.

The Edit mode is where the Inspector really shines, as it allows you to macro-edit note parameters such as Position, Length, Note (pitch) and Velocity. Here are a couple of situations where the inspector comes in handy in Edit mode:

Let's say you've recorded some chords and quantized the clip, but you also want all the notes to be of equal length because you didn't release the keys with perfect timing. First, make sure the first note has the correct length (tweak the note in the inspector to achieve this), and then select all notes inside the clip.
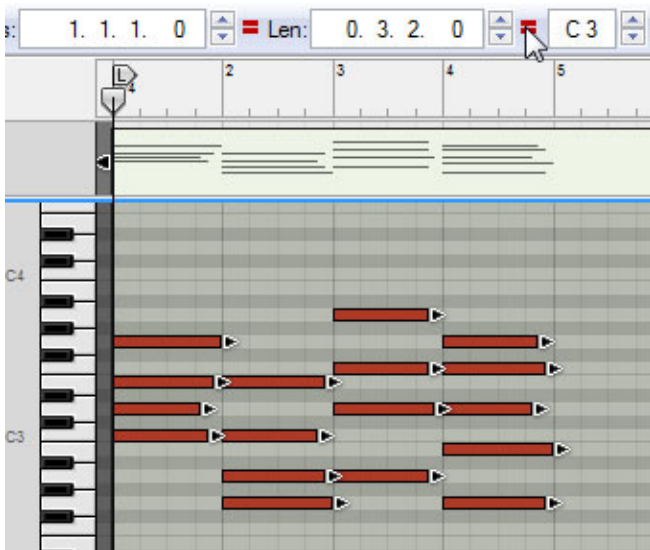
Then, simply click the equal sign to the right of the Length field, and all the notes will be adjusted accordingly.
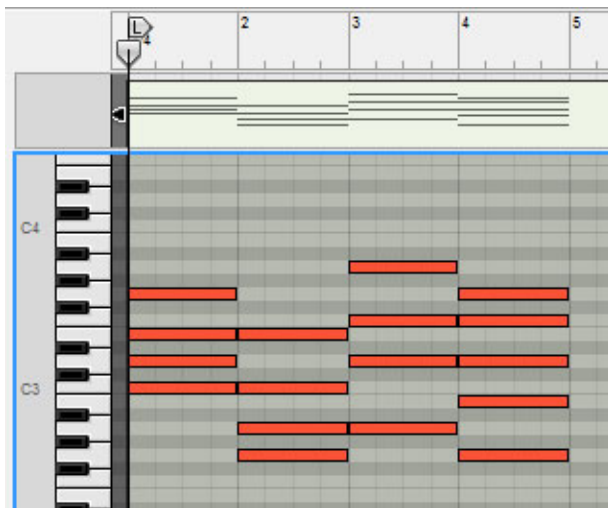
The notes in this example were already quantized, but had they not been, you would've been able to use the inspector to quantize them one by one by selecting each chord and clicking the Match Value button for Position. This will move all notes in the chord to the same starting point.

Here's another case scenario: Let's say you're working on a drum track, you've recorded some notes and you want all of them to have the same length and velocity. Once again, make sure the first note in the timeline has the desired properties by adjusting these in the inspector, and then select all notes you want to apply these properties to.
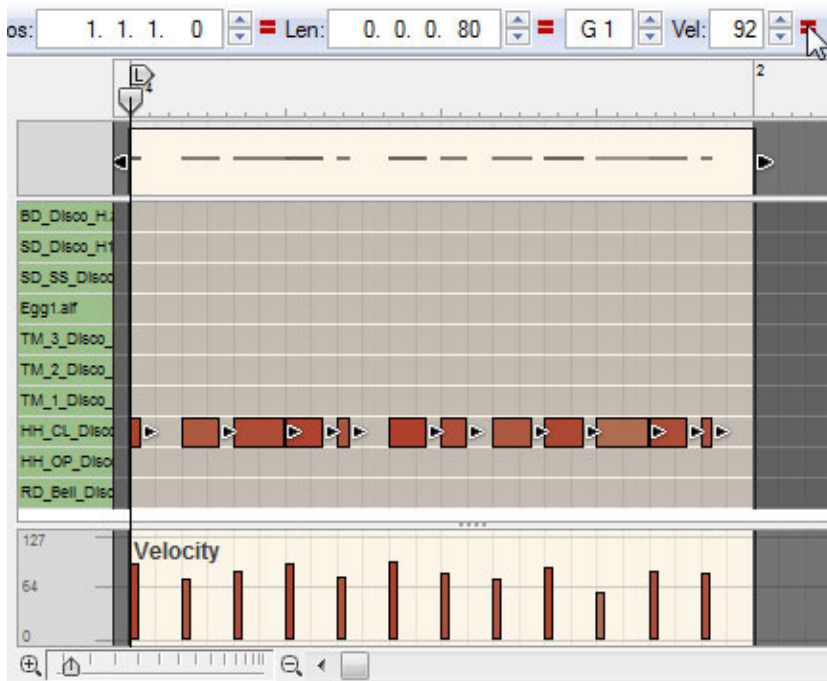
Now, click the two Match Value buttons and you'll have a neat row of notes with identical duration and velocity.
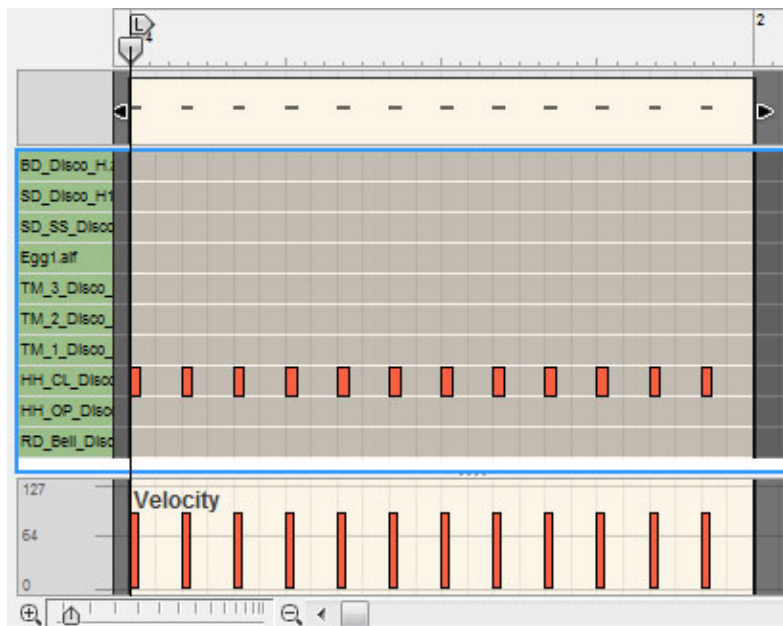
The Tools Tab

In the space now occupied by the Inspector, there used to be quantization tools in older versions of Reason. These were moved to the Tools tab in the Tool window, your one stop supply of note-tweaking tools. Here you can quantize – the old way, not the ReGroove Mixer way – as well as transform the pitch, velocity, length, tempo and order of recorded events. We'll go over these tools from top to bottom.
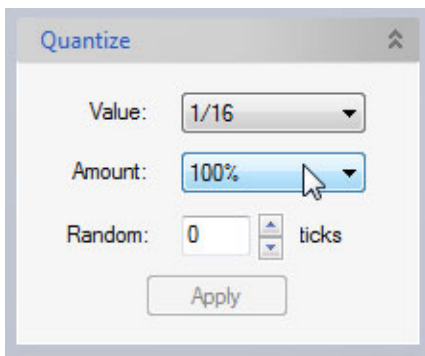
The **Quantize** tools include an Amount percentage setting. The value is 100% by default, meaning that the selected events will be hard-quantized to the exact value currently selected on the Value drop-down menu. If the value is, say, 50%, notes will only be moved halfway to their closest quantize value positions. This is useful if you want to tighten up a sloppy take while still retaining some human feel. If for some reason you want more human feel than the original recording can offer, you can add this artificially by using the Random setting.
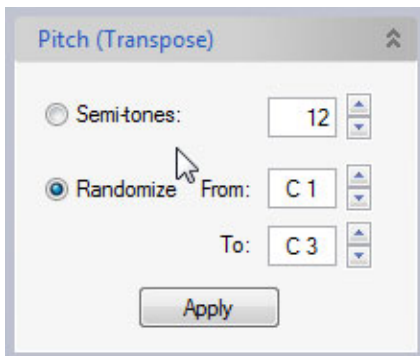
The **Pitch** tools feature a basic transpose function plus a randomization tool. When Randomize is active, the pitch of the selected notes will be randomized within the specified range. This may not be of much use on tonal instruments, but it's very useful for REX loops since randomizing notes on a Dr.REX track will result in the loop segments being completely rearranged.
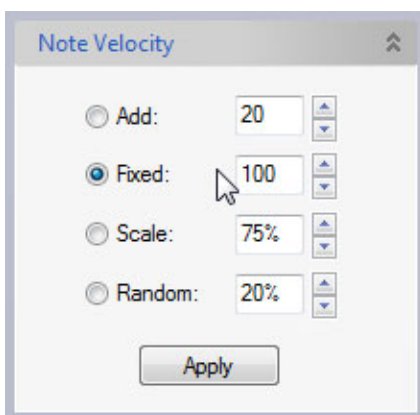
The **Note Velocity** tools are: Add, Fixed, Scale and Random, and their functions are pretty self-explanatory. The Add field lets you add (or subtract) an integer value to (or from) the velocity on all selected notes. A value of 20 will change velocity 92 to velocity 112, and velocity 115 to 127 (since 127 is the maximum value). The Fixed tool will apply a fixed velocity value to all selected notes. The Scale tool will change velocity values according to a percentage value, as follows: If Scale = 100%, nothing will

be changed. If Scale = 50%, velocity 110 will become 55, 26 will become 13, etc. The Random tool uses a percentage value to randomize velocity values. For example, with a Random value of 10%, velocity 100 will be changed to anything between 90 and 110.
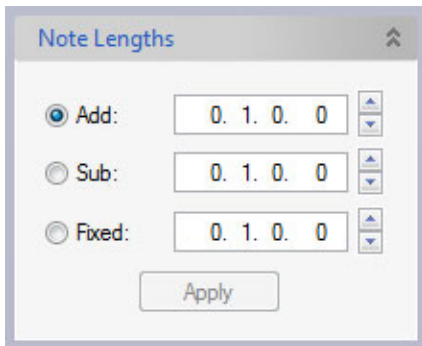
The **Note Lengths** tools let you extend or shorten note lengths, or make them equal. When Add is selected, notes will be extended by the amount specified in in the Add field. The Sub (subtract) tool has the opposite effect. The Fixed tool will simply apply the note length specified in the Fixed field to all selected notes.

Legato adjustments lets you fill the gaps between notes by tying them together, even to the point where they overlap. Side By Side (Abut) will extend a note to the exact point where the next note begins, Overlap will extend notes beyond that point, and Gap By will truncate notes in order to leave gaps of equal length between them. One situation where legato adjustment comes in handy is when you have rearranged the notes for a REX loop so that the slices will play back in a different order (this can be done with the Alter Notes function, more on this below). When you use the Alter Notes function on a Dr.REX track, the notes will be shuffled around, but since REX notes are often of different lengths, this will leave gaps between some notes, while other notes overlap. For example, this…
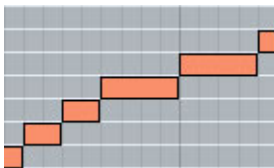
…may end up looking like this:

Notice the overlapping notes in the beginning and the gap in the middle. After applying Legato Adjustments > Side By Side (Abut), the result will be…
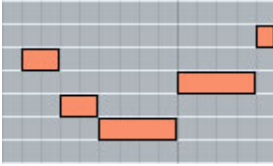
…which eliminates the gaps and the overlapping.

Scale Tempo does exactly what it says, although it has nothing to do with the global tempo or automation thereof. What it does, rather, is stretch out or compress a sequence of notes. This is useful when you want to mix double-tempo or half-tempo elements (such as drum loops) with "normal" tempo elements.
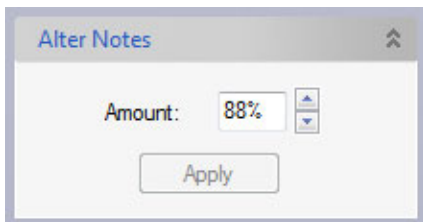
Alter Notes lets you randomly rearrange the positions of recorded notes. This is a more musical function than Pitch randomization, since random pitch will lead to atonal results more often than not. It works best on drum loops, drums and monophonic parts such as basslines and arpeggios.
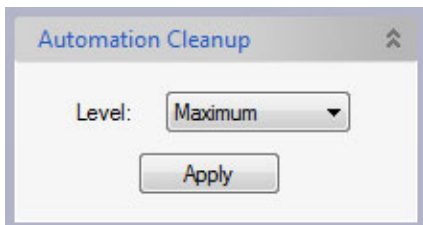
Finally, there's **Automation Cleanup**. When you record automation, you will often end up with large clusters of automation points out of which many are redundant, since automation in Reason 4 is vector-based. For example, you may have an automation clip that looks like this:

As you can see, many of these automation points are located along straight lines between other automation points, which renders them useless. The Automation Cleanup function can spot the redundant points and delete them automatically, so that the above clip will look like this:

This was achieved with the Maximum setting. If you're worried about the risk of nuances getting lost you should go for a more moderate setting.

By Fredrik Hägglund

Thor demystified 9: An introduction to FM Synthesis - part 1

DISCOVERING REASON Frequency Modulation (FM) has become the bogeyman of synthesis. Whereas, in the 1960s, people quickly grasped the concepts of these new-fangled oscillators, filters and contour thingies, the second generation of players shied away from the concepts of FM, to the extent that most FM synths were used for little more than their presets and the professionally programmed sounds that you could buy for them. Even today, if you look closely at Thor's refills, you'll find very few patches based upon its FM Pair Oscillator. This is a great shame, because FM is a very elegant system capable of remarkable feats of sound generation. So, this time, I'm going to introduce you to the principles of FM, and show you how to create what may well be your first FM sound.

Let's start by considering what happens when you change the frequency of a pure tone (which, in synthesiser parlance, is called a sine wave). If you increase it steadily, you hear the pitch glide upward until it exceeds the bandwidth of your system, or the limit of your hearing, and becomes inaudible. If you lower it steadily, the pitch decreases until the sound drops below the limit of your hearing. But if you vary it up and down slowly (using another sine wave as a modulator) you hear a very common musical effect: vibrato.

Now let's increase the speed of the up/down modulation. At first, you hear a faster vibrato, but as the frequency of the modulator moves into the audio range, the vibrato effect disappears and a new tone replaces the original, the nature of which is determined by the relationship between the frequencies of the original sine wave (the "carrier") and the modulating sine wave (the "modulator"), and the amplitude of the modulator. To prove this would mean delving into some rather nasty equations, but don't panic... I'm not going to do this. Instead, I'm going to present you with the results that you would obtain if you had a degree in mathematics.

If you refer back to the second tutorial in this series, you'll find that, if you modulate the amplitude of one oscillator (the carrier) of frequency ω_c (the character ω means "the frequency" so ω_c means "the frequency of the carrier") using another oscillator (the modulator) of frequency ω_m, the result is two new signals (called side-bands) with frequencies $\omega_c + \omega_m$ (the sum) and $\omega_c - \omega_m$ (the difference). So, if you increase ω_m while ω_c remains constant, the frequency of the sum increases, while the frequency of the difference decreases. We can express this in a very neat way in the following form...

... which means that "the frequencies of the side-bands are equal to the frequency of the carrier plus or minus the frequency of the modulator".

Now let's turn our attention to frequency – rather than amplitude - modulation. If you work through the maths, you'll find that this also creates side-bands, but instead of generating just two of them, you obtain a series expressed as...

... where the symbol "n" is an integer number: 0, 1, 2, 3, 4... and so on.

To put this into English, each side-band lies at a frequency equal to the carrier frequency plus or minus an integer multiple of the modulator frequency. Since the value of "n" can be 0, 1, 2... or hundreds, or thousands, or millions ... then, in theory, FM produces an infinite series of side-bands. This, as you might imagine, poses certain problems, as does the issue of "negative frequencies" when $n.\omega m$ is greater than ωc, but I'm going to ignore both of these issues until the next tutorial.

OK... we now know the frequencies at which the side-bands are generated, but what about their amplitudes? What is the spectral content of an FM-generated signal? Intuitively, it would seem sensible to suppose that, as you increase the depth of the modulation, the output signal would become more complex, and this indeed proves to be the case. Nevertheless, the relationship between the modulation depth and the output spectrum is not quite as simple as one might hope. It's described by a thing called the modulation index, denoted by the character "β", which is the ratio of the maximum change in the carrier frequency divided by the modulation frequency. As β changes, side-bands can be introduced, diminish, disappear altogether, or even reappear with inverted phase. But the overall picture is as described; the greater the amplitude of the modulation, the more complex the output signal becomes.

To illustrate this, figure 1 shows a case where β is low: in the region of 0.1 or less. The only significant side-bands are those closest to the Carrier frequency, and the result is similar to what you might obtain using Amplitude Modulation. But if β is significantly higher - say, in the region of 5 - we obtain a much broader series of side-bands, and a much more complex spectrum results. (Figure 2.)
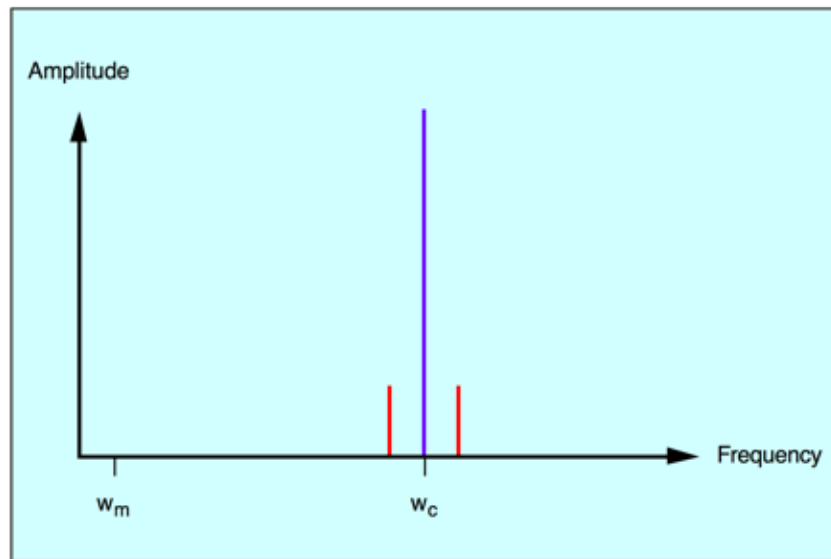
Figure 1: FM sidebands with low Modulation Index
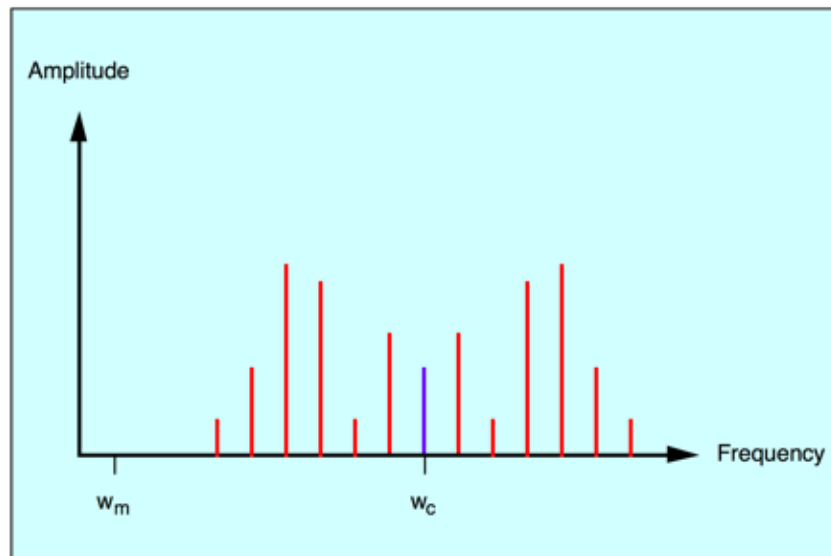
Figure 2: FM of the same signals when the Modulation Index is increased

The nature of the Modulation Index has an interesting consequence: if you want the tone of an FM-generated sound to be consistent as you play up and down the keyboard, you need all three of the modulator, the carrier, and the amplitude of the modulator to track by 100%. If you make the modulator amplitude track by more or less than this, the resulting sound will differ in tone from note to note.

We can now state the simple rules of FM synthesis as follows:

Rule #1:

The frequency of the modulator determines the spacing of the side bands in an FM-generated signal.

Rule #2:

The number of audible side-bands is determined by the modulation index, which can be described in general terms as the amplitude of the modulator divided by the frequency of the modulator.

Programming an FM sound

Inevitably, there are complications to this picture, but if you keep these rules in mind, you can programme useful FM sounds. Thor allows you to do so using a dedicated oscillator called an FM Pair Osc. This has two "operators" – a modulator (called Mod) which is permanently connected to the frequency modulation input of a carrier (happily, called Carrier). This is the simplest possible version of FM, and the way in which the operators are connected (called an Algorithm) is represented by figure 3.

Figure 3: The algorithm of an FM Pair Osc

Let's start with a basic configuration of Thor, with a single FM Pair Osc inserted into the Osc3 slot. [It's usual practice to place the audio output operator(s) at the bottom of the algorithm.] Set the Carrier and Mod values to "1", set the FM amount to zero and create a simple organ-shaped Amp Env with maximum Sustain and a short Attack and Release to eliminate the clicks that would otherwise occur. (Figure 4.) If you now play a note, you'll hear nothing but a sine wave... the aforementioned 'pure tone' that contains nothing but the fundamental of the note.

Figure 4: Creating a sine wave by playing an un-modulated carrier
(Click to enlarge)

Now let's introduce some frequency modulation. The FM knob in the FM Pair Osc determines the amount by which the Mod affects the Carrier, so if you turn this clockwise you'll hear the tone change. Sound #2 was generated with the FM knob set to 12 o'clock, with everything else unchanged. Figure 5 shows the waveform of this sound, and you can see clearly how the sine wave generated by the carrier is being pitch-shifted (i.e. stretched and compressed) by the modulator.
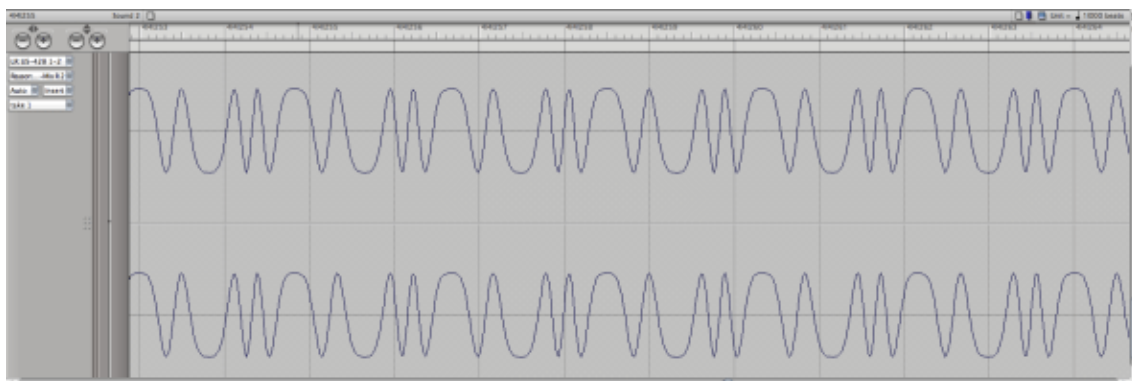
Figure 5: A clear visualisation of FM
(Click to enlarge)

As you can hear, sound #2 is brighter than sound #1, so this tells us how we can programme FM sounds with a dynamic spectral content. Figure 6 shows a patch in which the Osc3 FM amount is being swept by the Mod Env generator. The contour has no Attack, but a moderate Decay that sweeps the amount of modulation from its maximum down to zero. Sound #3   demonstrates the effect that this creates.

Figure 6: Sweeping the amount of frequency modulation
(Click to enlarge)

This sounds to me like a very bad approximation to the tonal changes that occur during the course of a note played on an electro-mechanical piano. This shouldn't surprise us; FM synthesis is famous for its so-called "DX piano" patches. The most popular of these used six operators but, remarkably, I'm now going to emulate these sounds using nothing more than a single FM Pair Osc!

The next stage is to adjust the Mod Env and the Amp Env in figure 6 to shape the changes in tone and volume in ways that are appropriate for an electric piano sound. I found that a Mod Env Decay of around a second is about right, with an Amp Env Decay of around four seconds. Why is the Mod Env contour shorter than the Amp Env's? Well... think about the sound of a real piano or electric piano. When the hammer hits the string (or tine or reed) the sound contains a lot of high-frequency components, so it is

initially bright, but the tone decays quickly to something more mellow, long before the note ends. The Mod Env is controlling the brightness, and the Amp Env is controlling the duration of the note, so all should now make sense.

Unfortunately, the sound that this produces is even more unrealistic than before. Sound #4   demonstrates that, even when played in a piano-like fashion, the result is horrendous. Nonetheless, we're on the right track. The problem is that (a) the amount of frequency modulation is too great, and (b) the patch contains no parameters that affect the tone as you play across the keyboard or as you change the dynamics of how you play.

Sorting out the first problem is easy. Reduce the FM modulation amount in the modulation matrix from 100 to around 40. Sound #5   shows that this is a dramatic improvement, and you may even be happy to use this patch as it is. However, we're currently in the territory of early 1970s non-sensitive electric pianos such as the Crumar Roadrunner and RMI Electrapiano, so I now need to look at how we can introduce timbral changes and dynamics.

The first thing to do is to add a modicum of velocity sensitivity to the tone by directing the Key Vel to the OSC3 FM Amt. This means that when you play a note harder (or, to be precise, with greater velocity) the tone is brighter than when you play it more softly.
   Next, we can imitate a very important attribute of plucked and hammered instruments... that notes become shorter as they become higher pitched. We do this by adding another line in the modulation matrix to shorten the decays of the Mod Env and Amp Env as you play up the keyboard. Sound #7   , which was generated by the patch in figure 7, is now starting to show promise.

Figure 7: Heading toward an FM e-piano sound
(Click to enlarge)

One of the things that makes this patch a tad unrealistic is that the notes at the top of the scale are too bright. Electric pianos have limited bandwidth and, because they dissipate their high frequency energy extremely quickly, the tines (or reeds) at the top end tend to be duller than you might otherwise expect. We can emulate this by adding another line in the modulation matrix that reduces the OSC3 FM Amount as you play up the keyboard.

Next, I want to make the sound respond to the sustain pedal in a realistic way, so I have added yet another line that increases the length of the Amp Env Decay and Release when I press the pedal. This then leads to note stealing so we need to increase the polyphony to its maximum of 32 notes. We now obtain figure 8 and sound #8.

Figure 8: Adding key-scaling and sustain
(Click to enlarge)

We're nearly there, but the loudness of the sound is still insensitive to velocity, and we're missing a very important effect that is crucial to the best e-piano sounds: panning tremolo. We can sort both of these out in the Amp section. First, we turn the Amp Vel knob fully clockwise so that MIDI Velocity affects the loudness of each note. Finally, we add yet another line to the modulation matrix that directs LFO1 to the pan of the amplifier. I find that an LFO rate of around 4Hz and a depth of a little less than maximum is rather pleasing. The resulting patch is shown in figure 9, and you can hear it in sound #9.

Figure 9: The final 2-op FM electric piano patch
(Click to enlarge)

This is another remarkable result, and it demonstrates what can be done with just two operators. However, I'm now going to add a second FM Pair Osc to reinforce the fundamental of the sound. I'm not going to apply any FM modulation in this pair; the waveform it produces will simply be a sine wave at the fundamental frequency, somewhat quieter than the output from Osc3, but tuned upward by a few cents to create a slight thickening of the overall sound. The final patch, now with three operators, is shown in figure 10, and you can hear it in sound #10.

Figure 10: 3-op FM electric piano
(Click to enlarge)

Before finishing, let's look at the algorithm used in figure 10, which I have drawn as figure 11. As you can see, this is beginning to look very much like the graphics that you see on the control panels of Yamaha's DX-series synths, and it leads us into a whole new area of FM synthesis; the use of multiple operators within a patch. But that will have to wait until next time. Until then...

Figure 11: The 3-op algorithm used in figure 10

Thor demystified 10:
An introduction to FM Synthesis - part 2

"More than just DX pianos"

In the last tutorial, I introduced the concept of FM algorithms, the ways in which FM operators can be connected together to create sounds. A Thor FM Pair Osc comprises two operators - a modulator that is permanently connected to the FM input of a carrier - and we can represent this as shown in figure 1.

Figure 1: The algorithm of a Thor FM Pair Osc

Of course, Thor has three oscillator slots, and you can place an FM Pair in each of these so, by default, the standard algorithm offered by Thor is as shown in figure 2: three pairs that can be mixed together, but which don't - for the moment - interact in any other way. There are many uses for this algorithm, which can be used to create very interesting string ensemble and organ patches, among others. But I fancy stepping beyond these, so I'm going to demonstrate this algorithm by showing you how to create an evolving pad that has a rich, analogue flavour.
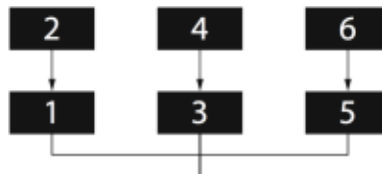
Figure 2: Placing a Thor FM Pair Osc into each of the oscillator slots

I know what I like...

Let's start by inserting an FM Pair Osc into each of the three slots in Thor. However, instead of leaving all the Carrier and Modulator frequencies at "1" as we did last time, we're going to create a more complex tone by setting them to "3/1", "1/3" and "7/1" respectively, but with the third oscillator tuned down by two octaves. (See figure 3.)

"What do these numbers mean?", I hear you ask. Happily, the answer is simple: they are multiples of the fundamental frequency of the note that you play. Let's imagine that you play a note with a fundamental pitch of 100Hz. Selecting "1" for both the carrier and the modulator means that the FM Pair will produce a carrier of 100Hz and a modulator of 100Hz, resulting in $\omega c + n.\omega m$ side-bands of 200Hz, 300Hz, 400Hz... and so on, and $\omega c - n.\omega m$ side-bands of 0Hz, -100Hz, -200Hz, -300Hz, -400Hz... and so on. Don't worry about these "negative frequencies"; they are simply the same as positive frequencies with the phase inverted. However, because the amplitudes of these components are not the same, they don't cancel each other out and (fortunately) we don't obtain silence!

Clearly, using "1" and "1" (or, for that matter, any equal numbers) as the carrier and mod values generates an harmonic series, albeit with an unusual amplitude distribution. So, what about using a carrier value of "1" and a modulator value of "3"? Now we obtain $\omega c + n.\omega m$ side-bands of 100Hz, 400Hz, 700Hz, 1kHz... and so on, and "reflected" $\omega c - n.\omega m$ side-bands of 200Hz, 500Hz, 800Hz, 1.1kHz... and so on, which is an harmonic series every third harmonic missing. This is similar to a pulse wave with a duty cycle of 1/3, although it has a somewhat different timbre.

I'll leave you to work out what other combinations of the carrier and modulator values can generate, and return to the sound described a few paragraphs ago. With no frequency modulation taking place, this gives us the composite tone presented here as sound #1. ▶ As you can hear, this has components at the fundamental, 3rd harmonic and 7th harmonic.

Figure 3: The basis for an evolving 6-op pad
(Click to enlarge)

Let's now apply Frequency Modulation. However, instead of dialling in a static amount by using the FM knob in each of the oscillators, or even using the same external source for controlling the amount of modulation, I'm going to apply three different modulators that determine the amount of FM taking place in each FM Pair so that the modulation index (and therefore the tone of each oscillator) will change dynamically without reference to the others. Figure 4 shows how I have used the modulation matrix to direct the outputs from LFO1, LFO2 and the Global Env in looping mode (so that it acts as another LFO) to the three FM Pair Osc FM amounts. I have ensured that the rates of each of these LFOs are different from the others, and I've even added some key-follow in LFO1 so that the rate changes gently as I play up and down the keyboard. The result of playing just a single note (in this case, middle C) of this patch is captured in sound #2. ▶

Figure 4: Making the pad evolve
(Click to enlarge)

This is not a type of sound normally associated with an FM synth, but now I'm going to warm it up in ways that were not possible on the 'classic' FM synths such as the DX7 and its ilk. For example, I can use the Shaper to add a degree of grittiness, plus a low-pass filter in the Filter 3 slot, which I use as a tone control to eliminate the higher frequencies introduced by the distortion. Next, I can add some un-modulated delay to give the sound more ambience. If I play this patch (shown in figure 5) a little more than an octave lower than the previous sample and record it a little too 'hot' so that it's on the edge of further clipping (yes, the residual distortion is intentional) those of you old enough to have been listening to Genesis in 1973 should go... "Oh, wow!" ▶ As you can hear, FM synthesis has a wholly undeserved reputation for creating only thin, percussive and generally uninteresting sounds.

Figure 5: "Me? I'm just a lawnmower..."
(Click to enlarge)

FM as a noise generator

I've now explained what happens when FM side-bands drop below 0Hz, but that leaves an interesting question... what happens when they exceed the bandwidth of the system? I'm going to answer this while discussing the concept of FM feedback.

If you look closely at most hardware FM synthesisers, you'll notice that many algorithms show a loop in the algorithm, as illustrated in figure 6. This is called "feedback", and it shows that the output of (in this case) operator 6 is being fed back into its own FM input.

Figure 6: Classic FM feedback

Thor is unable to do this because operator 6 is one of the Mod oscillators in an FM Pair Osc, and we cannot access it. However, all is not lost, and we can feed the output of an FM Pair into its own FM input (as shown in figure 7) using the modulation matrix. If the Mod oscillator is contributing no FM, we are now feeding operator 5 back into itself.
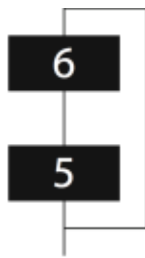
Figure 7: Feeding back an FM Pair

To investigate what this does, let's start with a simple FM set-up that has no frequency modulation taking place. As we discussed in the previous tutorial, this patch will generate the consistent sine wave heard in sound #4. ▶

Now, instead of rotating the FM knob to modulate the FM Pair Osc's carrier using its built-in modulator, let's use the modulation matrix to feed the output of the oscillator back into its own FM input. I'll use an envelope generator (in this case, the Mod Env) to control the amount of feedback so that you can hear what happens as it is progressively increased. This patch is illustrated in figure 8 and its output presented as sound #5. ▶ As you can hear, the tone begins to change dramatically as soon as the feedback level is increased above zero and, after a transition zone in which the frequency is unstable, the output quickly turns to noise.

Figure 8: Progressively increasing the amount of feedback
(Click to enlarge)

If you consider what's happening, this makes sense. As we've discussed, side-bands are created when the carrier is frequency-modulated, and when the modulation index (which, in this case, is determined by the amount of signal in the feedback loop) is small, an harmonic series is produced so we obtain a more complex tone than before.

Now let's skip over the unstable region (which, to be honest, defies simple explanation) and consider what's happening when the modulation index is large. As explained in the previous tutorial, the bandwidth of the spectrum is determined by the modulation index, but no digital synthesiser has infinite bandwidth, so when the upper end of the harmonic series tries to exceed the bandwidth of the system, an effect called "aliasing" occurs. This is, in effect, a reflection of component frequencies off the upper limit of the system (called the Nyquist frequency) and it generates new frequencies that - except by the most remarkable co-incidence - will not lie on the harmonic series. Now the carrier is being fed all manner of unrelated frequencies, and these produce a hugely complex new spectrum that is again fed back into the oscillator, further complicating the output. This is again fed back again into the operator and, as you might imagine, the output soon includes thousands, then millions of component frequencies, which we hear as noise. This makes FM an unlikely, but hugely powerful generator of drum, percussion and other noise-based sounds. Use the modulation index to determine the noise colour and then shape the sound appropriately and... voila! Sound #6 (which was generated using two instances if Thor) demonstrates what you can achieve using FM Pair oscillators to emulate an analogue drum machine.

A few more tips

As we have seen, Thor's FM Pair Osc offers just integer relationships (1, 2, 3... and so on, up to 32) between the carrier and modulator frequencies. This means that, aliasing notwithstanding, it will always generate an harmonic series. But many of the more interesting FM sounds, such as bells, chimes and gongs, have enharmonic components generated by non-integer relationships between the carriers and modulators. Happily, Thor allows us to do this, although not necessarily with the same flexibility as a DX7.

Figure 9 shows a patch in which the output from the FM Pair Osc in slot 2 is modulating the FM Pair Osc in slot 3 via the modulation matrix. There are a number of things to note about this patch. Firstly, Osc2 is detuned with respect to Osc3, so the side-bands are "enharmonic" - i.e. they do not lie on harmonic frequencies. This is hugely important, because many natural resonators are enharmonic, and it's hard to generate their sounds in any other way.

Secondly, the amount of frequency modulation (determined by the first line in the modulation matrix) is being scaled by the MIDI Note number. This is also hugely important. You may remember that I wrote in the previous tutorial, "if you want the tone of an FM-generated sound to be consistent as you play up and down the keyboard, you need all three of the modulator, the carrier, and the amplitude of the modulator to track by 100%". Scaling the amount of FM in the modulation matrix is the way that we achieve this.

Thirdly, I have used the note number to shorten the decay of the Amp Env as I play up the keyboard, thus emulating one of the natural characteristics of acoustic sounds and, finally, I have added just a smidgen of pitch modulation to Osc3, driven by LFO1. This doesn't just add vibrato; it also modifies the nature of the sound slightly because it's changing the relationship between the modulator and carrier frequencies.

You can hear the result of this patch in sound #7.

Figure 9: An FM musical box
(Click to enlarge)

Final thoughts

I could write many, many tutorials about the capabilities of FM synthesis and how you can take advantage of them within Thor. For example, I haven't touched upon the amazing brass sounds that FM is capable of generating, or the solo strings that can almost fool you into thinking that you're listening to a genuine orchestral instrument. But I'm going to finish by showing you the hugely complex FM algorithm that Thor is capable of supporting when you insert FM Pair oscillators into each of the three slots available. The black lines in figure 10 are "hard-wired", and any of the blue, red and green lines can be invoked (or not) in the modulation matrix. If you choose the appropriate lines and then redraw the relationships between the operators, you'll find that subsets of figure 10 can recreate many of Yamaha's FM algorithms. But in addition to the FM paths shown, there are three amplitude modulation and sync paths hard-wired within Thor, not to mention the more esoteric FM and other interactions that you can set up using scaled paths and functions in the modulation matrix. Ultimately, there are many things that you can do on a DX7 that you can't do in Thor, but there are just as many more that you can do in Thor that you can't do on a DX7. In all likelihood, only a limited number of these will be musically useful but, if you're prepared to experiment, who knows what you might discover? Now it's over to you... enjoy FM.
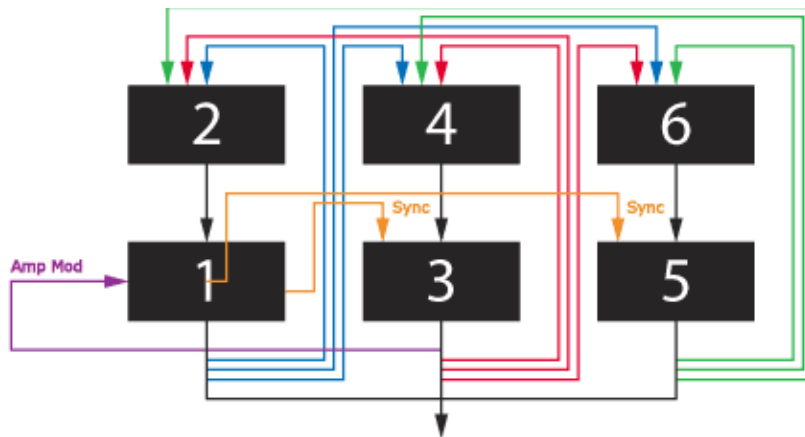
Figure 10: Thor's FM algorithm

Text & Music by Gordon Reid

Thor demystified 11:
The Wavetable oscillator - Part 1

So, we finally come to the sixth and final oscillator in Thor's armoury of sound generators. This is the wavetable oscillator that first appeared in general use in the PPG Wave Computers, and shortly thereafter in the PPG 2.x series. Like other digital oscillators, this is an often misunderstood beastie, so let's first discuss what a wavetable actually is.

Some definitions

There have been a number of different uses of the word *wavetable* in recent years, and some of them are rather misleading. For example, I have seen texts that use the name to describe a ROM that holds a selection of unrelated PCM samples such as clarinets, electric guitars, bouzoukis and Mongolian nose flutes. There are lots of waves in the ROM and these are accessed using a lookup table, so the ROM must be a wavetable, right? Wrong!

More justifiably, academics use the word to describe the sequence of numbers that represent a single cycle of a regular, periodic waveform. One can then talk about replaying one such "wavetable" (say, a digital representation of a sine wave) or a second (say, a digital representation of a sawtooth wave).

But this is still not the definition used by most people who talk about *wavetable synthesis*. So let's be explicit. For our purposes, a wavetable is not a single wave, it is a selection of (usually related) single-cycle waveforms or their harmonic representations stored digitally and sequentially in such as way that that a sound designer can create musically pleasing timbres by stepping though them while a note is being played.

Unfortunately, while the principle makes sense, this would not sound very pleasant. Imagine a ROM containing just two waveforms: the aforementioned sine and sawtooth waves. Now imagine hearing a sound comprising a few cycles of the sine wave followed by a few cycles of the sawtooth wave, followed by a few cycles of the sine wave followed by the few cycles of a sawtooth wave... and so on. The resulting waveform would exhibit discontinuities each time that one waveform replaced the other, and you would therefore hear a succession of clicks polluting the sound. Consequently, a *wavetable synthesiser* has to be a bit cleverer than that, providing a mechanism for morphing from one wave to the next. Instead of swapping instantly from the sine wave to the sawtooth wave, there would be a transition period during which the waveform changed smoothly from one extreme (the sine wave) to the other (the sawtooth wave) and back again.

A simple wavetable synthesis patch

Of course, the waves chosen don't need to be sine and sawtooth waves, and this concept works with any two single-cycle waves of your choosing. It is, therefore, the simplest generalised form of wavetable synthesis.

To illustrate it, let's create a wavetable synthesis patch in Thor. Figure 1 shows the wavetable oscillator in its most basic configuration, with the Basic Analog wavetable loaded and everything set to sensible default values. As you can see, the "Position" parameter is set to zero in this figure. All other things being equal, if you now play a note the oscillator will generate a sine wave

Figure 1: Using the Wavetable Osc to generate a sine wave

Hold a key down and sweep the Position knob around from its minimum to its maximum value. If you do this slowly and carefully, you will be able to see the values at which the oscillator's output changes from one waveform to the next. Having done this, I know that the waveforms contained within this wavetable are as shown in the table below.

Position Value	Waveform generated
0 – 36	Sine wave
37 – 72	Triangle wave
73 – 108	Square wave
109 – 127	Sawtooth wave

So let's now consider what happens when, instead of turning the knob manually, we use some form of modulator to adjust the Position parameter for us. Figure 2 shows a simple patch with no filters, no effects, and no complex modulations. There's just a very slight slope to the Attack and Release in the Amp Env to eliminate keying clicks, and a single modulation route in the matrix that directs the output from LFO1 to the Position parameter within the Wavetable Osc itself.

If we now set the initial Osc1 Position to 36 (so that it's right on the cusp of the transition between the sine and triangle waves) and apply just enough LFO to push the Position above 37 on the positive side of its sweep and below 36 on the negative side, we can hear the sound switching regularly between the sine and triangle waves. But even with such similar waveforms, the glitch at the transition is very unpleasant.

Figure 2: Using an LFO to switch between simple waveforms
(Click to enlarge)

To illustrate what's happening, I've drawn figure 3, which looks a bit like a sine wave drawn over some type of multi-layer cake. Because you're used to seeing waveforms drawn in this fashion, you might think that this is in some way showing the waveform that you're hearing, but it isn't. If you look again, you'll see that each of the layers in the "cake" represents a waveform, and that the position and "thickness" of the layer corresponds to the table that I drew above. In other words, figure 3 illustrates the Basic Analog wavetable, and the red line shows the modulation curve that determines the nature of the output at each moment in Sound #2.
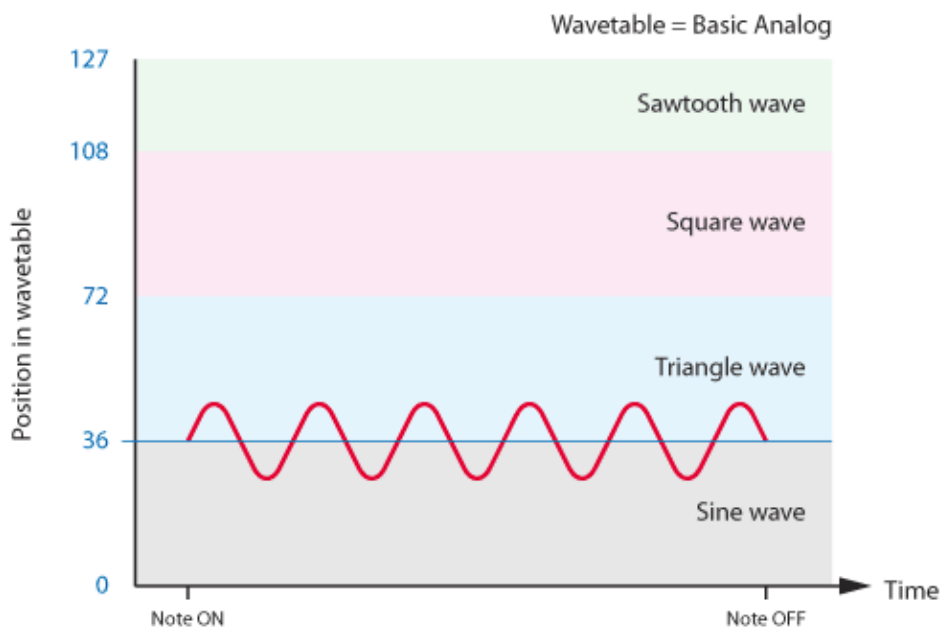
Figure 3: Representing movement within a wavetable

Fortunately, as I mentioned above, any usable implementation of wavetable synthesis is capable of generating smooth transitions between waveforms. Figure 4 shows the Wavetable Osc with its Position at 36, and its X-FADE (cross-fade) button switched on. If you now play the patch in figure 2, you'll obtain Sound #3 ▶ , which exhibits a pleasing harmonic modulation as the waveform sweeps gently between the sine wave at one extreme and the triangle wave at the other.

Figure 4: Removing glitches by cross-fading between waves

More ways to use a wavetable

Of course, you're not limited to using cyclic modulators to move between waves, and there's nothing stopping you from replacing the LFO in figure 2 with a contour created by one of Thor's envelope generators. Figure 5 shows how you can use the Filter Env to sweep your Position in the Basic Analog wavetable slowly from its maximum value to zero. In harmonic terms, this means sweeping from a sawtooth wave (in which all the harmonics are present) to a square wave (all the even harmonics are eliminated) to a triangle wave (which also has only odd overtones, but with lesser amplitudes than in the square wave) and finally to the sine wave with only the fundamental surviving. You can hear this without cross-fading in Sound #4(a) ▶ , and with X-FADE On in Sound #4(b) ▶ .

Figure 5: Using the Filter Env to sweep through the wavetable
(Click to enlarge)

Sweeping across four idealised waveforms in this fashion is interesting, but hardly earth-shattering, even though it generates a sound that is impractical to obtain using other synthesis methods. So let's hear a more startling illustration of a wavetable sweep by replacing the Basic Analog wavetable with one of the others provided within Thor. Sticking with the patch in figure 5, and simply stepping through the tables, Sound #5(a) ▶ was generated by the Synced Sine table, Sound #5(b) ▶ was generated by the Synced Ramp table, and Sound #5(c) ▶ was generated using the Square Harmonics table.

To clarify what's happening within the Wavetable Osc in this patch, figure 6 represents a wavetable comprising twelve waveforms (many of the tables have 32, but that would be too many for the figure to be clear) and illustrates the idea of sweeping through the table. The identity of the red line is now obvious: it's the Filter Env contour that's controlling which waveform within the table is being output at any given moment.
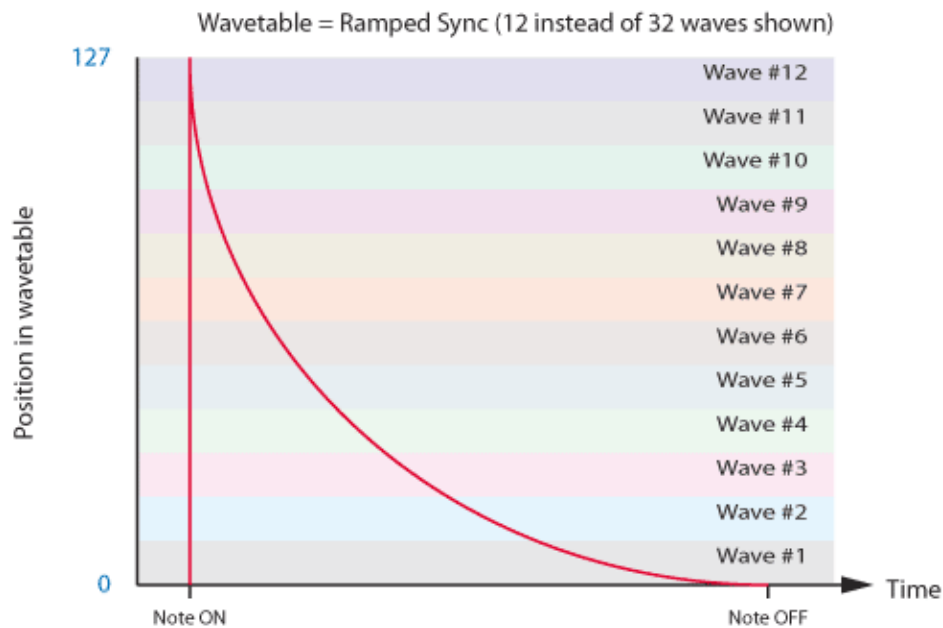
Figure 6: Sweeping through a wavetable

Interesting in isolation, these sweeps can create excellent sounds when combined with other types of oscillator. Figure 7 shows a swept wavetable oscillator paired with a Multi Osc that's producing a detuned sawtooth wave. The combination of the two creates Sound #6 ▶ , which emulates (or even improves upon) the sync'd bass sounds generated by powerful analogue synths such as the Moog Source. All this, and yet there's no sync, and still not a single filter or effect inserted into the patch!

Figure 7: Creating a 'sync' sound
(Click to enlarge)

So far, we've only looked at wavetables in the context of emulating sounds reminiscent of analogue synthesis. Let's now step beyond this and use the same patch to create one of the fragile, glassy tones first heard on the early PPGs. These are, after all, the sounds that made these synths so desirable.

Returning to the patch in figure 5, step through the list of wavetables in Thor and select the one named "10 Sines". If you now sweep through the table from its last element to its first without cross-fading, you'll hear the twelve distinct waveforms that were chosen by Thor's programmers and placed in the table in a quasi-random order. ▶ Now switch on the X-FADE function. Ah... that's more like it. Sound #8 ▶ is an excellent example of the type of timbres that made the PPG Wave 2.0 famous.

Creating imitative sounds using a wavetable

So far, we've been using wavetables to generate relatively simple sounds. Now, we'll step beyond these and move to genuine sound design using wavetable synthesis.

Let's imagine that we want to create a range of solo brass sounds using a wavetable synthesiser. To do this, we could imagine a wavetable that included snapshots of the timbres generated by one of the brass instruments, ranging from the very bright tone that exists just after the start of the note to the slightly duller tone that tends to make up the body of the sound. We could envisage a wavetable that includes these as shown in figure 8.
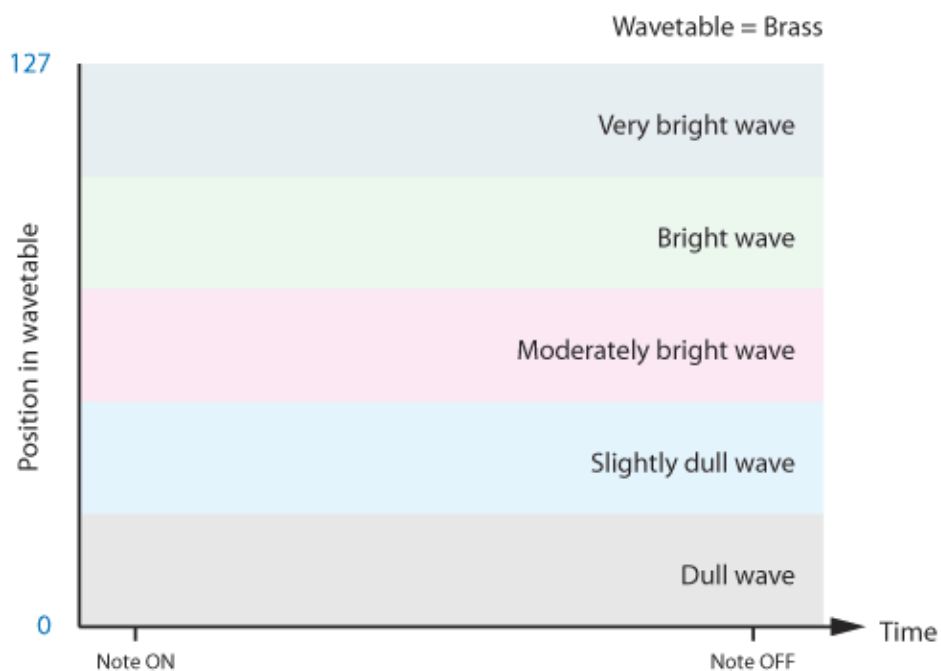
Figure 8: A hypothetical brass wavetable

Now we need to decide how we wish to move through this wavetable. Obviously, applying an LFO to oscillate between waves is not appropriate, nor is sweeping through the entire table. Instead, we want the sound to start with a dull timbre and then, quite rapidly, become very bright. Next, the brightness will diminish until it reaches the timbre that it will maintain while the note is being sustained. Finally, we want the note to become dull again and disappear quite rapidly when the key is released. We can draw this as in figure 9.
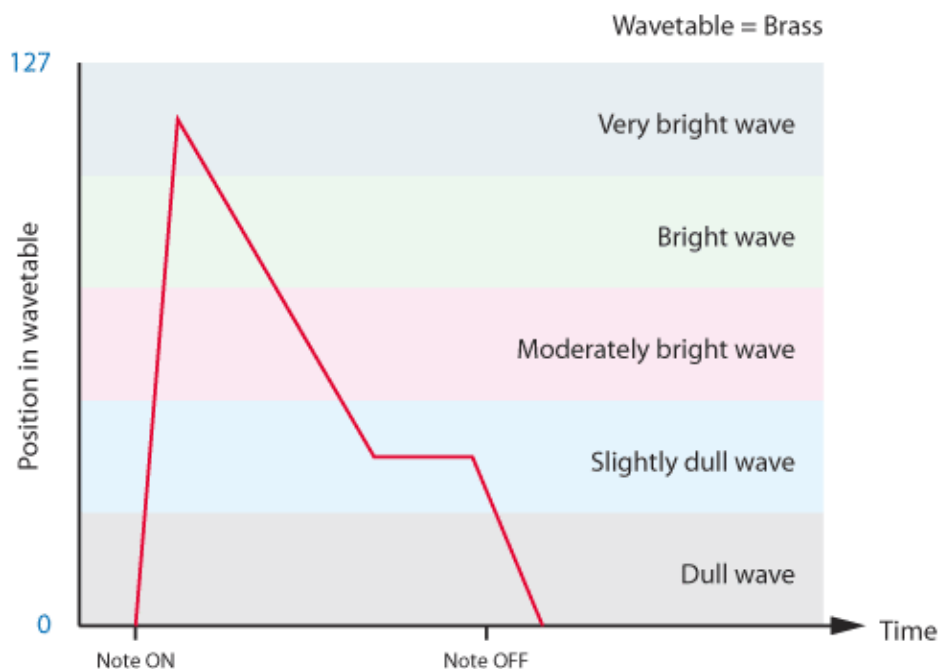
Figure 9: Moving through a suitable wavetable to create a brass sound

Hang on a minute... if we were creating a brass sound on a subtractive synth, this would be the filter contour for the patch. Of course it would, but instead of taking a waveform of constant harmonic content (a sawtooth wave) and then opening and closing a filter to create timbral modifications, we're taking a sequence of waveforms with the correct harmonic content at a handful of points, and then interpolating between these to create the required tonal changes as the note progresses. No filter is needed.

Naturally (or I wouldn't have chosen this as an example) Thor contains a wavetable that allows us to convert these ideas into reality. However, the brightest sound in this table exists at Position = 0, and the dullest is at Position = 127 so we need to invert figure 9 to create figure 10.
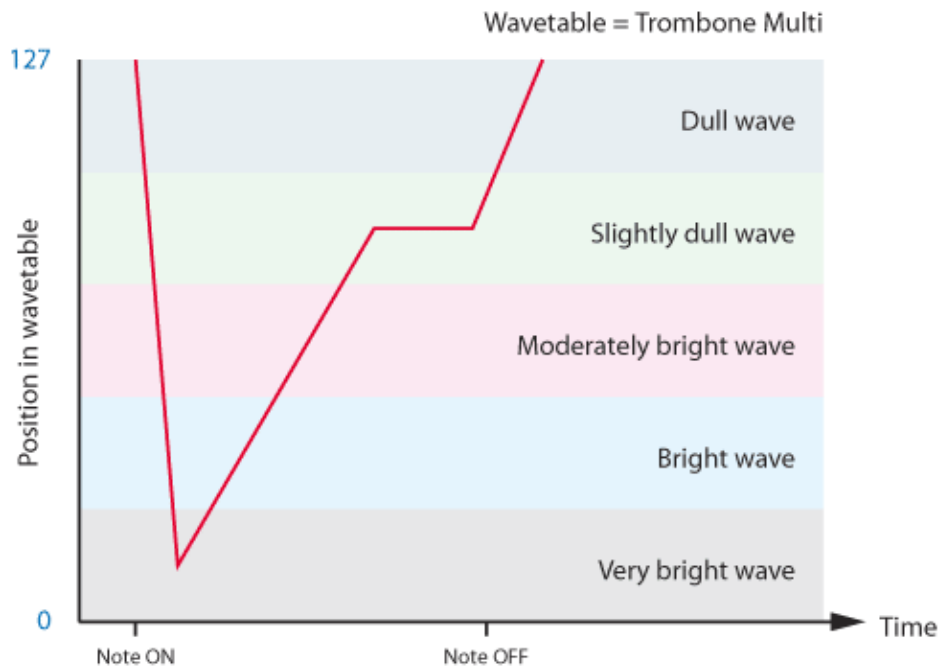
Figure 10: Using the Trombone Multi wavetable in Thor

This then tells us how to proceed. First, select the Trombone Multi wavetable and set the Position parameter to 127. Now create the appropriate ADSR contour in the Filter Env and use the modulation matrix to direct this in inverted form to the Osc1 Pos parameter. A neat trick here is to make the amount of modulation dependent upon the velocity so that if you hit a key softly the sound is less bright than if you hit it hard. Finally, direct a little LFO to the pitch of Osc1 for a gentle vibrato, simply because it adds interest to the sound. (Figure 11.)

Figure 11: A wavetable synthesis brass patch
(Click to enlarge)

The values in the modulation matrix are quite critical for achieving the desired result, but I found that a Position modulation of -88 works quite well (it has to be negative to make the contour in figure 10 go down and then up rather than vice-versa) and a velocity amount of around 72 seems to work nicely. Now you can set up the Filter Env contour and the LFO to taste and play.

Sound #9 is an example of this patch played with lashings of reverb to achieve a nice, ambient effect. You can create innumerable variations on this by adjusting the parameter values within the patch, and if you don't stray too far the timbre will remain brash and brassy, getting brighter and darker in the way that a brass sound should. But there are still no filters anywhere to be seen.

I love this stuff.

Text & Music by Gordon Reid

Thor demystified 12:
The Wavetable oscillator - Part 2

Why were wavetables developed?

We now hold early wavetable synths such as the PPGs in high esteem and think of them as expensive examples of early digital synthesisers. However, wavetables were actually invented to make possible relatively low-cost instruments that avoided the shortcomings of existing digital synthesis methods such as FM and additive synthesis, and to overcome the immense technological limitations of the day.

To understand this, imagine that you want to use a piece of audio equipment today to record, store and replay the sound of a someone saying "wow". You choose a suitable recorder or sampler, capture the sound and, without any need to understand how the equipment does what it does, you can then replay it. If you used a digital recorder, you do so simply by pressing the Play button; if you used a sampler, you allocate the sample to a key or pad, then press that to listen to the recording that you've made.

However, we don't have to travel back many years to a time when none of this was practical. The problem was two-fold. Firstly, early memory chips were extremely limited in capacity, and storing anything more than a fraction of a second of audio was very expensive. Secondly, even if you could store the audio, the primitive microprocessors available at the dawn of digital synthesis were barely able to address the memory and replay it at an adequate speed.

Let's consider the word "wow", which takes about a second to say. Using the sampling specification introduced for the audio CD (44,100 16-bit samples per channel per second) you would require 88.2KB of RAM to record the word in mono, and double that if it were recorded in stereo. Nowadays, we wouldn't blink at that sort of requirement, but when the earliest digital audio equipment appeared, you needed as many as eight chips for just 2KB of RAM. Sure, samplers were shipped with boards stuffed full of dozens of these, but you would have needed no fewer than 352 of them to store the word "wow" at CD quality!

Clearly, this was impractical so, while various digital tape formats were able to provide the hundreds of megabytes of audio data storage needed to edit and master whole albums of music, developers of digital musical instruments were looking at much more efficient ways to record, store and replay sounds for use in synthesis. The wavetable was one such method.

Synths that make you go "wow!"

So… it's 1979 and you want your keyboard to say "wow". It's not impossible – a mere £20,000 (around £100,000 at today's values) would buy you a Fairlight CMI, which is just about capable of doing this. But instead of emptying your piggy-bank, let's imagine that you can slice the word into eight pieces, one beginning at the start of the sound, the next 1/8 of a second after the start, the next 1/4 of a second after the start… and so on until it has run its course. If each of these slices was a full eighth of a second (0.125s) long, you could reassemble the entire waveform simply by replaying them in the correct order. But each slice would still require about 10KB of storage, and this will have saved you nothing. But now imagine making each snippet much shorter… say, 0.0125s (of the order of the length of a single cycle of audio when spoken by a human) and separating each by the appropriate length of silence. Each slice

would now require about 1KB of RAM and, instead of being a sample in the conventional sense, would represent the sound at a discrete moment as the word was spoken. This is not as daft as it sounds. Tests had showed that – depending on how much data you removed – you could analyse the harmonic content at various moments and then, on replay, use a mathematical method called 'interpolation' to fill the gaps between slices with an estimate of the sound that had previously existed. This allowed you to obtain a close approximation to the original sample, but with a much reduced data storage requirement.

With around 8KB of memory required, recreating the word "wow" was getting closer to being practical, but additional space saving measures were still necessary. For example, 16-bit data was a luxury in the early 1980s, so samples were often recorded at a resolution of 8 bits per word, which decreased the memory requirement to just 4KB. Indeed, early digital audio systems often used quite severe compression techniques to reduce the storage still further… in this example, down to around 2KB.

While you don't need to store all of the audio data to be able to reconstruct a recognisable approximation to the original sound, it becomes increasing difficult as you discard more and more of it. Nonetheless, if you have eight single sample snippets derived in this way, you can reconstruct something that is recognisable. You want proof…? No problem. Let's turn to the Wavetable Osc in Thor and select the Voice wavetable, as shown in figure 1. If you start with its Position parameter set to zero, and then sweep through the position manually to record the points at which the timbre changes, you'll find that the wavetable comprises seven waveforms, as shown in table 1.

Figure 1: Selecting the Voice wavetable

Position Value	Waveform generated
0 – 19	Wave 1
20 - 39	Wave 2
40 - 58	Wave 3
59 - 78	Wave 4
79 - 97	Wave 5
98 - 117	Wave 6
118 – 127	Wave 7

If you now sweep through these using the Filter Env contour generator to control the Position as shown in figure 2, you'll hear that the waves have a slightly vocal quality, although none of them actually sounds like a vocal sample.

Figure 2:Sweeping through the Voice wavetable
(Click to enlarge)

However, I have a bit of inside knowledge, and I know the nature of the sample from which the waves in this wavetable were derived. Armed with this knowledge, I can reassemble it. First, I have to switch the X-Fade button ON. (X-Fade is a very simple example of the mathematical interpolation that I mentioned above, and it will generate a rough estimate of the audio that existed in the gaps between the snippets in the wavetable.) Second, as show in figure 3, I need to set up a different set of envelope parameters to sweep through the waves at the correct rate. Having done so, I can now play a note. Wow indeed!

Figure 3: Synths that make you go "wow"
(Click to enlarge)

Of course, this method of storing and reconstructing sounds is not constrained to the word "wow", and Thor contains three tables designed to be used in this way: Voice, Piano and Didgeridoo. Figure 4 shows a patch that makes great use of the Didgeridoo wavetable. The thing to note here is that I couldn't use the Filter Env or Amp Env to sweep through the table because the sound I want to create requires a loop. Furthermore, an LFO wouldn't be ideal because the two directions of the sweep need to be of unequal rate. But a looping contour generator, on which you can independently determine the length of the A and D stages, is perfect. Sound #3 demonstrates this patch, and there's no doubting what the original sound was before it was cut up into tiny pieces.

Figure 4: Didgeridoo
(Click to enlarge)

A different way to use a wavetable

Instead of creating a table that's designed to play successive snippets of a single sample, imagine one that's created using snippets from samples recorded at different pitches. For example, you could ask a brass player to play a succession of different notes and then extract the waveform that lies exactly two seconds into each. In theory, you could then allocate these to zones on the keyboard so that the timbral variations of the instrument are correctly mapped from the lowest to the highest notes played. This is the basis of "multi-sampling", the technique used to create sample libraries and the ROMs within PCM-based digital synthesisers.

Thor contains two wavetables designed for use in this fashion: the Trombone Multi and the Sax Multi. In tutorial #11, I used (or perhaps abused) one of these – the Trombone Multi – to create a sound whose harmonic content varied in time, so let's now hear how this sounds when the waves are distributed across the keyboard as they were meant to be.

Fig 5: Hearing the individual waves in the Trombone Multi wavetable

(Click to enlarge)

Figure 5 shows a single Wavetable Osc with its Position parameter set to 40. Below this, you can see three identical paths in the modulation matrix. These cause the Position to change rapidly with respect to MIDI note number, condensing the range of notes over which the waves are distributed so that I can demonstrate them effectively.

Sound #4 was played using this patch, and you can clearly hear the zones in which each wave lies. These discontinuities would be a nightmare if you were creating the ROM for a digital synthesiser, but in a wavetable synthesiser the differences between groups of notes can be used creatively. To illustrate this, I've further developed the patch above to include two Trombone Multis and one Sax Multi, as shown in figure 6. Cheating a little by passing the output from this through Reason's UN-16 Unison processor and adding a touch of reverb, I obtained sound #5. Hmm… that sounds like something I know and love!

Fig 6: Composite brass patch
(Click to enlarge)

Of course, there's nothing stopping you from using a wavetable designed to be played sequentially (such as the Voice wavetable) and distributing it across the keyboard, just as last month I took the Trombone Multi (designed to be distributed across the keyboard) and played it sequentially. Figure 7 shows a wavetable patch with the Voice table inserted, and Position tracking that creates bands in which the various waves play. You can hear this in sound #6.

Fig 7: Inverse tracking to obtain different "zones"
(Click to enlarge)

To turn this into a useful vocal sound, I'm going to add some gentle vibrato using LFO1, and a slight pitch instability from LFO2. As shown in figure 8, both of these are routed via the modulation matrix to the pitch of Osc1, and you can hear the result in sound #7. Clearly, each of the four notes has a markedly different timbre, so all I need do is add some chorus and reverb to obtain a wonderful patch that exhibits different vocal characters from one end of the keyboard to the other. All this and, as always, there's not a single filter in sight.

Fig 8: Adding vibrato and instability to the "voices"

(Click to enlarge)

Inevitably, life at the dawn of digital synthesis wasn't quite as simple as these examples imply, and there were many problems to be overcome before it was possible to build a general-purpose wavetable synthesiser. Most notably, it was discovered that the sample snippets usually have to be resynthesised and phase-aligned so that you can loop single samples and cross-fade between adjacent ones. But, while awkward, these problems were not insurmountable and, when the first wavetable synthesisers appeared, their lesser demands for processing power and memory meant that they were very much cheaper than the samplers of the era. But while the PPG Wave 2.2 and Wave 2.3 are now revered by many players, I reckon that they would kill to be able to create the sounds in this tutorial. So don't underestimate what Thor's Wavetable Osc can do for you. The world of wavetable synthesis is not limited to delicate chimes, glassy pads and 1980s synth-pop, as some might have you believe.

So many sounds, so little time…

We have now covered the two fundamental aspects of wavetable synthesis: making the sound evolve in time, and changing the timbre at different points across the keyboard. But there's no reason to stop there, and many fascinating sounds can be derived from using both techniques simultaneously. Unfortunately, I'll have to leave you to discover them for yourselves, because we've come to the end of my tutorials explaining Thor's oscillators. I hope that they have given you some ideas, and illustrated why there is so much more to synthesis than tweaking the cut-off knobs of resonant low-pass filters. Indeed, I hope that they have illustrated why you don't need filters at all to be able to create a huge range of fabulous sounds. If so, my job is done. Thank you for reading; I appreciate it.

Text & Music by Gordon Reid

Control Remote

Remote – Propellerhead's protocol for communication between hardware control surfaces and software applications – quietly celebrated its 5th birthday this January. It was introduced as one of the top billed new features of Reason 3.0 at Winter NAMM in 2005. The purpose of Remote was to save Reason users from the tedium of setting up control surfaces manually, and to provide a tight, seamless integration that makes the control surface an organic and dynamic extension of the software – not unlike having a hardware synth in front of you. Hence the catchphrase "Play Your Reason System", which has a more musical ring to it than "Program Your Reason System" – it's all about eliminating distractions that interrupt the creative flow.

In this article we will be peeking under the hood of Remote and learning how to customize Remote Maps, so that you'll be able to tweak existing maps to your liking. Why would you want to, you might ask? Well, perhaps you find that the rack device parameters prioritized by the default Remote Map aren't the ones you consider to be the most important and useful. Maybe you wish that the parameters controlled by the fader set were controlled by the knob set, and vice versa. Maybe you want to create your own custom setup for live performances. Either way, you have our blessing to hack the Remote Maps to bits!

It should be noted that everything that applies to Reason in this article, also applies to Record – both applications support the Remote protocol – but technically this is "Discovering Reason", so we will only be referring to Reason henceforth.

Maps vs. Codecs

There are two main types of Remote files:

1) **Remote Codecs** are definition tables that tell Reason which physical controls exist on the control surface. Keys, knobs, buttons, pedal inputs, data entry sliders, meters, displays and so forth. For example, the Codec may specify that there's a Pitch Bend wheel and that its minimum and maximum values are 0 and 16383, or that there are 8 rotary controls (defined as "Knob 1", "Knob 2" etc.) which can transmit values from 0 to 127. There is little point in modifying these Codecs, due to the obvious futility of trying to convince Reason that the control surface has fewer, more or different controls than it really does. Having said that, in case you own a control surface that isn't officially supported by Remote, you can use the existing Codecs as templates. Remote Codecs are created using the Lua scripting language. If you're fluent in Lua and want to create or edit codecs, you should apply for the Remote SDK here.

Note: There is also another type of Codec file format called MIDI Codec, based on simple tab delimited text rather than Lua. This older format is deprecated, but still supported for legacy reasons.

2) **Remote Maps** are the main mapping tables that link the physical controls on a hardware surface to controls on the currently selected device in the Reason rack. These are the files that we'll be taking a closer look at here. Remote Maps are simple, tab delimited text documents which can be edited without any prior knowledge of scripting languages and such – all you need is basic text editing software, half a brain and perhaps a little bit of insight into the basics of MIDI. The maps are made up of separate sections for each Reason device, which often adds up to thousands of lines of data, but they're fairly easy to navigate and you can always use search functions to quickly jump to any device you're particularly interested in tweaking.

Locating the Remote Maps

All the Remote files are stored in a folder aptly named Remote, which contains two subfolders – one for codecs, the other for maps. The location of the Remote folder varies depending on operating system, as follows:

OS X
Macintosh HD/Library/Application Support/Propellerhead Software/Remote

Windows Vista/Windows 7
C:/Program Data/Propellerhead Software/Remote

Windows XP
C:/Documents and Settings/All Users/Application Data/Propellerhead Software/Remote

Please note that on Windows, the files are stored in hidden folders, so you'll have to go into the Folder and Search Options of Windows Explorer and make sure that "Show hidden files, drives and folders" is selected, in case you haven't done so already.

You can leave Reason running while editing the files. If something goes wrong when editing a Remote Map, you can simply delete it and restart Reason – the program will restore the original version again. If you want to troubleshoot your modified version rather than start from scratch, make sure to copy the modified map to a text document or the clipboard before you delete it.

Thor demystified 13:
An introduction to filters

To understand what filters do, and why they are one of the most important building blocks in synthesis, you have to understand a little about the nature of sound itself and, in particular, the nature of waveforms. So I'm going to introduce this series of tutorials about Thor's filters by talking first about what constitutes the sound generated by an oscillator, whether analogue, virtual analogue or sample-based.

Mathematics tells us that any waveform can be represented either as a wave whose amplitude is plotted against time or as a series of components whose amplitudes are plotted against frequency. Often, these components will lie at integer multiples of the lowest frequency present – for example, 100Hz, 200Hz, 300Hz… and so on – and these are then known as the fundamental and its overtones, or as harmonics. In more complex sounds, there may be components that lie at non-integer frequencies, and these are sometimes called enharmonics. The sound may also include noise that is distributed over a range of frequencies and whose precise nature is determined by all manner of factors that we need not discuss here. Figure 1 illustrates all of these.
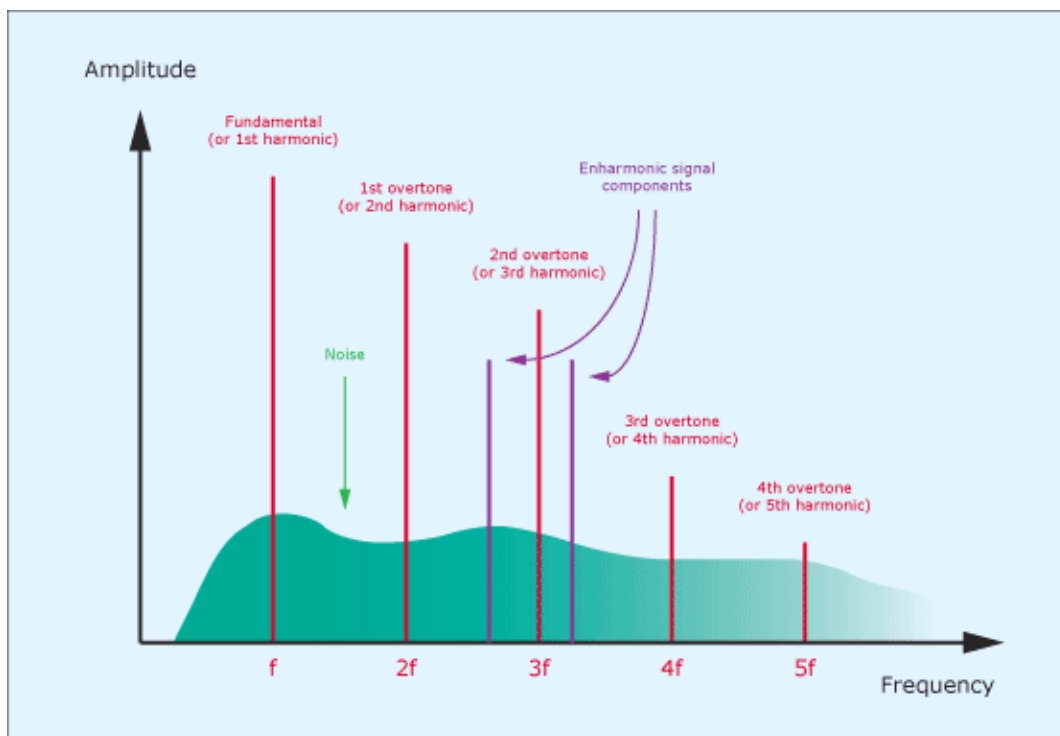
Figure 1: Signal components

Let's now look at figures 2 and 3, which represent a very common wave – the sawtooth – expressed in both these ways. The first of these figures shows it as a waveform, while the second shows the first five of its harmonics. The two look very different, but they describe the same thing. A relatively simple mathematical operation called a Fourier Transform converts one to the other, and it can easily be proven that they are exact equivalents.
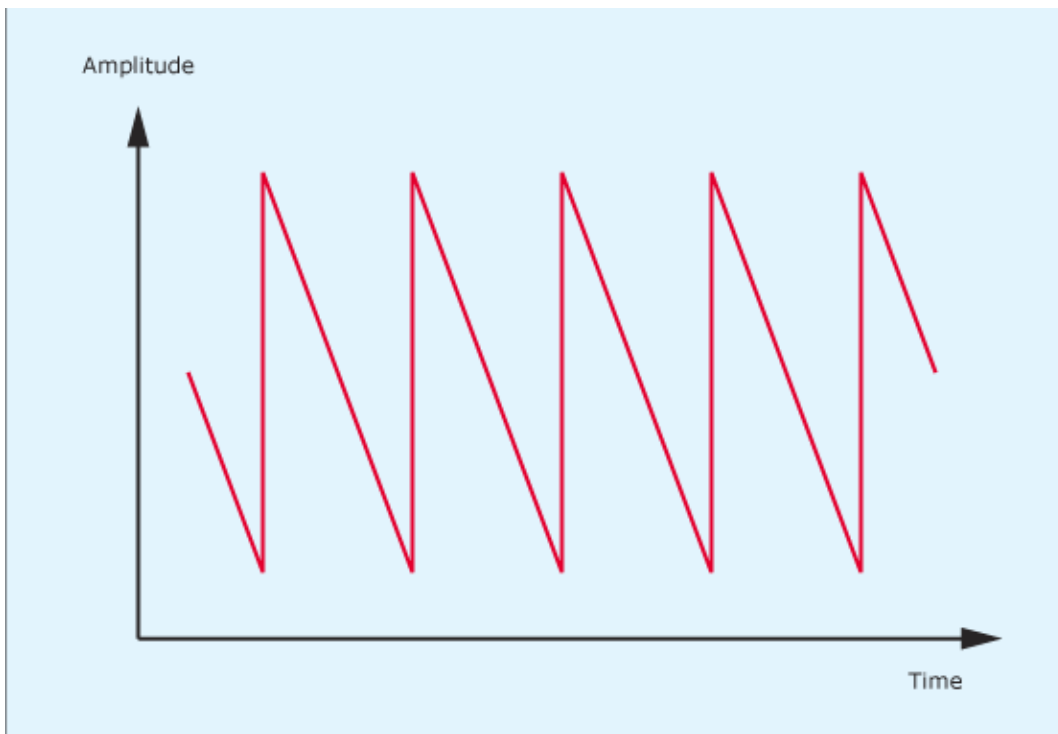
Figure 2: A sawtooth wave
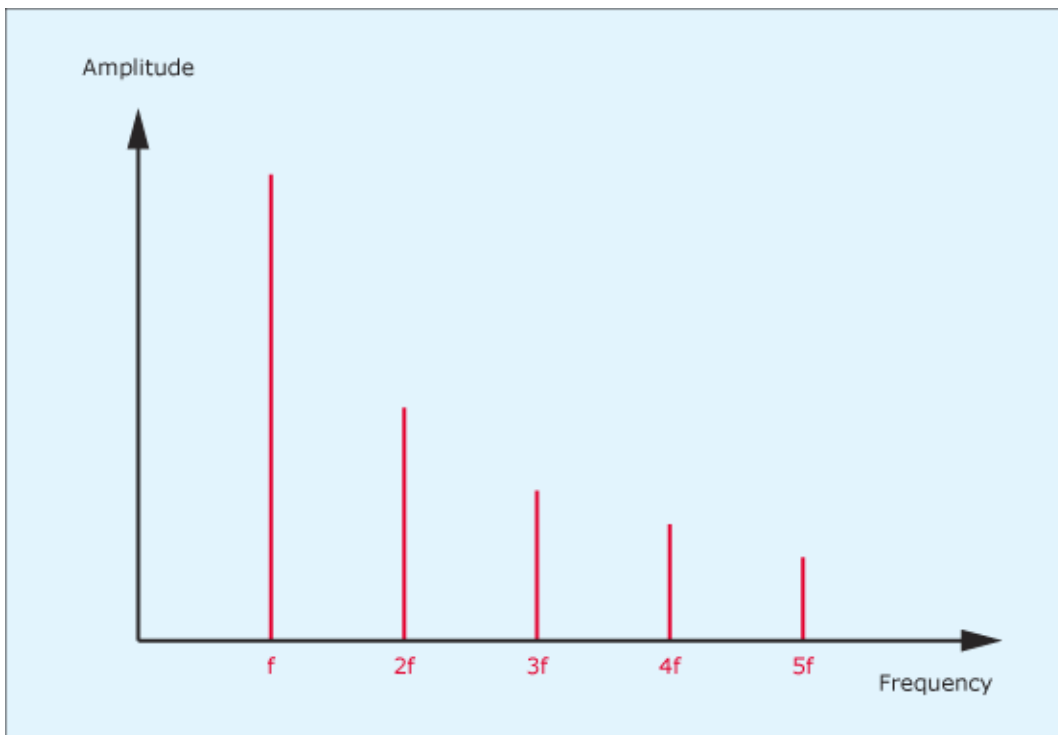
Figure 3: The first five harmonics of a sawtoth wave

What does all this have to do with a synthesiser's filters?

Let's assume that you wish to modify the nature (and therefore the sound) of a wave generated by an oscillator. In theory, you could do this by warping the wave itself in some fashion, and this is the basis of

one of the common forms of digital synthesis: Phase Distortion. But wave shaping can be a non-trivial exercise so, in the earliest days of synthesis, designers instead adopted devices that changed the amplitudes of the components that comprised the sound's spectrum. These were, of course, filters.

To describe the actions of various types of filters, let's consider the waveform with the spectrum shown in figure 4. As you can see, this has an harmonic series in which every component has equal amplitude. (It would be a horribly bright and harsh sound, but we need not worry about that.)
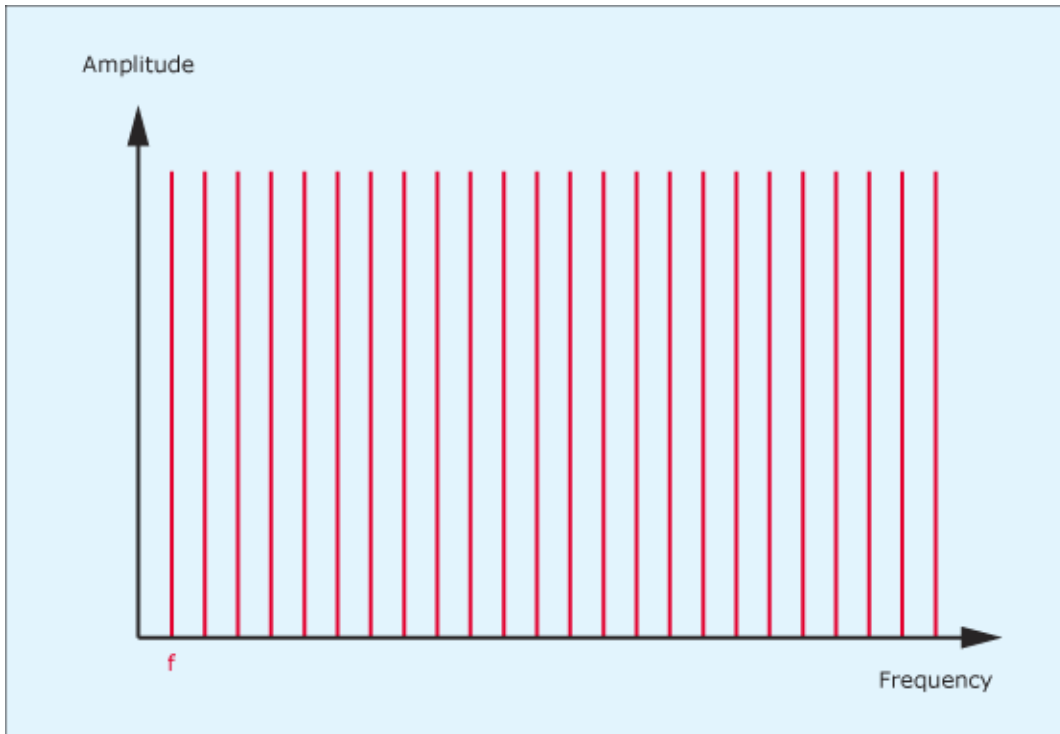
Figure 4: A sound whose harmonics all have equal amplitude

Now let's modify this sound using the most common filters found on an analogue or virtual-analogue synth. The most useful of these is the low-pass filter which, as the name implies, attenuates the signal components above a certain frequency (called the 'cut-off frequency', or fc) and passes those below it. The result can be expressed as figure 5.
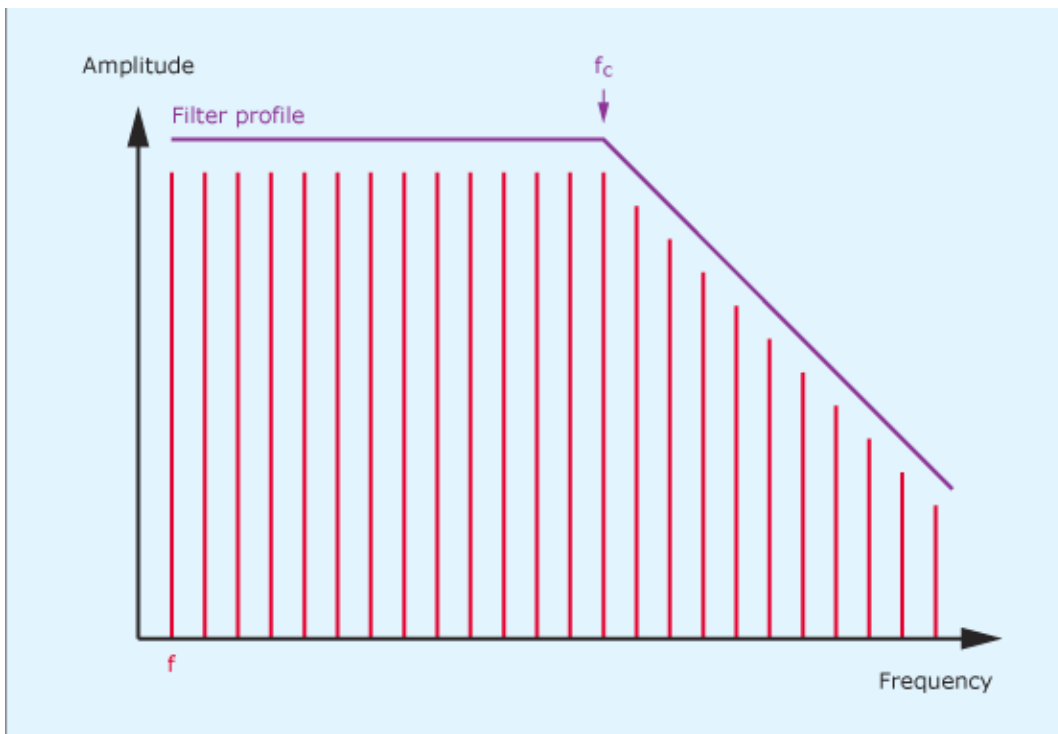
Figure 5: The action of a low-pass filter.

The high-pass filter is another common device. Again, its action is described in its name: signal components above the cut-off frequency are passed unhindered, while those below it are attenuated. (See figure 6.)
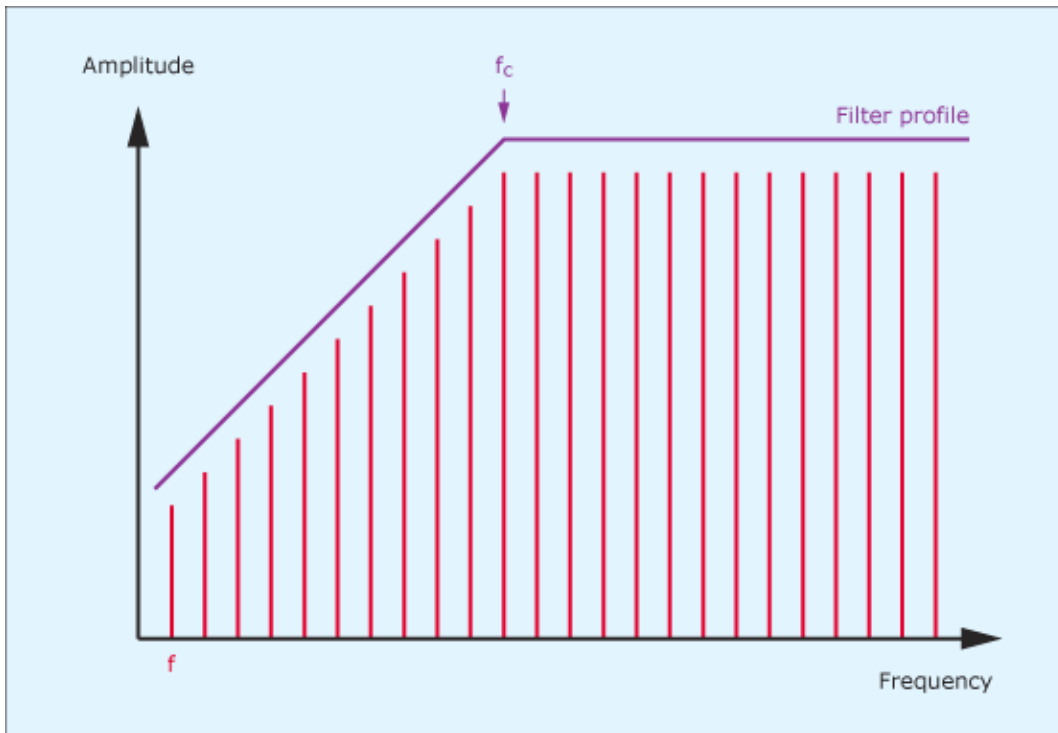
Figure 6: The action of a high-pass filter.

A third type of filter can be thought of as a combination of a low-pass filter and a high-pass filter placed in series. If the signal is first passed through the low-pass filter with cut-off frequency fcL, and then through a high-pass filter whose cut-off frequency fcH is lower than fcL, both the highest frequencies and the lowest frequencies of the original signal will be attenuated, as shown in figure 7. This, for obvious reasons, is called a band-pass filter.
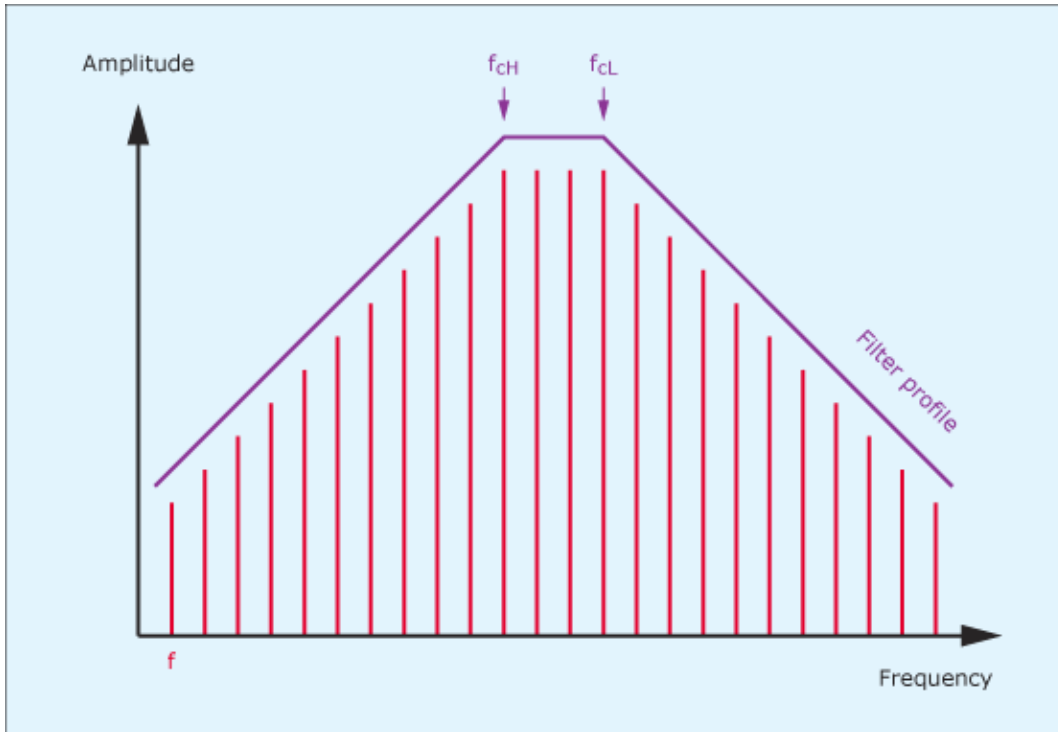
Figure 7: The action of a band-pass filter.

Finally (for today) we can consider the situation where the signal is passed through the low-pass and high-pass filters in parallel rather than in series, and where fcL is lower than fcH, rather than the other way round. In this case, a range of frequencies lying between f_{cL} and fcH are removed so, again for obvious reasons, this is called a band-reject, or 'notch' filter.
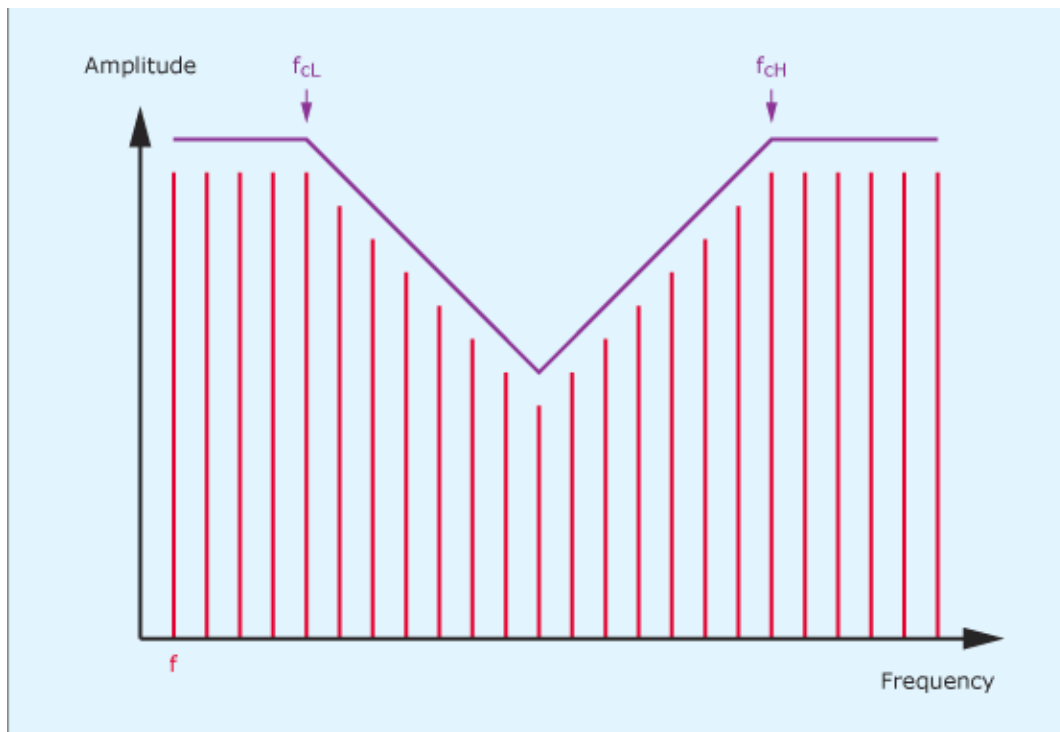
Figure 8: The action of a band-reject filter.

The slope

A filter isn't fully defined fully by its nature and its cut-off frequency (or, in the case of the band-pass and band-reject filters, its centre frequency). The 'cut-off slope' is another important attribute because it describes the degree to which the filter attenuates components at further and further distances from fc. A low-pass filter that halves the amplitude of any component lying at twice the cut-off frequency (or "2fc") is called a 6dB/octave filter because the amplitude has been attenuated by approximately 6dB an octave above the cut-off frequency. (There's not room here to explain these terms in greater detail, so please take my word for this.) Likewise, a high-pass filter that halves the amplitude of a component of ½fc is also called a 6dB/octave filter because the amplitude has been attenuated by approximately 6dB one octave below the cut-off frequency.

This figure of 6dB/octave is a bit of a magic number; far from being arbitrary, it's a consequence of the way that the universe works. However, a 6dB/octave filter is not a very powerful mechanism for altering a signal's character; it's little more than a tone control. Fortunately, it's possible to cascade such filters to create more powerful devices with slopes of 12dB/octave, 18dB/octave, 24dB/octave… and so on, and these are the filters most commonly found in analogue synthesisers or emulated in virtual analogue synths such as Thor.
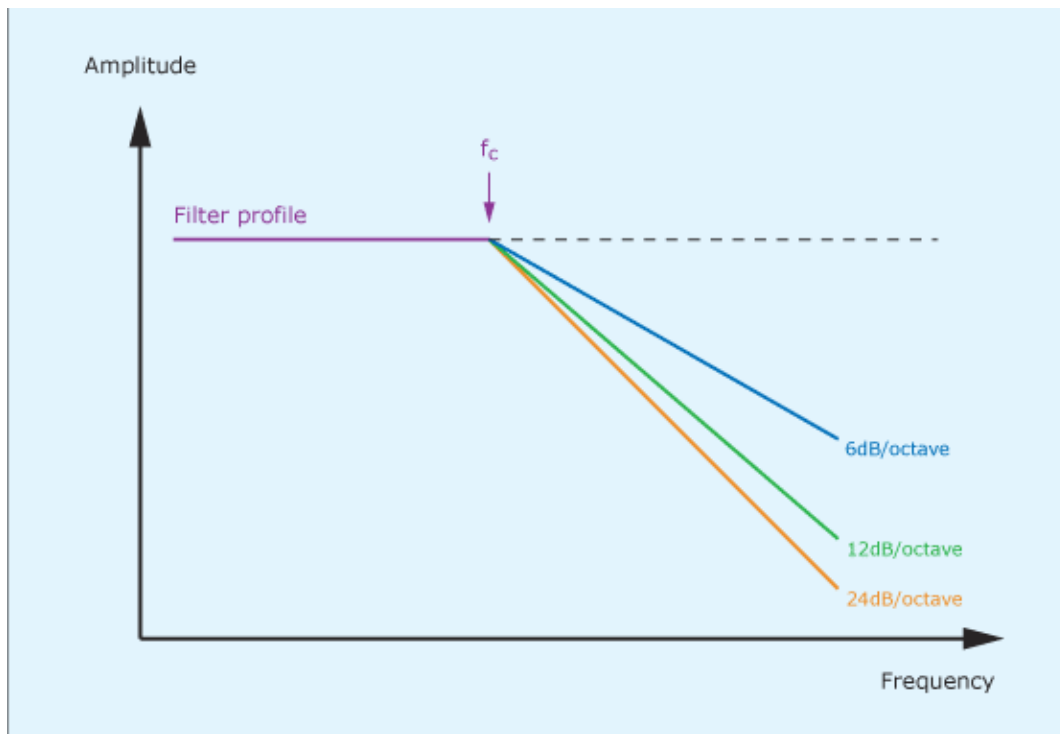
Figure 9: Low-pass filters of different cut-off slopes.

Before moving on, it's only fair to admit that this is still only an approximate description of the various filters. For example, I have drawn diagrams 5 – 9 to show the attenuation starting at the cut-off frequency, but that's wrong; the cut-off frequency is defined as being the frequency at which the signal is already attenuated by 3dB. In addition, it's too simplistic to describe a 24dB/octave filter as simply four 6dB/octave filters of exactly the same cut-off frequency. A closer representation of the truth is shown in figure 10, which shows the profile that you obtain when each of the stages' cut-off frequencies are only approximately aligned with one another. Finally, there's the issue of phase response. Analogue filters really mess with the phase relationships of the signal components, and this can drastically alter the shape of the output waveform, with all manner of subtle but potentially important consequences when creating complex synthesiser patches. Nonetheless, these are all secondary effects, so let's now put theory into practice and hear how each of these filters affects the harmonically rich sawtooth wave.
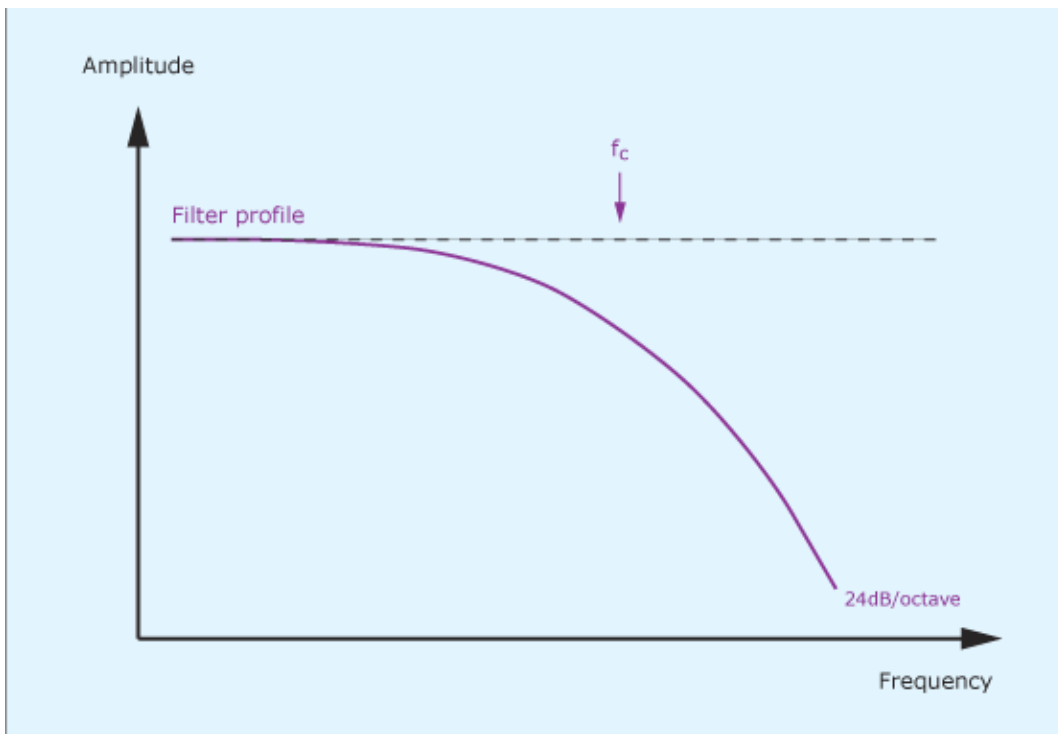
Figure 10: A better representation of a low-pass filter

Discovering these filters in Thor

We'll start with the patch in figure 11. This shows the output from a single oscillator generating a sawtooth wave directed straight to an Amp controlled by an envelope that causes it to pass sound when a key is pressed. There is no modification of the sound while the note is playing so, if you press a key, you'll obtain the easily recognisable buzz of a sawtooth wave, as you would expect.

Fig 11: An unfiltered sawtooth wave
(Click to enlarge)

Let's now load Thor's State Variable Filter into Filter slot #1 and use it to demonstrate each of the four filter characteristics described above. As you can see in figure 12, I have set the ENV (envelope follow) to its maximum, whereas the VEL (velocity follow) and KBD (keyboard follow) knobs are set to zero so that these will have no effect. The Drive level is only moderate, because overdriving the input would

create complications that would confuse the issue, and I have set the RES (filter resonance) to zero.

Note: Filter resonance is a very important characteristic of synthesiser filters that we will come to in later tutorials.

Figure 12: Setting up the filter

We'll continue by selecting the LP12 (low-pass, 12dB/octave) filter option, with the cut-off frequency ('FREQ') at its minimum. You can now use the Filter Env to apply a simple Attack/Decay (AD) contour to the cut-off frequency, thus opening and closing the filter smoothly. (Do this by switching on the GATE TRIG in the Filter Env section and raising both the Attack and Decay sliders to their maximum values, as shown in figure 13.) If you now play the same note as before, you'll hear it start with a very dull, muted character, then become steadily brighter until the unfiltered sawtooth wave is heard, and then become dull again as the filter closes.

Fig 13: A low-pass filter sweep
(Click to enlarge)

You can hear the high-pass filter equivalent by selecting the HP12 (high-pass, 12dB/octave) filter mode and playing the note for a third time. Now you'll hear the unfiltered sawtooth wave at the start of the note. The sound will then become progressively thinner as the cut-off frequency is raised and only the highest frequency components survive, before the sound becomes fuller again as the cut-off frequency returns to its minimum value.

Next, selecting the BP12 (band-pass, 12dB/octave) filter mode allows you to sweep a window (called the "pass-band") up and down the frequency axis, letting through a moderately narrow band of frequencies from the low end to the high end of the spectrum, and back again.

Finally, selecting the NOTCH or "band reject" filter (with its associated knob in the 12 o'clock position) creates a rather interesting sound as a 'hole' in the spectrum is swept up and down the frequency axis.

Using a low-pass filter to create a specific sound

You now have all the tools necessary to generate a huge range of classic and innovative synthesiser sounds. To illustrate this, I'm going to shape the response of the low-pass filter in sound #2 to create one of the staples of all synth libraries: the brass patch.

This requires only one oscillator, so we can continue to use the single Analog Osc used above. Also, as above, we'll use an almost 'square' Amp Env (figure 14) to ensure that the note speaks quickly and dies away almost immediately that you release a key. But the key to the patch is how we alter the cut-off frequency of the low-pass filter to imitate the way in which the sound of a brass instrument becomes very bright during its Attack phase, them subsides during its Decay before reaching its Sustain level.
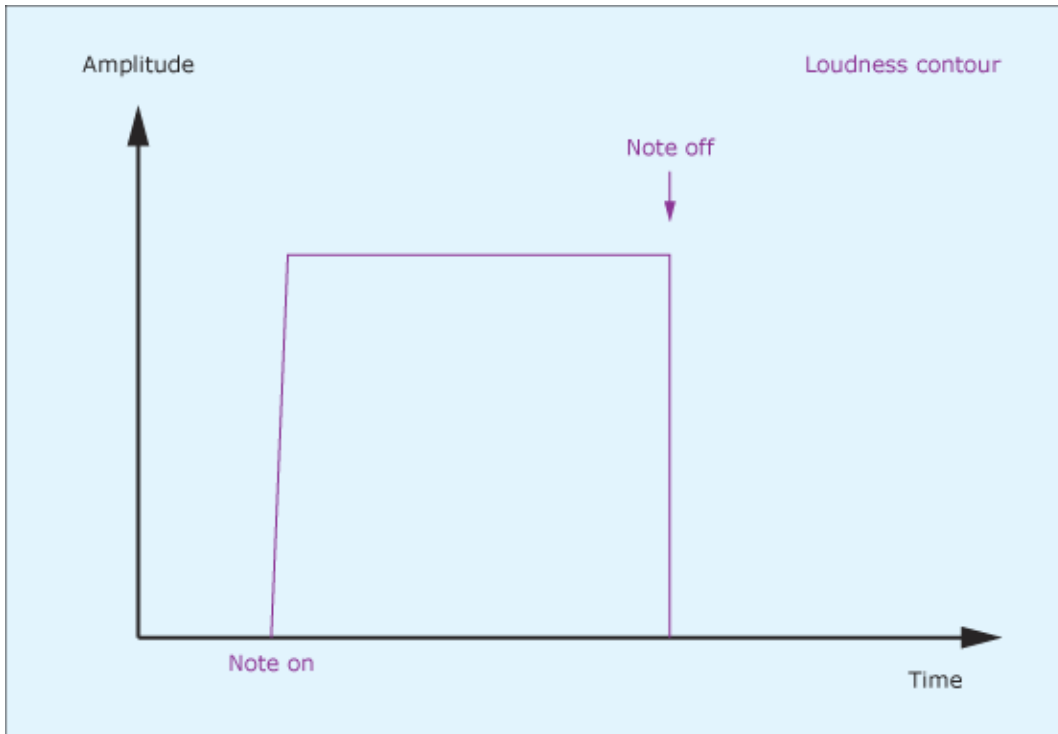
Figure 14: The loudness contour for a brass patch

Figure 15 shows the Attack and Decay stages of the filter contour that will determine the brightness at the start of the note, and the Sustain level that is then maintained until you release the key. Since the Amp Env in figure 14 returns to zero almost at the moment that you release a key, you may think that there's no point in setting the "R" (Release) in the Filter Env to anything other than zero. However, Thor's envelope generators are very rapid, and you will hear the filter snap shut if the Release is set too close to zero. (Many novices complain that the contour generators on their synths generate clicks when the Attack and/or Release controls are set to low values, not realising that this is a compliment to the designers, not a fault.) Consequently, you should set the Filter Env Release to a few tens of ms to ensure that each note ends smoothly.
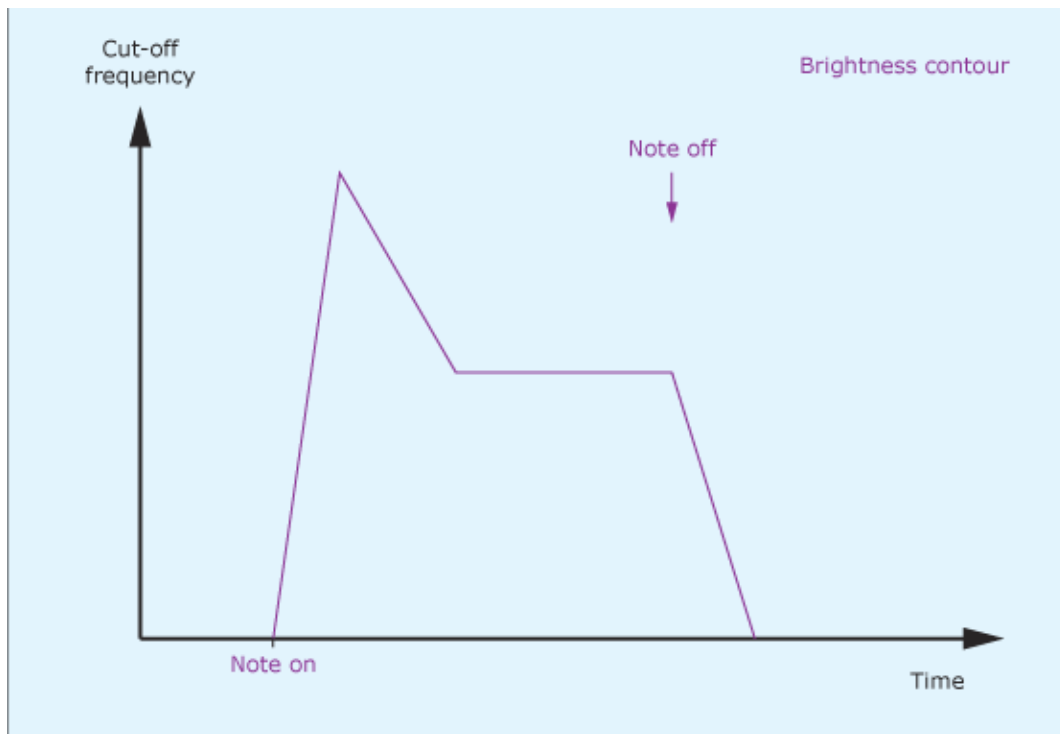
Figure 15: The brightness contour for a brass patch

Of course, the Filter Env contour will be irrelevant if the filter itself is not set up correctly, so you must ensure that its cut-off frequency is turned to its minimum, whereas its ENV knob must be near or at its maximum so that the tone is maximally affected by the Filter Env. You can hear this in sound #6, which is already beginning to sound "brassy".

Nonetheless, the sound is still rather sterile because it lacks the "rip" that occurs at the start of each note on a genuine brass instrument. You can inject this by modulating the filter cut-off frequency using a fast LFO – in this case, LFO2 running at 60Hz or more – directed to the cut-off frequency via the modulation matrix. However, you must also shape the amplitude of the modulation using the Mod Env to ensure that it's curtailed not long after the start of the note. (See figure 16.) You can see each element of this patch in figure 17 and hear the results here:
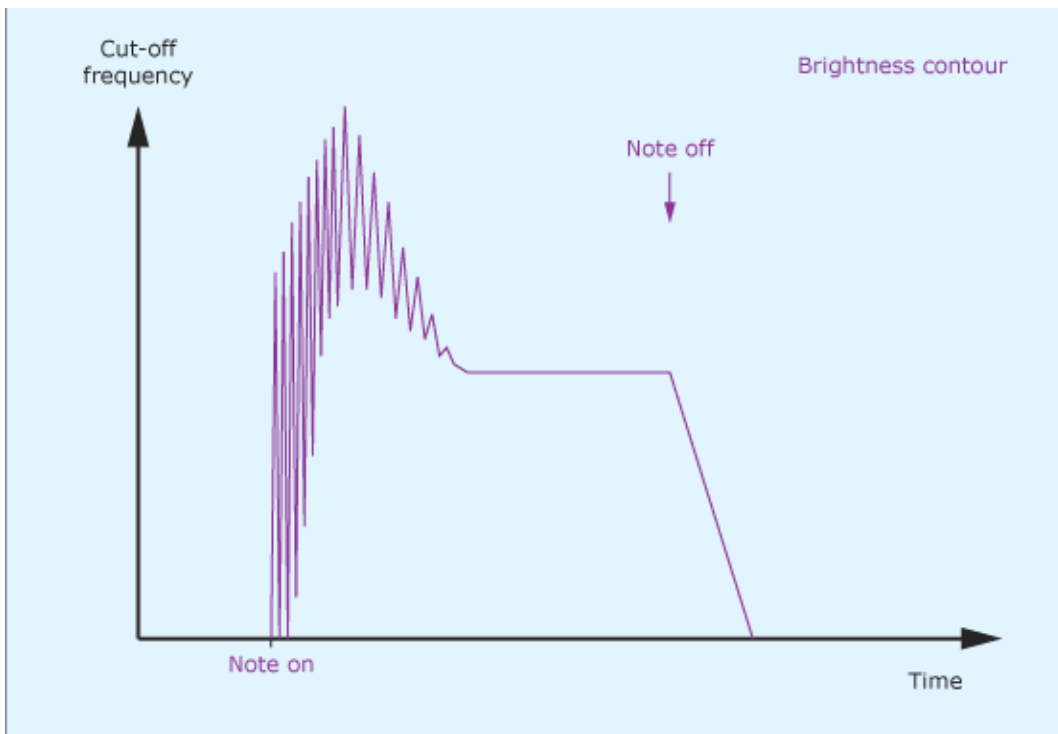
Figure 16: "Rip"

Fig 17: A Thor brass patch
(Click to enlarge)

To be honest, there are many more things that we could do to make this sound more authentic. For example, we could add vibrato. What's more, the harmonics of a real brass instrument develop at different rates after a note is blown, so we could add additional filters to emulate this, and it would also be good to add noise to emulate the breathy turbulence that occurs in a blown pipe. But doing so would take us off into realms that we're not going to address today, so all that's now necessary to play one of the classic analogue synth patches is a sprinkling of reverb and the right choice of notes and we're done. ▶

Text by Gordon Reid

Thor demystified 14: Filters pt 2: High-pass filtering

 There's one thing you'll never hear when synthesiser enthusiasts wax lyrical about their instruments: an argument about which has the sweetest or fattest high-pass filter. They'll debate endlessly the benefits of Moog's discrete component low-pass filters, argue about the pros and cons of CEM and SSM low-pass filter chips, and possibly come to blows over whether the 12dB/octave low-pass filter in the early ARP Odysseys is better or worse (whatever that means) than the 24dB/octave low-pass filter in the later models. But nobody ever got punched because they insulted someone's high-pass filter.

What's more, there was a time when you had to work quite hard to find a high-pass filter on an integrated (i.e. not a modular) synth. The groundbreaking instruments of the late '60s and early '70s - Minimoogs, ARP2600s and EMS VCS3s - didn't have them and, by and large, it was left to emerging manufacturers such as Korg, Yamaha and Roland to bring them to the public's attention.

So why is the high-pass filter such a poor relation when compared with its twin, the low-pass filter? To understand this, we again have to consider the nature of natural sounds.

Let's pluck, hit, blow into, or otherwise excite something!

Before electronic instruments were invented, non-vocal sounds were generated by exciting physical objects in some fashion. This usually involved blowing into them (brass, woodwind and organs), plucking them (guitars and harpsichords), smacking them with lumps of wood or hammers (drums, percussion and pianos) or scraping abrasive substances across them (violins, 'cellos and so on). But while all of these instruments are very different and can sound very different, they all abide by the laws of physics, which means that they share a number of important characteristics, two of which I'll outline here.

Charateristic #1:

Let's imagine that you excite any one of the above by plucking it gently, blowing into it softly, hitting it lightly, or whatever is appropriate. The resulting spectrum may look like figure 1, in which the lowest signal components are prominent, and the higher ones contribute little or nothing to the tone.
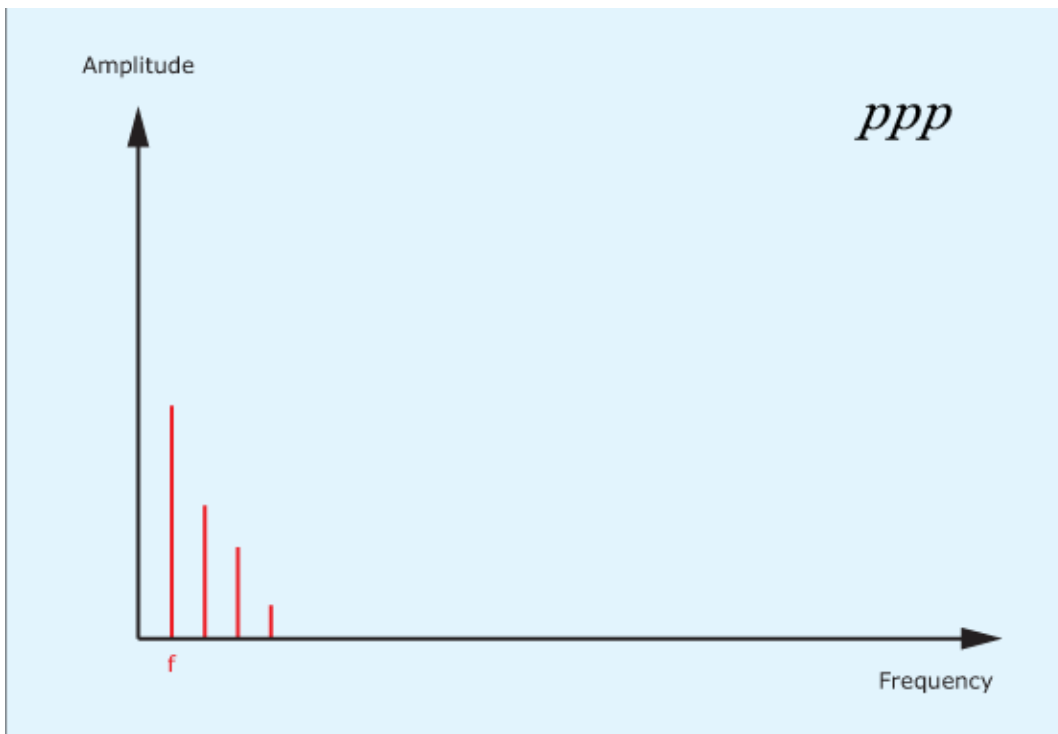
Figure 1: A ppp signal spectrum

Now let's imagine that you pluck, blow or hit it somewhat harder. In almost every case (with a few esoteric exceptions) not only will the previous signal components be louder, but higher frequency components will be introduced, so this might result in the spectrum shown in figure 2.
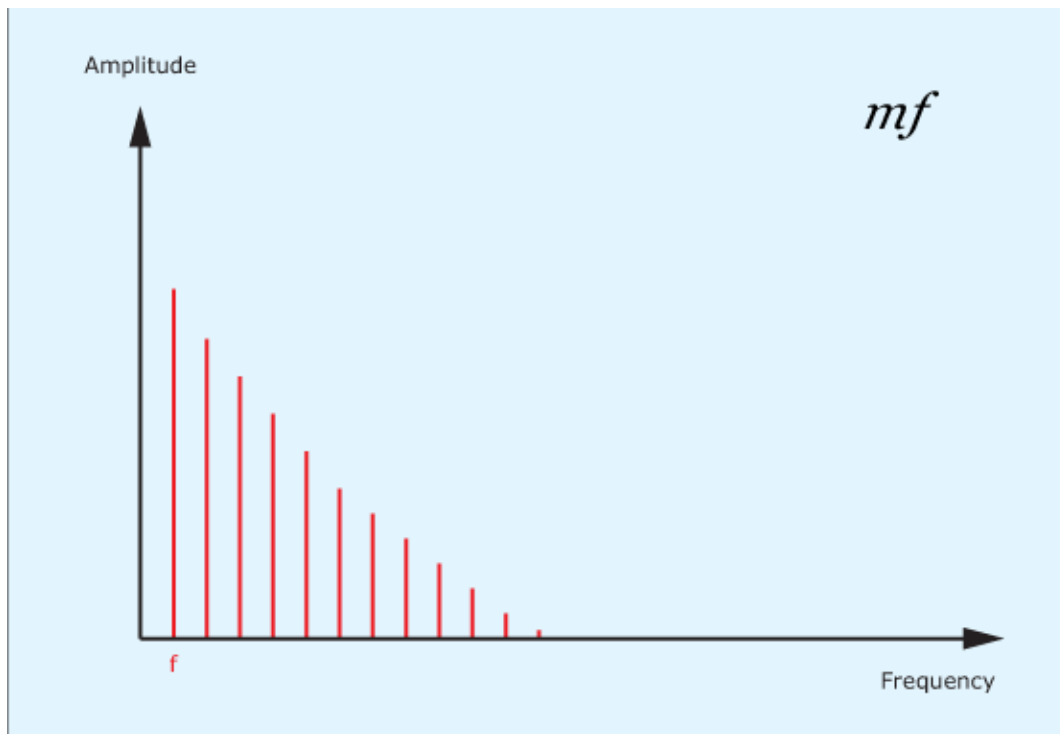
Figure 2: A mf signal spectrum

Finally, let's go a step further and excite the thing really energetically. As you would expect, the instrument becomes louder still but, in addition to this, even more high frequency components are introduced. (See figure 3.)
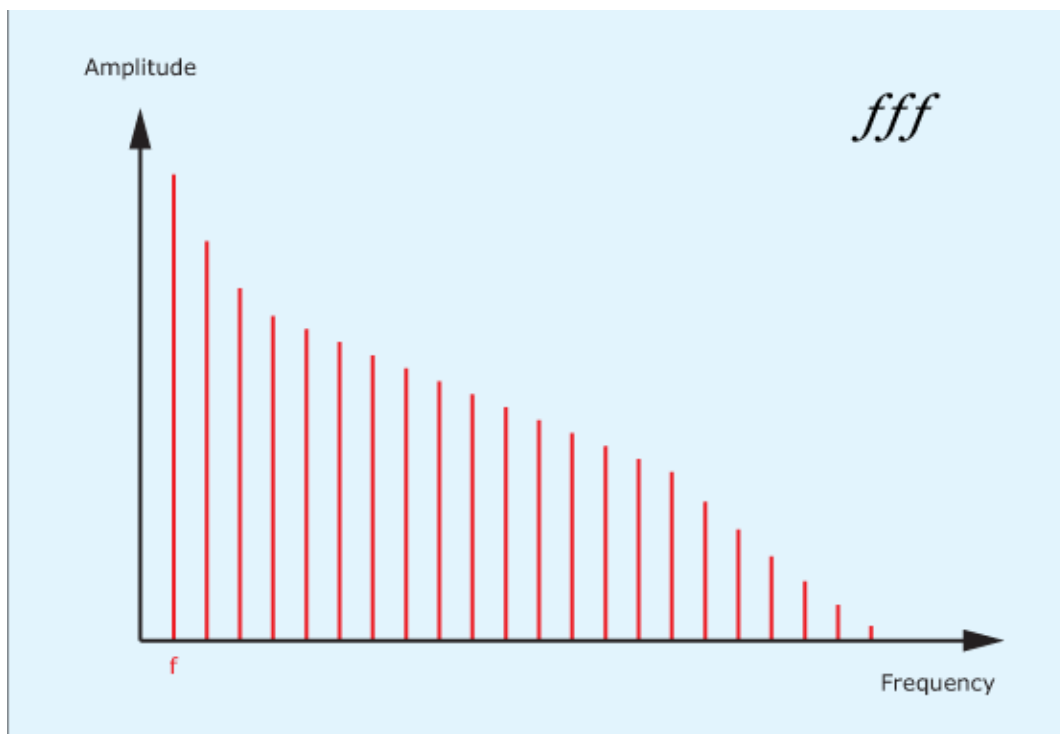
Figure 3: A fff signal spectrum

This means that, the louder you play an acoustic instrument, the greater the proportion of the sound is contained within high frequency components. In other words, when the sound of an acoustic instrument becomes louder, it also becomes brighter.

Characteristic #2:

Let's now imagine that you have played a note on an instrument with a nice, long decay - say, a bass note on a piano &ndash and that you are listening as it fades gently to silence. At the moment that you play the note, it will be bright to the degree determined by how hard you hit it. Consequently the spectrum (ignoring enharmonics and hammer noise) might look like figure 4.
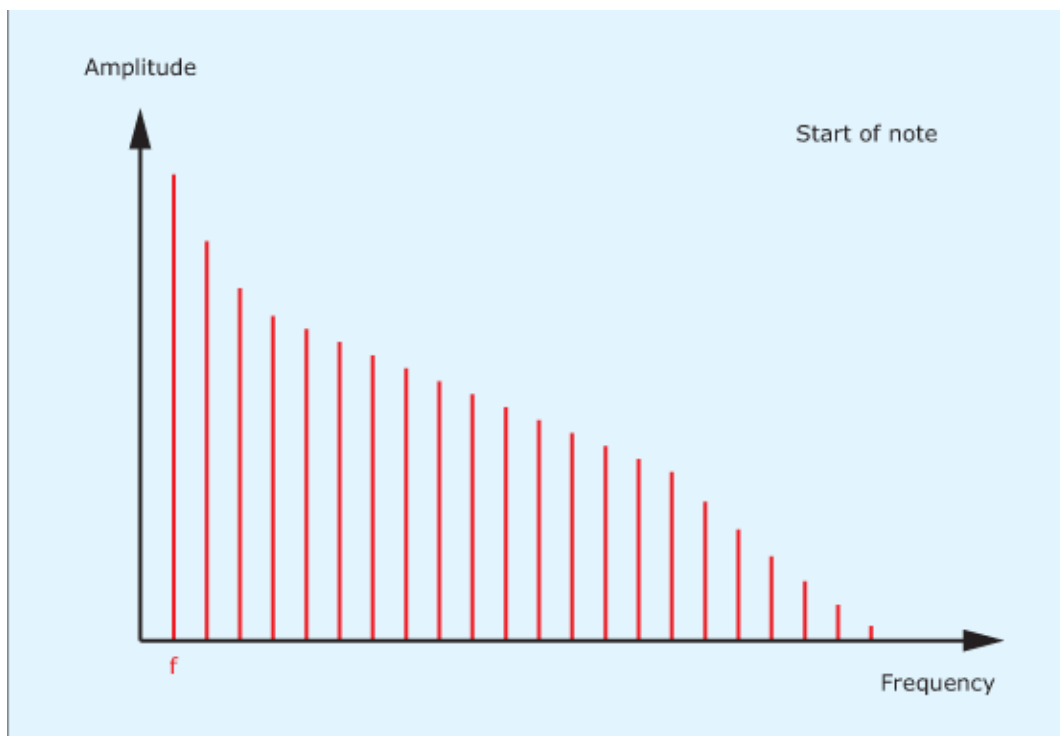
Figure 4: A simplified representation of the spectrum at the start of a note

Now hold the note and listen carefully as time passes. You will hear that the highest frequency components in the sound fade away quite rapidly and that the note quickly becomes warmer and rounder. (See figure 5.)
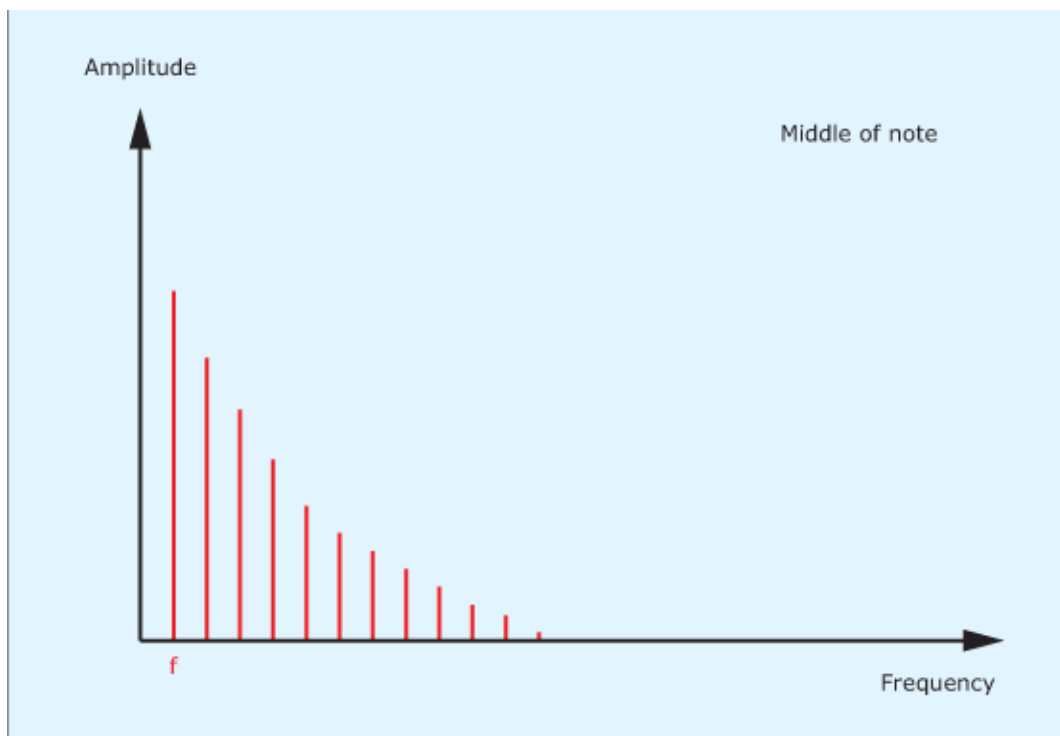
Figure 5: The spectrum in the middle of the note's duration

Finally, the middle frequencies will dissipate, and before the note decays to silence you'll be left with just a deep, rolling bass sound (see figure 6) with few if any overtones. This is true of all acoustic instruments: the energy in the high frequencies components is dissipated more rapidly than the energy in the low ones.
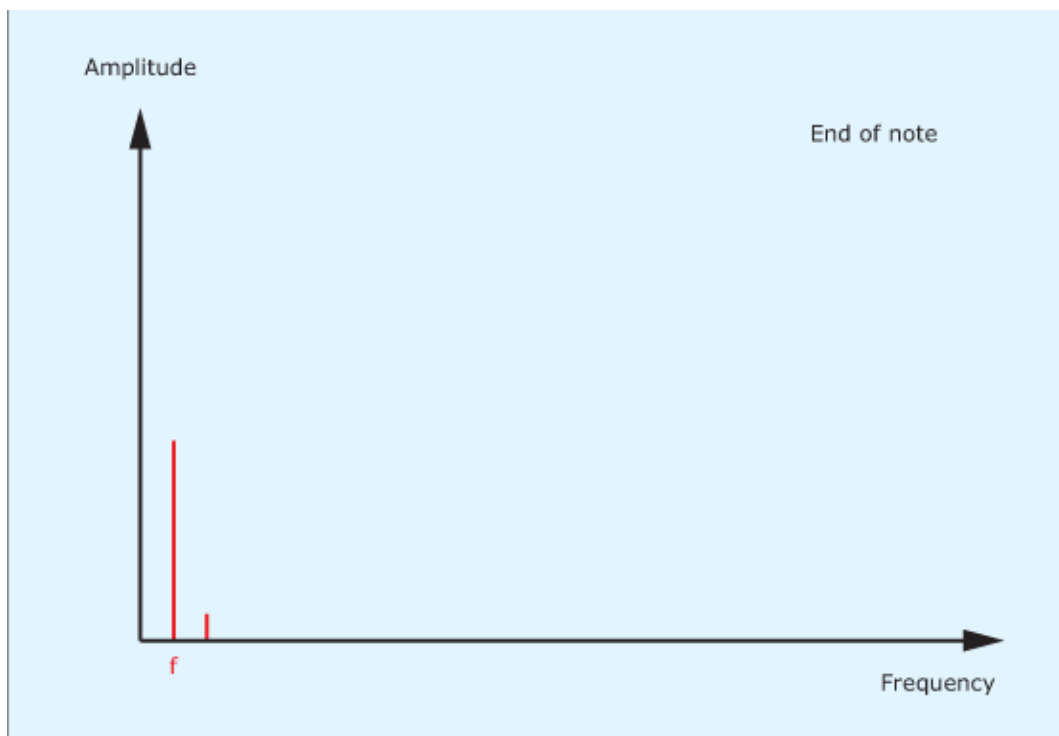
Figure 6: The spectrum at the end of the note

These examples explain why the low-pass filter lies at the core of a traditional, subtractive synthesiser: if you start with a bright waveform that contains all the signal components you might ever need, you can use the filter to attenuate or even eliminate the higher frequencies in ways that emulate sounds in the real world. Characteristic #1 is achieved by making the filter's cut-off frequency velocity-sensitive. Characteristic #2 is achieved by controlling the cut-off frequency using a contour generator. Other characteristics can be achieved by making the cut-off frequency proportional to the pitch of the note, or by modulating it using a low-frequency oscillator… and so on.

So it's clear that the low-pass filter is an essential element in synthesising many sounds. But what of the high-pass filter? Is there no equivalent explanation that shows that we can't live without them? In truth, there are some instruments in which the amplitudes of the low frequency components alter according to how you play them, but these are minor effects when compared with the changes in the high frequency components, so many of the greatest synths ever built don't bother with high-pass filters at all.

Thor demystified 15: Filters pt 3: Resonance

DISCOVERING REASON Most physical objects vibrate at frequencies determined by their size, shape, materials and construction, and the specific frequencies for each object are known as its resonant frequencies. Nonetheless, simply adding energy to an object doesn't guarantee that you'll obtain an output. Imagine an object having a single resonance of 400Hz placed in front of a speaker emitting a continuous note at, say, 217Hz. If you can picture it, the object tries to vibrate when the sound first hits it, but each subsequent pressure wave is received at the 'wrong' time so no sympathetic vibration is established. Conversely, image the situation in which the speaker emits a note at 400Hz. The object is now in a soundfield that it is pushing and pulling it at exactly the frequency at which it wants to vibrate, so it does so, with enthusiasm.

This suggests a simple rule: if an object is excited at one of its resonant frequencies, it will vibrate in sympathy; if it is excited at a different frequency, it won't. But what might the situation be if this object found itself in a soundfield that had a frequency of 399.9Hz or 400.1Hz? Would it refuse to vibrate, or would the excitation be close enough to 400Hz to establish a certain amount of sympathetic vibration? Depending on the nature of the object, it would indeed vibrate to some degree, and there is a mathematical term (called 'Q') that describes the relationship between the excitation (the input signal), the resonant frequency of the object, and the amplitude of the sympathetic vibration (the output). If the Q is low, the range of excitation that can elicit a response is wide; it the Q is high, the range is narrow. (Figure 1.)
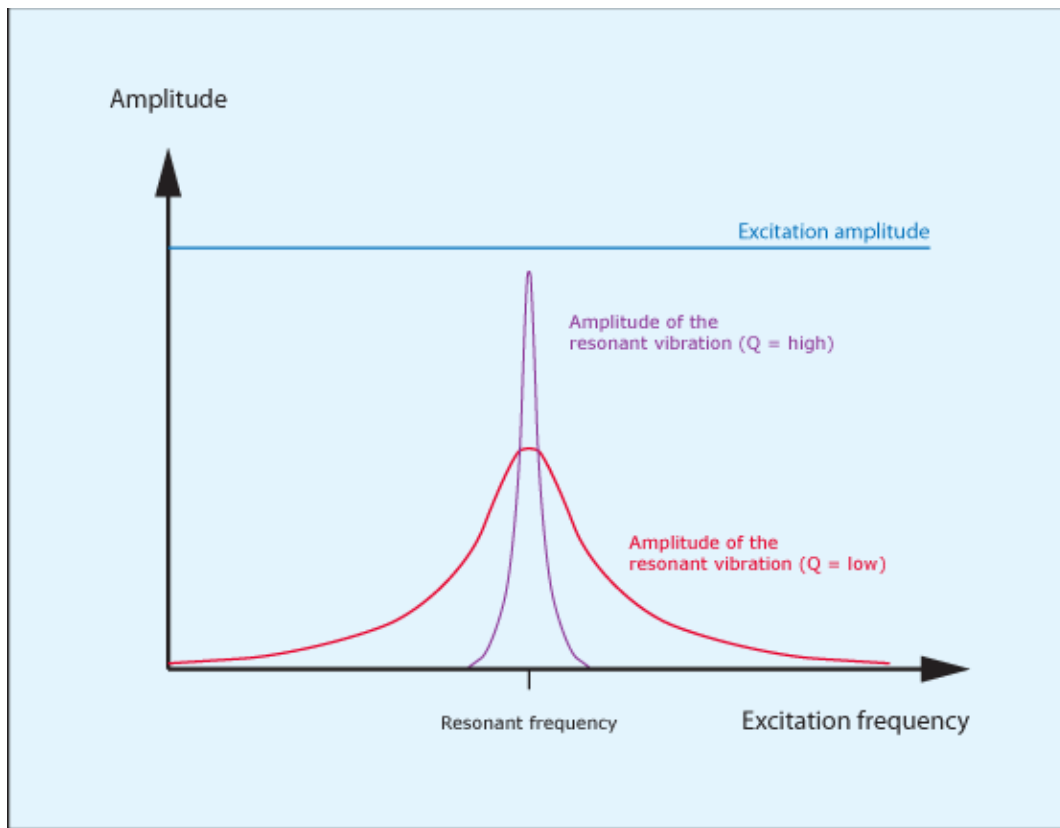
Figure 1: The resonant response of a physical system

Resonant filters

Interestingly, it's not only physical objects that resonate; analogue circuits and digital algorithms can do so too, again creating a bump in their responses at their resonant frequencies. Indeed, there is a special type of filter called a Peaking Filter that does nothing but add a resonant peak to the signal spectrum, and you can find one of these in Thor's State Variable Filter. However, this is a rare beastie, and you're much more likely to encounter resonance in low-pass and high-pass filters, in which the resonant frequency is the same as the cut-off frequency, (There are good reasons why this should be so, but we won't discuss them here because, as someone once said to me, "It's obvious; it's the solution to a second order differential equation." Umm... right!)

We can represent the response of a typical, resonant low-pass filter as shown in figure 2 and of a resonant high-pass filter as shown in figure 3. As you can see, the filters still attenuate the frequencies significantly above (LPF) or below (HPF) their cut-off frequencies, but in both cases a band around the cut-off frequency is boosted.
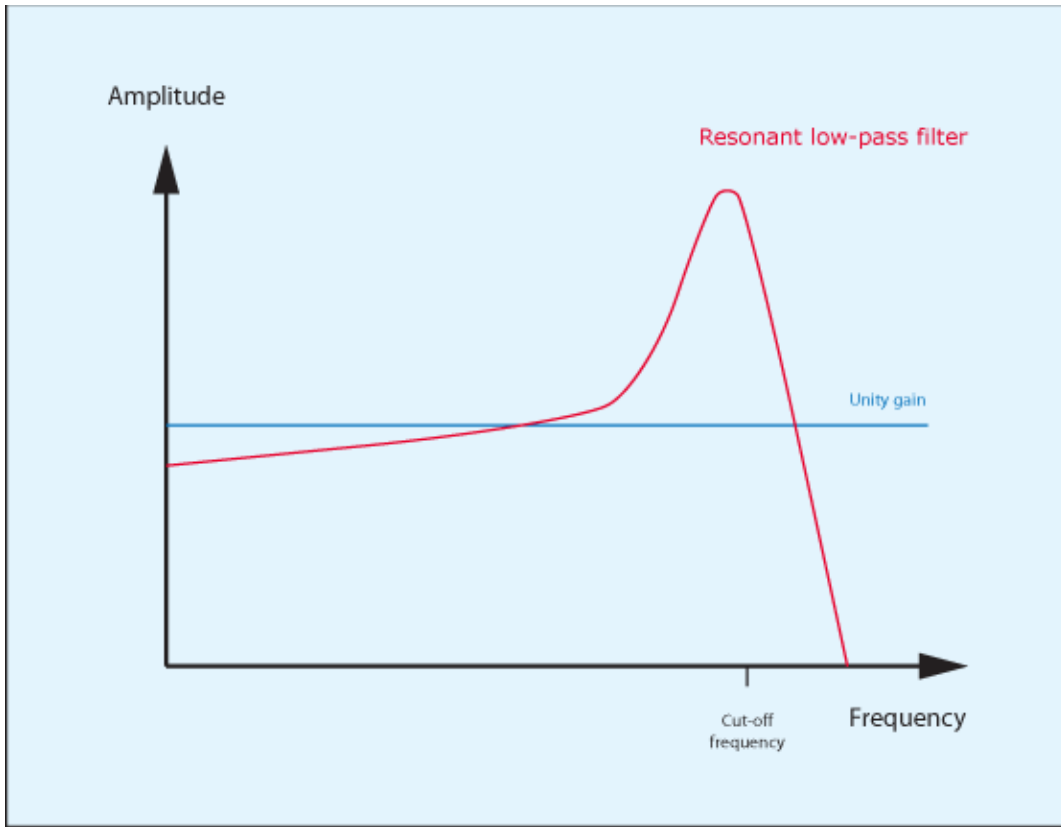
Figure 2: The typical response of a resonant low-pass filter
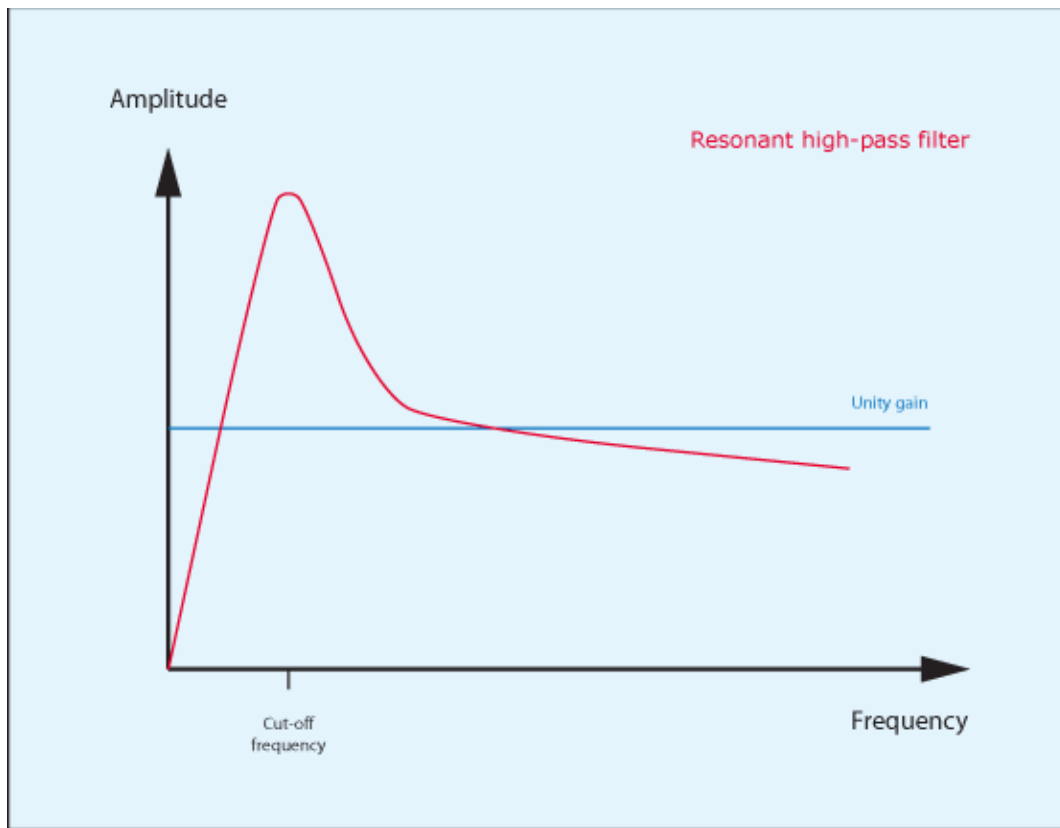
Figure 3: The typical response of a resonant high-pass filter

You might think that you could describe this response using two parameters – the cut-off frequency and the gain of the resonant peak – but there's another attribute to consider – the 'Q' which, as before, describes the width of the peak. In fact, the width and the gain of a the resonance in a synthesiser's filter are so closely related that, all other things being equal, a low Q means that the peak will be broad and low, whereas a high Q means that the peak will be narrow and high (figure 4) so only two controls are needed to control them: the cut-off frequency and the Q – or the "amount of resonance".
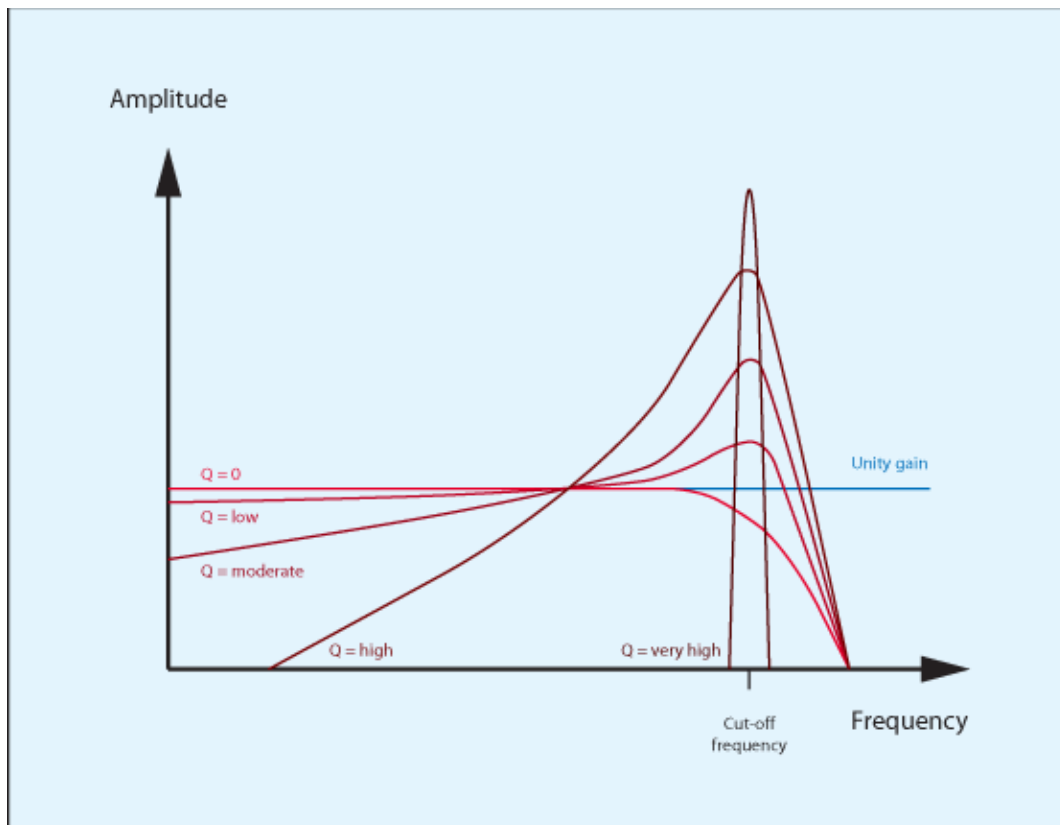
Figure 4: The response of a LPF at various Qs

To illustrate the action of a resonant low-pass filter upon the spectrum of an harmonic audio signal, let's consider one that I introduced in the previous tutorial, and which is shown again in figure 5. Passing this signal through a low-pass filter with a moderately high Q might result in the spectrum shown in figure 6. If the Q is increased further, you might obtain the spectrum shown in figure 7, which represents an enormous change in the nature of the sound. Clearly, the resonant filter is a much more powerful sound-shaping tool than the non-resonant low-pass and high-pass filters described in the previous two tutorials.
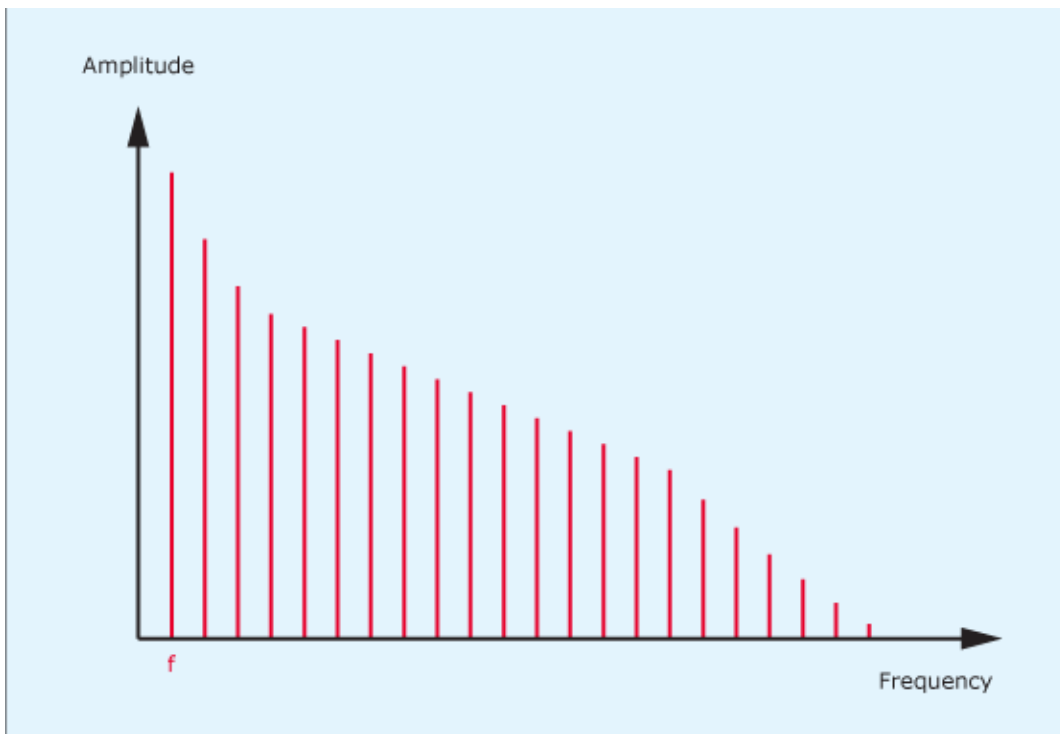
Figure 5: An harmonic signal spectrum
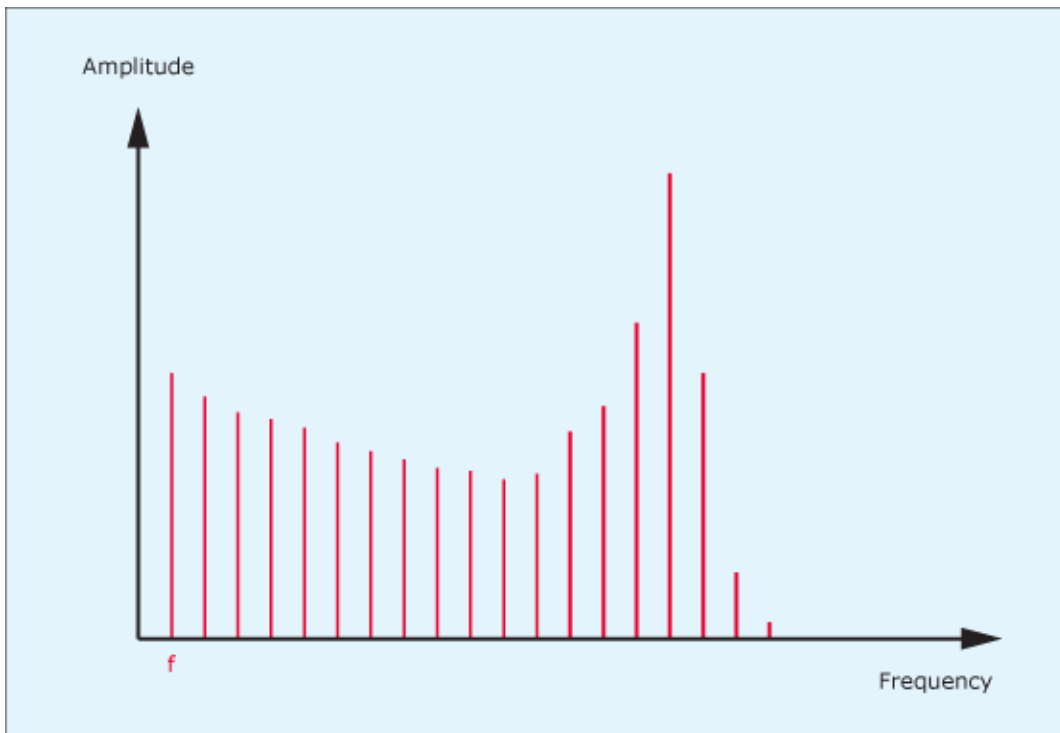
Figure 6: The spectrum after passing through a low-pass filter with moderately high Q
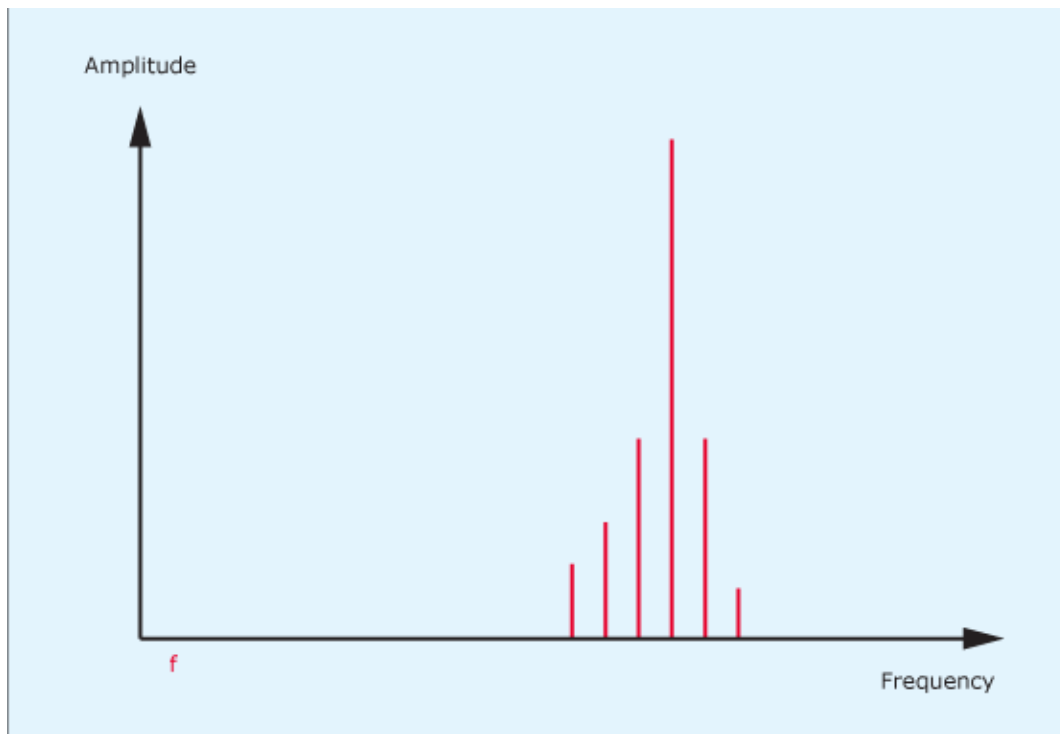
Figure 7: The spectrum after passing through a low-pass filter with very high Q

Using resonance to create sounds in Thor

In the first of these filter tutorials, I demonstrated how you could open and close one of Thor's low-pass filters with a simple AD contour (figure 8) to make a single note change from a dull tone at the beginning of a note, to a much brighter one as the note progressed, and then back to the original tone at the end.

Figure 8: Creating a low-pass filter sweep
(Click to enlarge)

Now, without making any other changes to the patch, I'll add filter resonance so that any of the note's harmonics lying close to the cut-off frequency will be accentuated as the filter sweeps opens and then closes. (Figure 9.) Sound #2　　was recorded with a resonance of 113, while sound #3　　had a resonance of 121.

Figure 9: Adding resonance to the low-pass filter sweep

Armed with resonance, you're now in a position to adjust the envelope and filter parameters to create all manner of filter sweeps, blips and other popular 'synth' sounds. For example, the simple bass line in sound #4　　was created by taking sound #3 and speeding up the Filter Env sweep by setting the Attack to zero and the Decay to around a second. Fortunately, patches that use filter sweeps can be far more interesting than this. Sound #5　　uses three sawtooth oscillators with one tuned an octave above the other two, all of which are detuned from one another to add 'thickness' to the sound. Then, as in the previous example, the Filter Env causes the filter's cut-off frequency to reinitialise at its maximum value each time that you press a key, and to sweep downward during the course of the note to create its instantly recognisable character. (See figure 10.)

Figure 10: A famous "filter sweep" sound from 1972
(Click to enlarge)

Interesting though this sound is, the Minimoog-style architecture of the patch (I'm sure that you guessed) would soon prove to be rather limited for creating some of the more complex sounds that rely on filter resonance. But in 1973 Korg introduced a series of affordable little synthesisers that offered not one but two resonant filters – a HPF followed by a LPF. In truth, the factory sounds for these synths (i.e. the printed patch charts that came with their manuals) made very little use of their resonant high-pass filters, but if they had one signature sound, it was the "analogue voice" that their dual-filter architecture made possible.

The human voice is defined in part by the presence of resonances known as formants. The positions of these and the way that they change during speech and singing are unique to each individual, which is why you can recognise people's voices. Unfortunately, the smallest number of synthesised formants that can contribute to recognisable speech is three, but two can still recreate a semblance of a vocal sound. So, long before the appearance of digital synths with choir samples, we old-timers used dual HP- and LP- filters to create all manner of "synthvox" voices based upon the combined filters' response shown in figure 11.
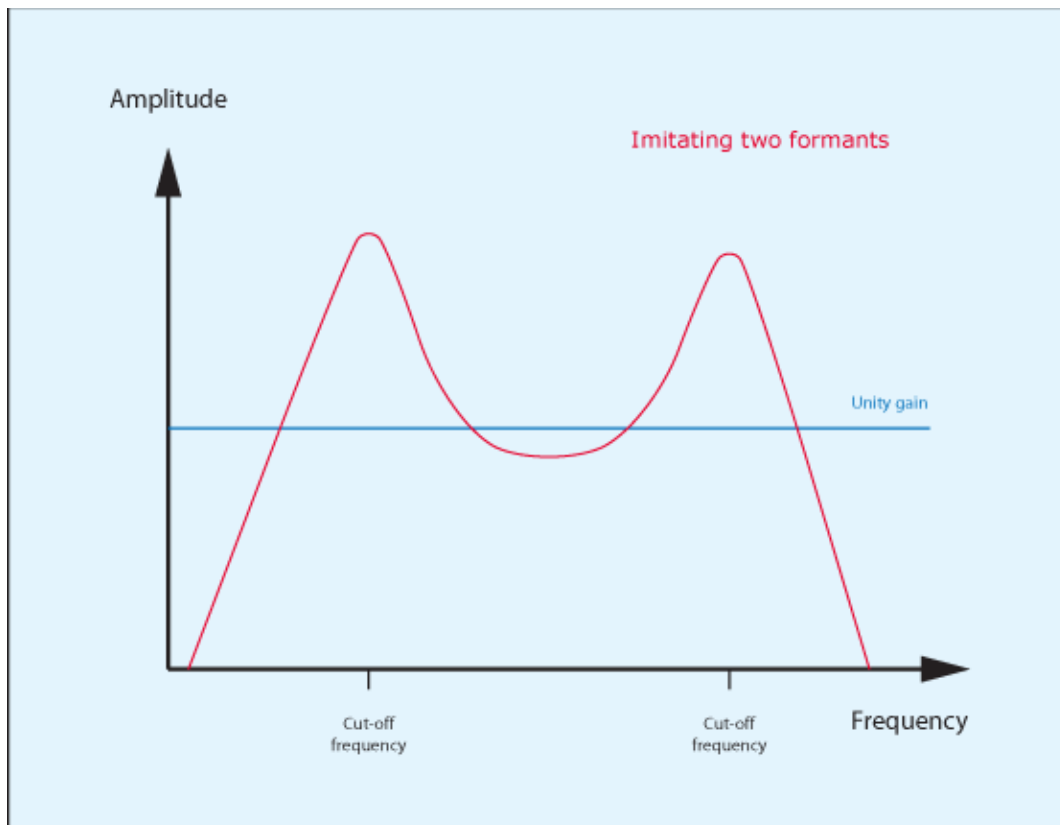
Figure 11: Dual resonant HP- and LP- filters

To create a synthvox patch, start by placing a single analogue oscillator in Thor's Osc 1 slot, select the pulse wave and set its width to somewhere around 14. Next, create a gentle ASR amplitude contour using the Amp Env. Now place an HP12 filter in the Filter 1 slot and a LP12 filter in the Filter 2 slot, and set up the routing appropriately. With the filters wide open you obtain exactly what you would expect: the raw buzz of an untreated, narrow pulse wave   . Next, add some vibrato by using the modulation matrix to direct the output from LFO1 to the oscillator pitch. Ah... a wobbly buzz that sounds like a cheap combo organ   . Now filter the sound. Set the cut-off frequency of the high-pass filter to around 600Hz, and that of the low-pass filter to around 900Hz. Ah... a dull, wobbly buzz!   Now let's do the clever bit and increase the resonance of both filters. I've chosen values that are high, but not at the very top of the range because this would make the sound 'ring'. With the HPF resonance at 115 and the LPF resonance at 117 you'll obtain sound #9   , which is very similar to the analogue 'voices' of the early 1970s. (See figure 12.) Finally, add a bit (well, a lot) of an external ensemble effect and some reverb to obtain the ensemble in sound #10   . Now you need only play the right notes to invoke the Electric Monks in sound #11   .

Figure 12: Electric Monks
(Click to enlarge)

There are dozens of other classic sounds to be discovered when you start experimenting with multiple resonant filters, including emulations of musical instruments that cannot be synthesised by simple attenuation of high or low frequencies. At the other end of the spectrum (no pun intended) the quality of the bass sounds created by the Minimoog is a consequence of its unusual resonant response, and understanding this makes it possible to create much better emulations of this classic synth. In short, the amount of resonance is dependent upon the cut-off frequency, and there's no emphasis when its filter cut-off frequency is below 150Hz or thereabouts. Happily, Thor allows you to imitate this by connecting the Note (Full Range) to the Filter Res in the modulation matrix... but I'll leave you to try this for yourself.

Thor demystified 16: Filters pt 4: Comb Filters

DISCOVERING REASON When we talk about an audio signal generated by an analogue (or virtual analogue) oscillator, we often describe it using three characteristics: its waveform, its frequency, and its amplitude. These, to a good approximation, determine its tone, its perceived pitch, and its volume, respectively. But there is a fourth characteristic that is less commonly discussed, and this is called the 'phase' of the signal.

Consider the humble 100Hz sine wave. You might think that this can be described completely by its frequency and its amplitude and, in practice, this is true provided that you hear it in isolation. But now consider two of these waves, each having the same frequency and amplitude. You can generate these by taking a single sine wave and splitting its output, passing one path through a delay unit as shown in figure 1. If no delay is applied, the two waves are said to be 'in phase' with one another (or, to express it another way, they have a phase difference of 0°) and, as you would imagine, you could mix them together to produce the same sound, but louder.
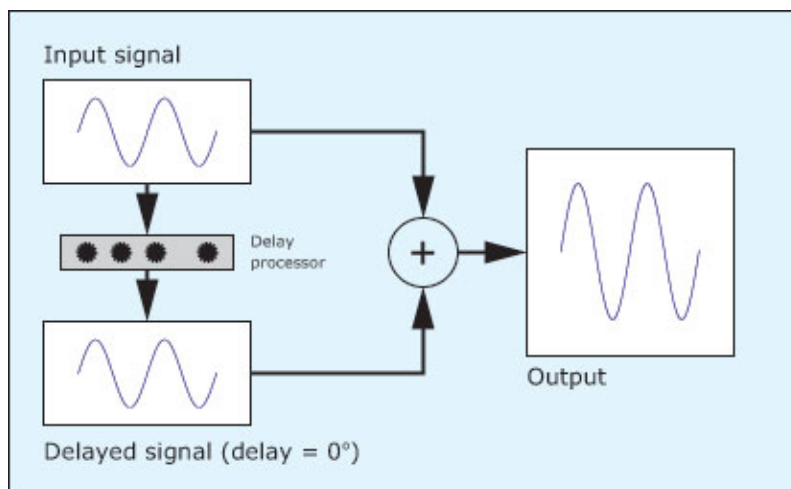
Figure 1: Adding in-phase sine waves

Now let's consider what happens when the waves are 'out of phase', with one climbing away from 'zero' at the same time as the other drops away, as shown in figure 2. In this case, the second waveform is offset by half a cycle (a phase difference of 180°) with respect to the first and, if you mix them, they cancel each other out. In isolation, both waves sound identical, but mixing them results in silence. Given that this signal has a frequency of 100Hz, its period is 10ms. Consequently, the offset between the two waves

in figure 2 (half a period) is exactly 5ms. This is a remarkable result because it tells you that, for a single frequency, a precise delay and a mixer can act as a perfect filter!
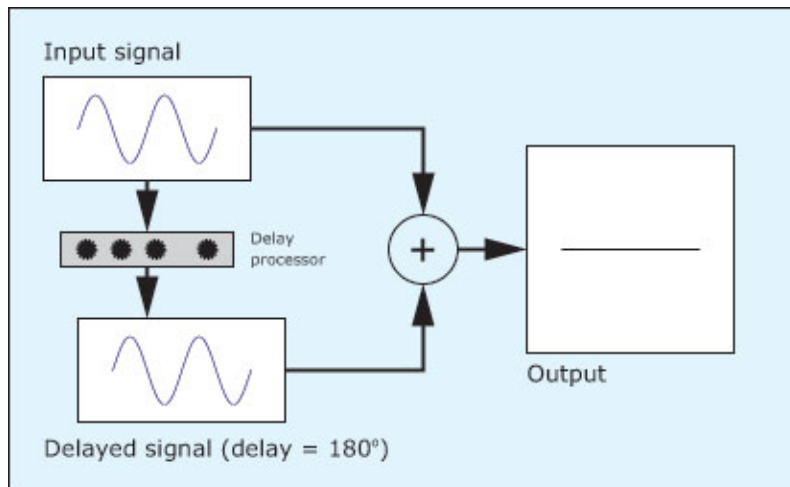
Figure 2: Adding out-of-phase sine waves

The results illustrated in figures 1 and 2 demonstrate perfect addition and perfect cancellation of a 100Hz sine wave but, if you combine the waves with phase differences other than 0° or 180°, you obtain different degrees of addition or cancellation. As you sweep the relative phase of the two signals from 0° to 360° the output from the system varies as shown in figure 3.
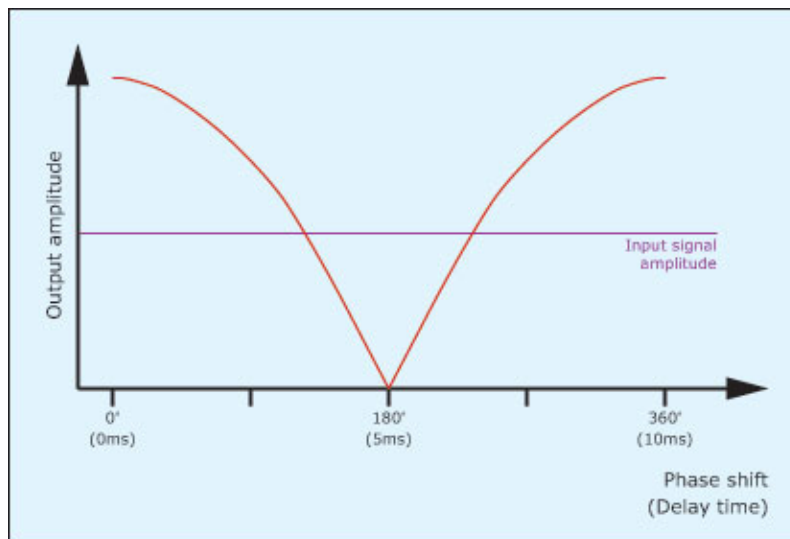
Figure 3: The output amplitude obtained as the phase difference is swept from 0° to 360°

So far, so good… A phase difference of 180° results in cancellation while a phase difference of 0° or 360° results in maximum reinforcement. But there's nothing stopping us

from applying delays that result in phase differences greater than 360º, and a phase difference of 361º has the same effect as one of 1º, 362º has the same effect as 2º… and so on ad infinitum. So, if you take a sine wave, split it into two paths, apply a delay to one path and mix the signals again, the sound will come and go as you adjust the delay time. In the case of the 100Hz sine wave, cancellation occurs with delays of 5ms, 15ms, 25ms…, and maximum reinforcement occurs with delays of 10ms, 20ms, 30ms… and so on. (Figure 4.)
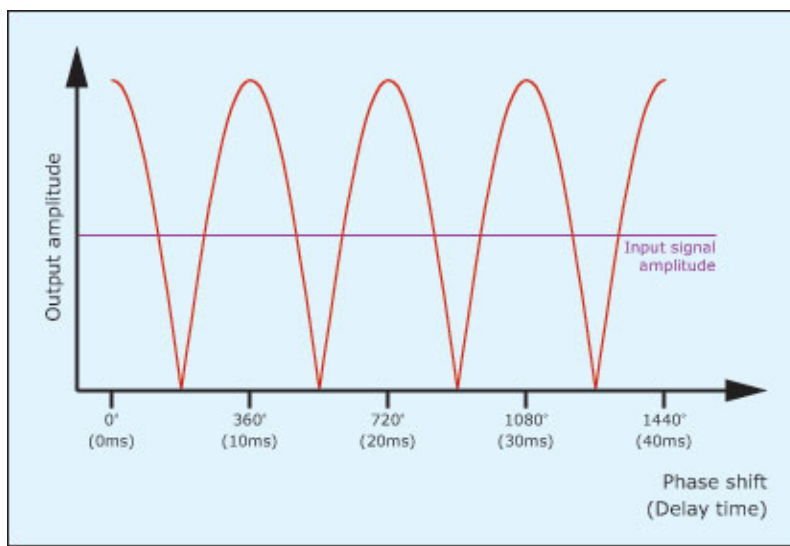
Figure 4: Extending the delay time beyond one cycle

Now let's turn this concept on its head. If there are many delay times that will result in cancellation of a 100Hz signal, it seems reasonable to surmise that there are many frequencies that will be cancelled by a given delay time. This turns out to be true, and the lowest frequency cancelled by a given delay "D" expressed in seconds is 1/2D. The next cancellation occurs at 3/2D, the next at 5/2D, the next at 7/2D and so on. Returning to our example, therefore, a delay of 5ms generates a fundamental cancellation at 100Hz, but it also cancels signals lying at 300Hz, 500Hz, 700Hz… and so on, all the way up through the spectrum, with frequencies lying between these figures attenuated as shown in figure 5. The resulting filter response looks much like a broad toothed comb.
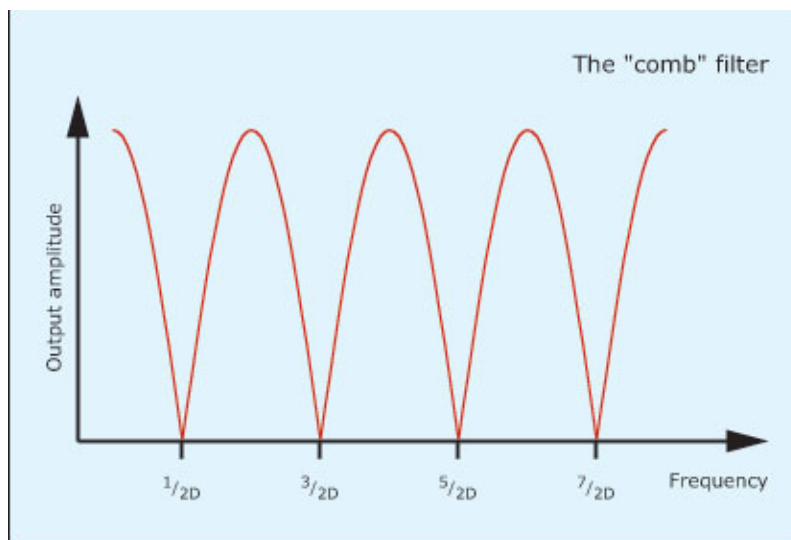
Figure 5: The comb filter

Of course, sine waves are rather special cases, but complex signals such as music and speech can be represented by an infinite number of sine waves that describe all the frequencies present. So, for a given delay, every signal component will be reinforced or attenuated according to its frequency and the delay applied, and the resulting holes in the output spectrum are what give the comb filter its instantly recognisable character.

Finally, let's consider what happens when the delay time is not constant but fluctuates in some fashion. In this case, the holes in the spectrum will sweep across the spectrum in some fashion, creating effects that include flanging, phasing and chorusing. Furthermore, these and other effects can be further reinforced by positive feedback (called 're-generation' or 'resonance') to create some of the most interesting sounds and effects obtainable from subtractive synthesis.

I've discussed the theory behind comb filters in a bit more depth than usual because, when you know *how* they do *what* they do, you can design some amazing sounds using them. But you probably won't stumble upon these by accident.

DISCOVERING REASON

Thor demystified 17:
Filters pt 5: Formant Filters

The final filter in Thor's armoury is a rather special one named a Formant filter, so-called because it imposes formants on any signal passed through it. But what are formants, and why would you want to impose them on anything?

Let's start to answer this by reminding ourselves of the four types of filters most commonly found in synthesizers. These are the low-pass filter (figure 1) the high-pass filter (figure 2) the band-reject or 'notch' filter (figure 3) and the band-pass filter (figure 4). Our journey into formant synthesis begins with the fourth of these.
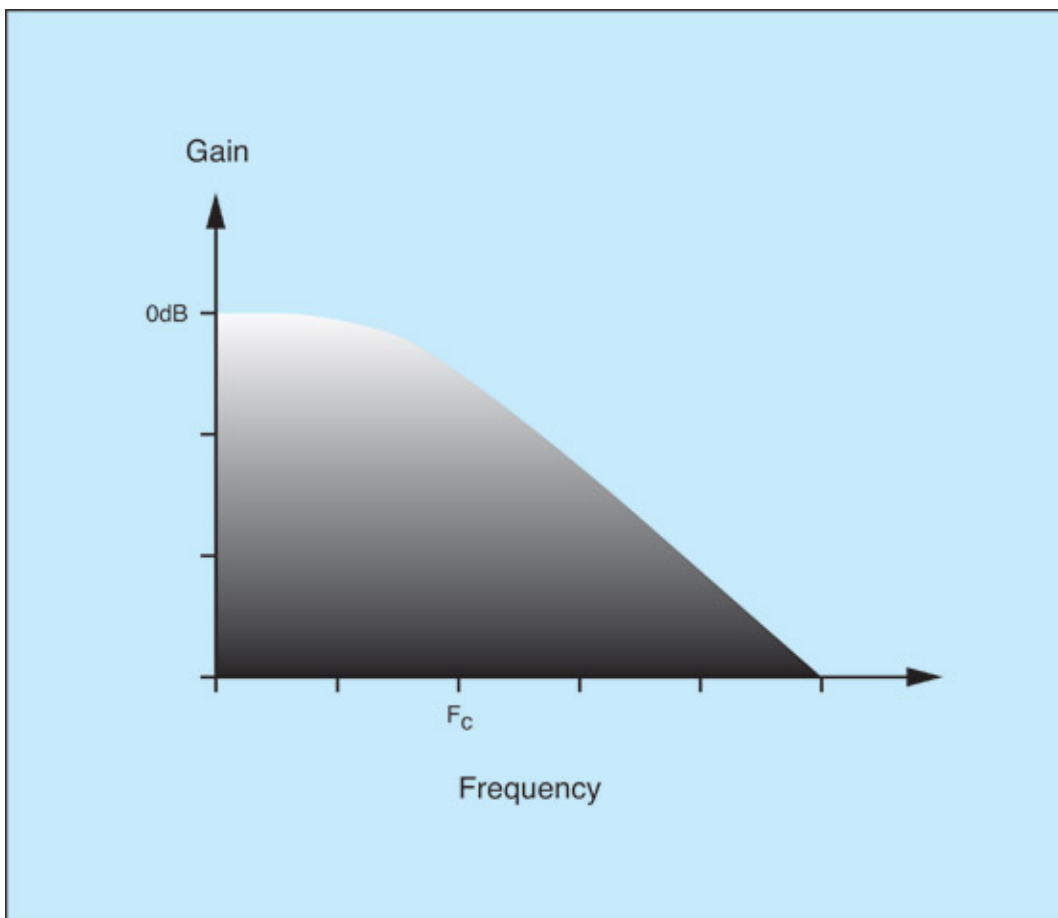
Figures 1: low-pass filter
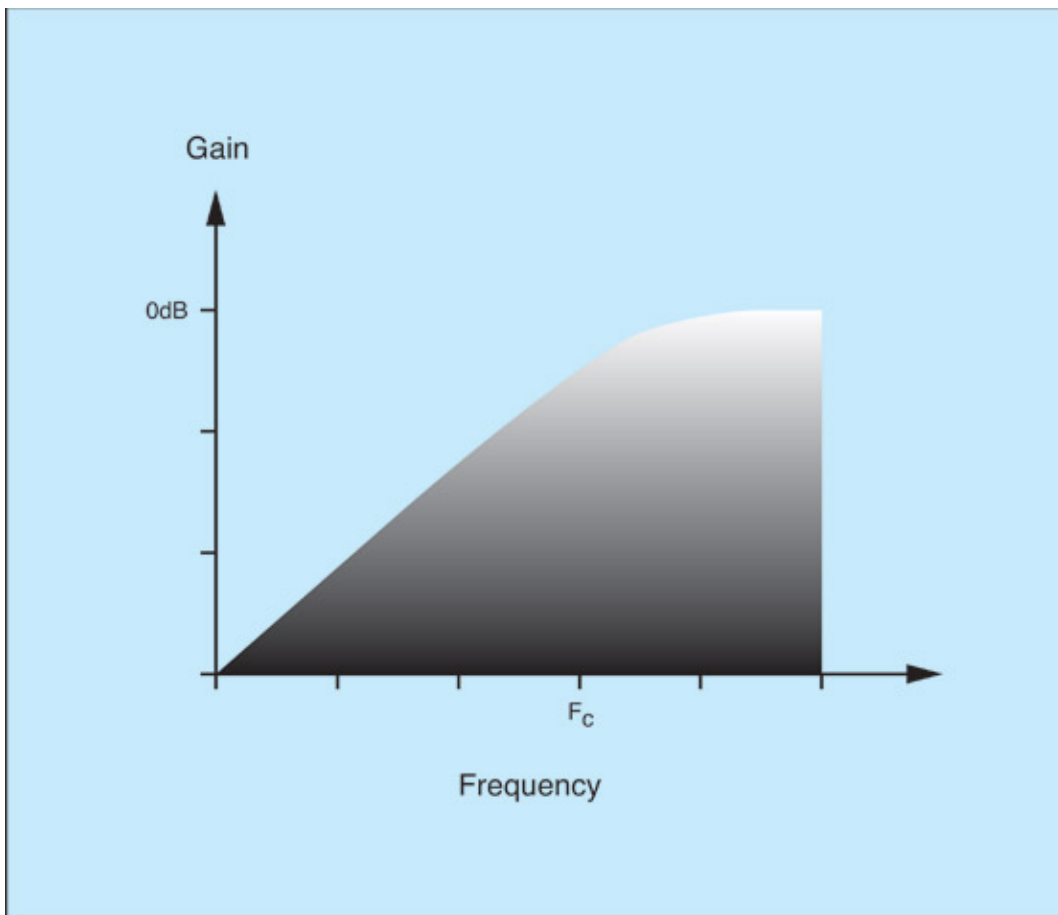
Figures 2: high-pass filter
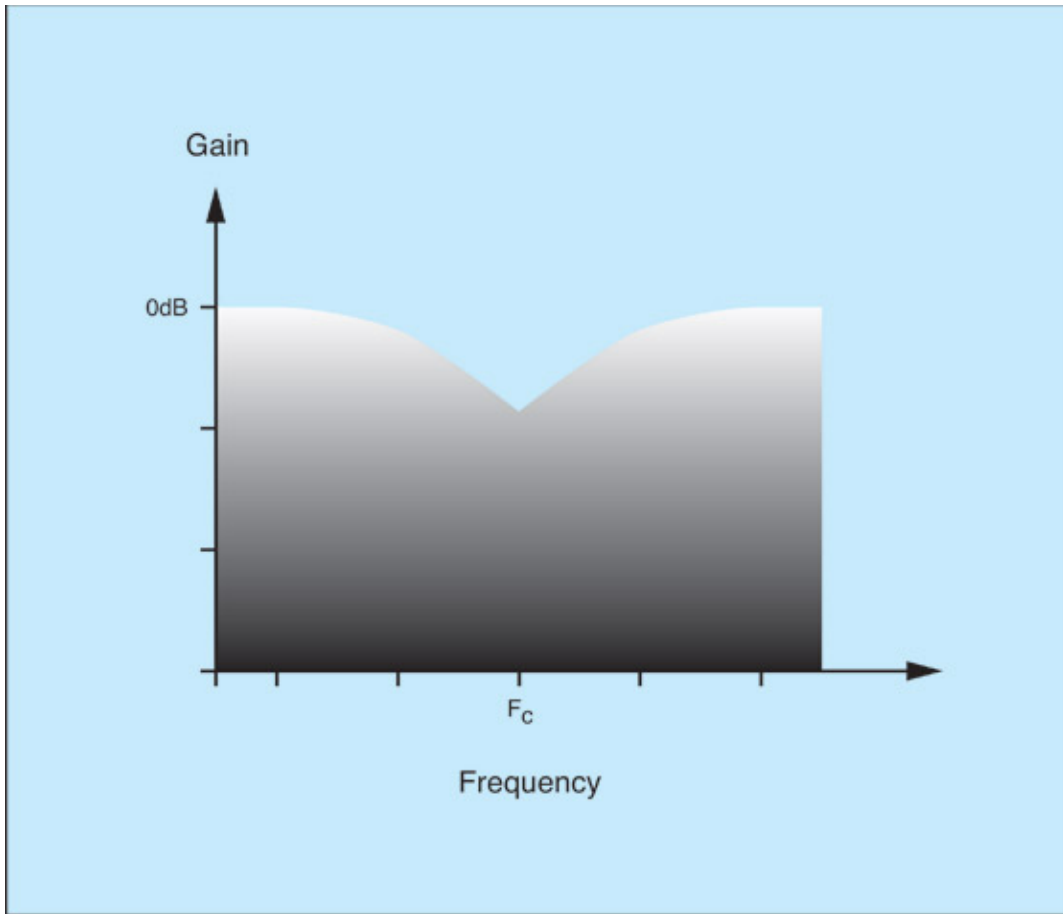
Figures 3: band reject (notch) filter
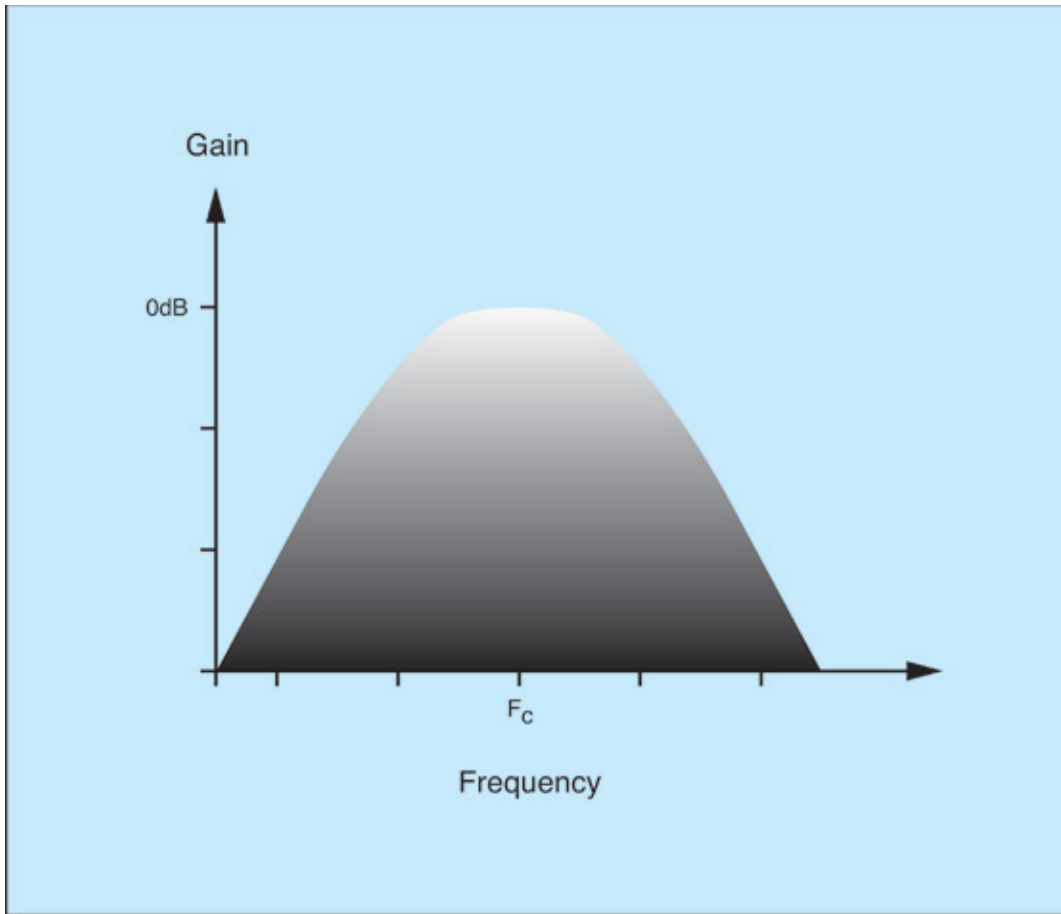
Figures 4: band pass filter

A simple 6dB/oct band-pass filter is a fairly weak shaper of a signal, but if you place a number of these with the same centre frequency in series, the width of the pass-band becomes narrower and narrower until only a limited range of frequencies is allowed through. (Figures 5 and 6.)
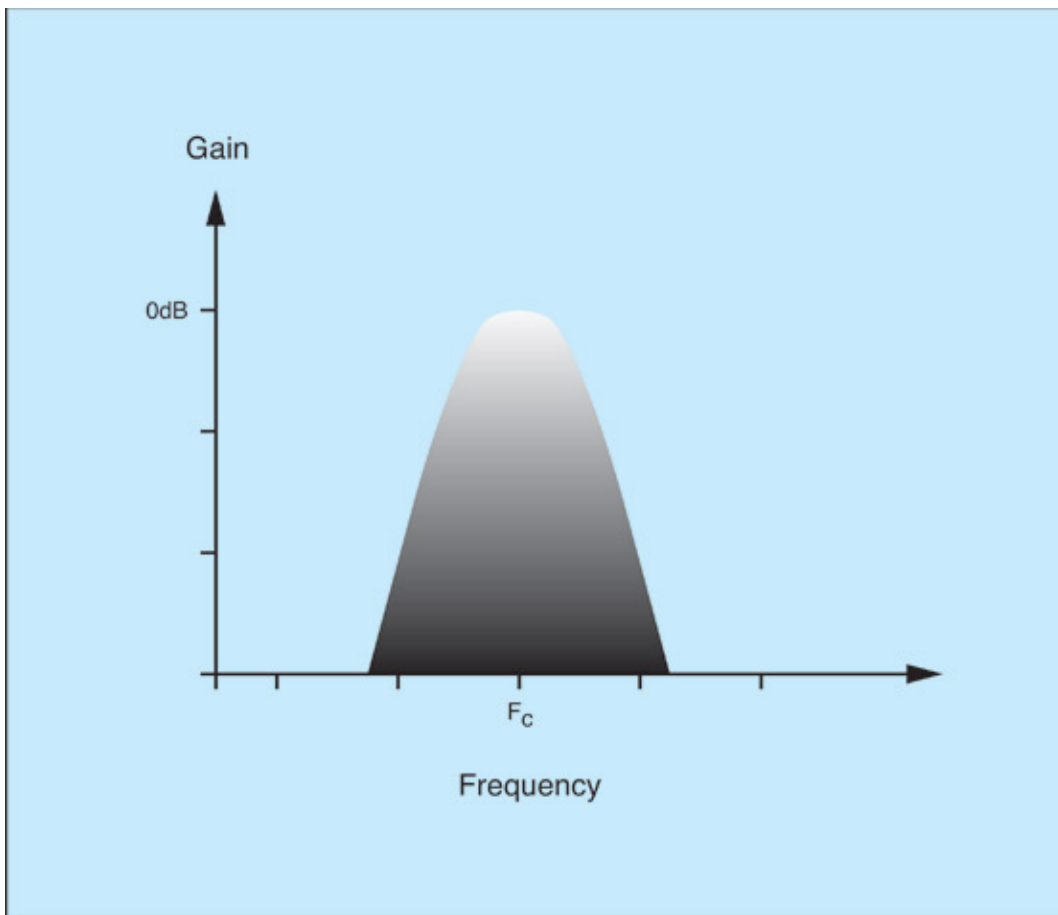
Figures 5: The responses of placing two band-pass filters in series
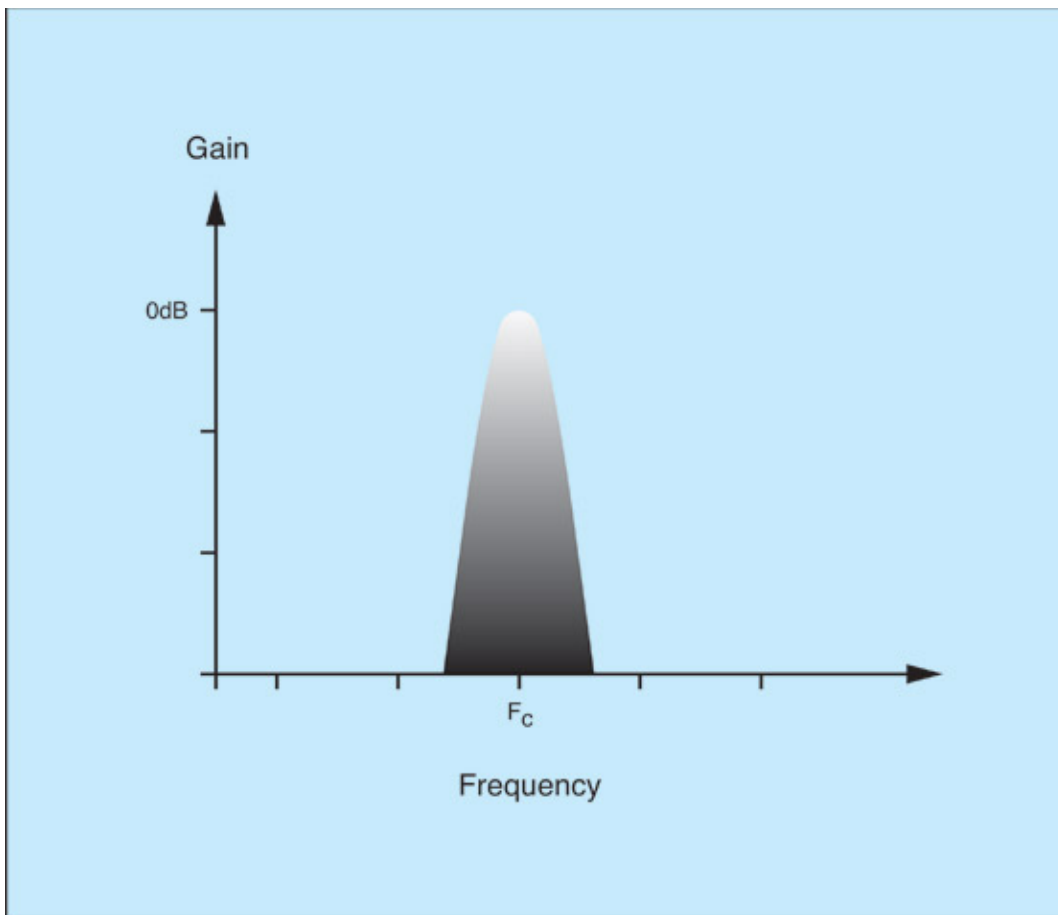
Figures 6: The responses of placing 4 band-pass filters in series

Now imagine the case in which you place, say, three of these multiple band-pass filters in parallel. If you set the cut-off frequency to be different for each signal path, you obtain three distinct peaks in the spectrum (see figure 7) and the filters attenuate any signal lying outside these bands. As you can imagine, any sound filtered in this way adopts a distinctive new character.

(A similar result can be obtained using parallel peaking filters or even low-pass and high-pass filters with high resonance values, and a number of venerable keyboards in the 1970s used architectures based on these. Although not strictly equivalent, the results look similar and for many synthesis purposes are interchangeable.)
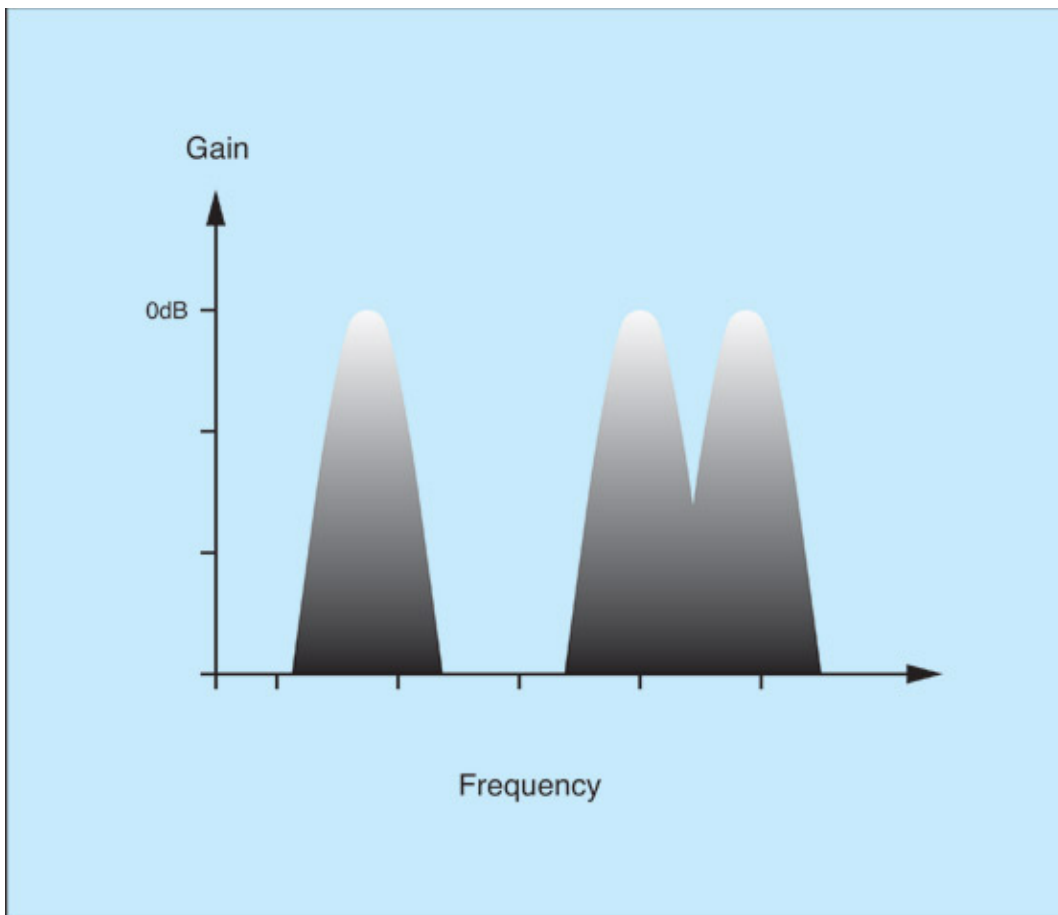
Figure 7: Multiple BPFs in parallel

If we wanted to pursue this path further, it would take us into a whole new domain of synthesis called physical modeling. This is because the characteristic resonances of acoustic instruments - the bumps in the instruments' spectral shapes - are recognisable from one instrument to the next. For example, all violas are of similar shape, size, and construction, so they possess similar resonances and exhibit a consistent tonality that allows your ears to distinguish them from say, classical guitars generating the same pitch. It therefore follows that imitating these resonances is a big step forward toward realistic synthesis. Today, however, we're going to restrict ourselves to the special case of this that is sometimes called 'vocal synthesis'.

The Human Voice

Because you share the architecture of your sound production system with billions of other people, it's safe to say that all human vocalizations - whatever the language, accent, age or gender - share certain acoustic properties. To be specific, we all push air over our vocal cords to generate pitched signals, and we can tighten and relax these cords to change the pitch that we produce. Furthermore, we all produce broadband noise.

The pitched sounds are generated deep in our larynx, so they must pass through our throats, mouths, and noses before they reach the outside world through our lips and nostrils. And, like any other cavity, this 'vocal tract' exhibits resonant modes that emphasise some frequencies while suppressing others. In other words, the human vocal system comprises a pitch-controlled oscillator, a noise generator, and a

set of band-pass filters! The resonances of the vocal tract and the spectral peaks that they produce are the formants that I keep referring to, and they make it possible for us to differentiate different vowel sounds from one another. (Consonants are, to a large degree, noise bursts shaped by the tongue and lips and, in general, you synthesise these using amplitude contours rather than spectral shapes.)

Table 1 shows the first three formant frequencies for some common English vowels spoken by a typical male adult. As you can see, they do not follow any recognisable harmonic series, and are distributed in seemingly random fashion throughout the spectrum.

Vowel Sound	F1	F2	F3
"ee"	207	2300	3000
"oo"	300	870	2250
"i"	400	2000	2550
"e"	530	1850	2500
"u"	640	1200	2400
"a"	660	1700	2400

Table 1: Some examples of the first three formants in English vowel sounds

Given table 1 and a set of precise filters you might think that you should be able to create passable imitations of these sounds but, inevitably, things are not quite that simple. It's not just the centre frequencies of the formants that affect the quality of the sound, but also the narrowness of their pass-bands (their Qs) and their gains. So we can extend the information in table 1 to create formants that are more accurate. Let's take "ee" as an example... (See table 2).

	Vowel Sound "ee"	Gain (dB)	Q
F1	207	0	5
F2	2300	-15	20
F3	3000	-9	50

Table 2: Adding amplitude and width to the formant values

This is an improvement, but it isn't the end of the story, because the sound generated by a set of static band-pass filters is, umm... static, whereas human vowel sounds are not. To undertake true speech synthesis, we need to make the band-pass filters controllable, applying controllers to all of their centre frequencies, Qs and gains. Unfortunately, this is beyond the scope of this tutorial; so let's now turn our attention to creating vocal-like sounds using the Formant Filters in Thor.

Creating a choral patch

Thor's Formant Filter imposes four peaks upon any wide-band signal fed through it, and we can see these if we apply white noise to its input and view its output using a spectrum analyser. You can move the peaks by adjusting the X and Y positions and the Gender knob, but the interactions between these are too complex to describe here. So instead, I created the simple patch in figure 8, and used this to capture four images and four audio samples, one of each taken at each corner of the X/Y display, all with the Gender knob at its mid position. You see the results in figures 9 to 12, and hear them in sounds #1 to #4.

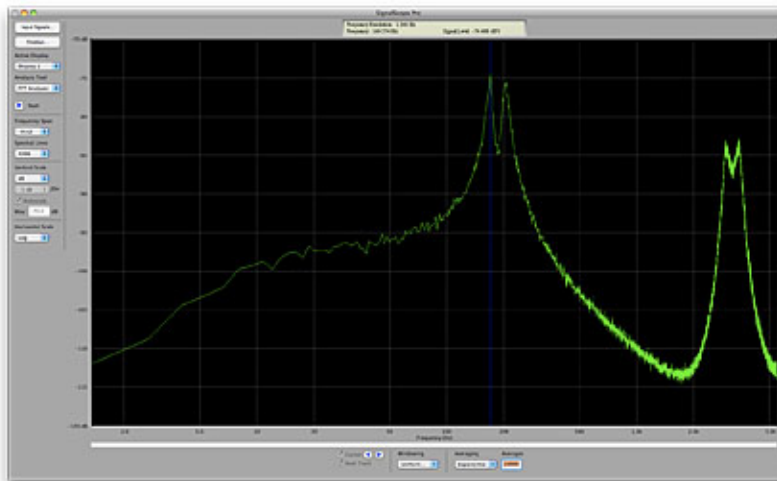*Figure 8: Measuring the output spectrum of the Formant Filter*

Figure 9: Output spectrum of Formant Filter: input = white noise, X=min, Y=min (Sound #1 ▶)
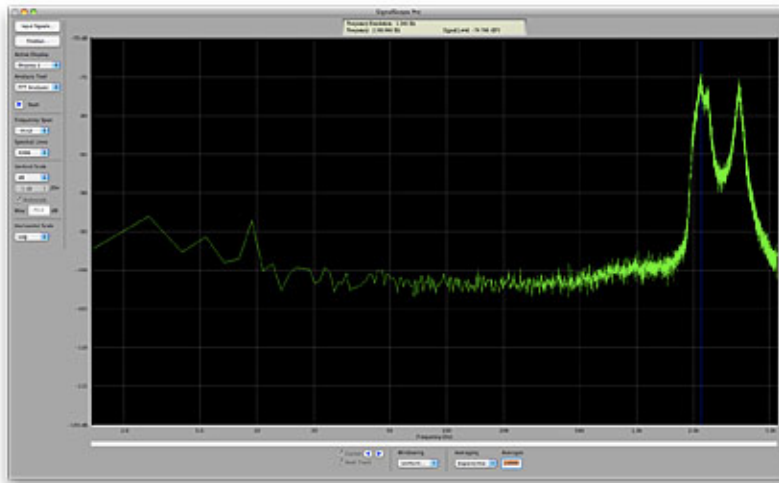
Figure 10: Output spectrum of Formant Filter: input = white noise, X=max, Y=min (Sound #2 ▶)
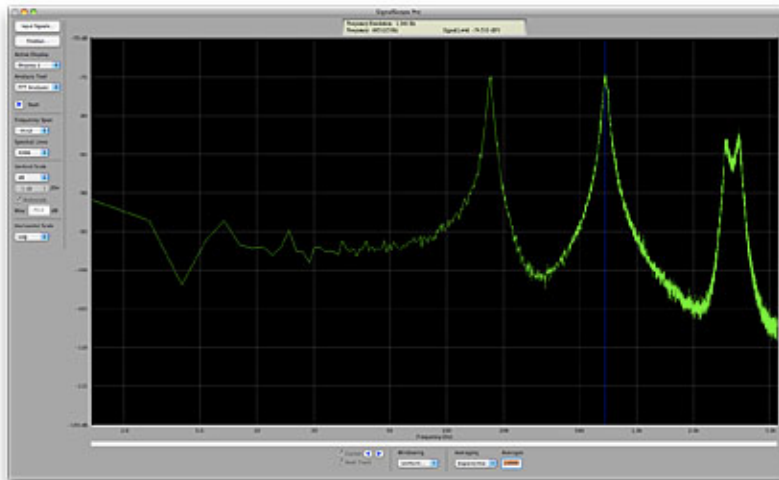

Figure 11: Output spectrum of Formant Filter: input = white noise, X=min, Y=max (Sound #3 ▶)

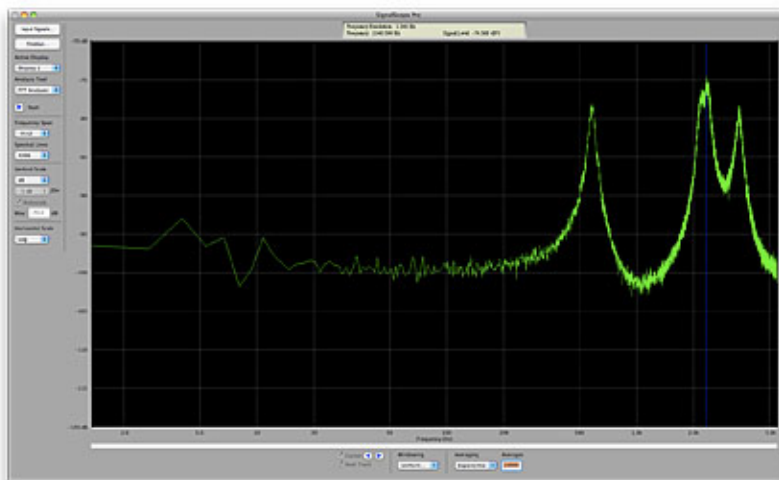*Figure 12: Output spectrum of Formant Filter: input = white noise, X=max, Y=max (Sound #4 ▶)*

These responses don't imitate the formants of a human voice in a scientifically accurate way but they are nonetheless quite capable of conveying the impression of a human voice if we replace the noise at the input with a signal that is closer to that generated by the vocal cords. I have chosen a pulse wave with a value of 23 (a duty cycle of about 18%) and shaped the output with a gentle ASR amplitude contour. With no effects applied, the patch looks like figure 13, and you can hear it in sound #5 ▶ .

Figure 13: Generating a simple pulse wave
(Click to enlarge)

Now let's apply the Formant Filter. I've inserted this into the Filter 1 slot, set the Gender to a value of 46 and set the X/Y values to 46 and 38. (See figure 14.) There is nothing magical about these numbers; I just happen to like the results that they give, especially when I add additional elements into the patch. You'll also see that the key tracking is set to its maximum, which means that the spectral peaks move within the spectrum as the pitch changes. This is not strictly accurate but I find that, for this patch, the high notes are too dull if I leave the tracking at zero.

Figure 14: Formant filtering the pulse wave

(Click to enlarge)

The patch now exhibits a 'vocal' timbre, but it's rather too static for my taste, so before recording a sample I've enhanced it a little by adding some movement to the positions of the filter peaks. I did this by applying a small amount of smoothed random modulation to the X position using LFO1 and by applying a small amount of smoothed random modulation to the Y position using LFO2. The resulting sound (shown in figure 15 and heard in sound #6) now has a touch of subtle instability that makes it a little more human than before. Nonetheless, it sounds like nothing so much as a late-70s vocal synth with the ensemble button switched off. Ah... there's the clue. Leaving figure 15 untouched and invoking some external ensemble, EQ and reverb results in sound #7 . Luscious!!

Figure 15: A vocal patch

(Click to enlarge)

Because the human voice comprises noise as well as tonal signal, we can enhance this still further by adding white noise at low amplitude to the input signal. Figure 16 shows this and, though the difference is again subtle, it can be a worthwhile improvement.

Figure 16: Adding noise to the vocal patch

(Click to enlarge)

Of course, you might say that the addition of the external effects made the last sound what it is, and to some extent that would be true, but let's check what the latest patch sounds like without the Formant Filter (Sound #8 ▶). As you can hear, it has the nuance of a vocal timbre, but at best you might call it a 'StringVox' patch. Clearly, it's the interaction of the filtered sound and the ensemble that achieves the desired effect, which is something that Roland demonstrated more than thirty years ago when they released the wonderful VP-330 Vocoder Plus, whose unaffected vocal sound was little more than a nasal "aah" but whose ensemble defined the choral sounds of the late-70s and early 80s.

Now let's ask what might happen if we replace the pulse wave that forms the basis of the previous sounds with something that is already inherently voice-like. We can investigate this by replacing the Analogue Osc with a Wavetable Osc, selecting the Voice table and choosing a suitable position with it. Figure 17 and sound #9 ▶ demonstrate this and, as you can hear, a different - but still very useable - vocal timbre results.

Figure 17: A wavetable-based formant-filtered vocal sound
(Click to enlarge)

You might think that you always have to start with a quasi-vocal waveform to obtain a vocal sound, but this is far from true. Take the swarm of giant, angry insects in sound #10 ▶ , which was created using the patch in figure 18. This is the unfiltered output from a Multi Osc with random detune being swept by the Mod Env from a large to a low value at the start of the note, the pitch being swept upward at the same time, and a delayed vibrato being supplied by LFO1. If we now add a Formant Filter to this patch (figure 19) the nature of the sound changes dramatically, becoming vocal in timbre and sounding almost like a male ensemble in a reverberant space, even though no effects have been applied (Sound #11 ▶).

Figure 18: Turning a swarm of angry insects into a male voice choir

(Click to enlarge)

Figure 19: Turning a swarm of angry insects into a male voice choir (cont.)

(Click to enlarge)

Other sounds

There are of course many other things we can do with vocal synthesis. Returning to the wavetable oscillator, I have created a new patch with the Gender set to maximum and the X/Y position in the centre of the display. (Figure 20.) I have added four paths in the modulation matrix to refine this, with a touch of vibrato supplied by LFO1, some random pitch deviation supplied by LFO2, a short sweep though part of the wavetable generated by the Mod Env, and an organ-like amplitude envelope generated by the Global Env that curtails every note eight seconds after you play it. ("Ah-ha!" I hear you say.) You might think that the use of a vocal wavetable and a Formant Filter with the Gender set to maximum would produce a female vocal timbre, but instead it emulates the strange tonal quality of a Mellotron. This is because the Mellotron's tape replay system exhibits strong peaks in its output spectrum, so the use of a formant filter is a good way to imitate this. Sound #12 ▶ demonstrates the patch in figure 20 played without external effects, while sound #13 ▶ demonstrates what is possible when ensemble is applied.

Figure 20: Melly strings
(Click to enlarge)

Finally, we come to the famous 'talking' synthesiser patch. There are many variants of this, mostly based around the sounds "ya-ya-ya-ya-ya" or "wow-ow-ow-ow-ow", but they all boil down to moving the formant peaks while the sound is playing. If we had a complex, scientifically accurate synthesiser, we could reproduce genuine vowel sounds, but few if any commercially available synths are capable of this. Figure 21 shows a Thor patch that says "wow" by shifting the Gender, X and Y values by appropriate amounts while opening and closing the audio amplifier. With no external effects applied, we obtain sound #14 ▶ from this. Wow!

Figure 21: Wow!

(Click to enlarge)

Epilogue

To be honest, concentrating on vocal and SynthVox sounds only scratches the surface of formant synthesis, and you can use formant filters to create myriad other sounds ranging from orchestral instruments to wild, off-the-wall effects. But, unfortunately, there's no space to demonstrate them here because we've come to the end of my tutorials introducing Thor's filters. I hope that they have given you some new ideas and - as I suggested when I concluded my tutorials on Thor's oscillators - have illustrated why there is so much more to synthesis than tweaking the cut-off knobs of resonant low-pass filters. Thank you for reading; I appreciate it.

Text by Gordon Reid

Creative Sampling Tricks

 Yours truly is old enough to have been around when digital samplers first arrived. Admittedly I never touched a Fairlight or an Emulator back when they were fresh from the factory – those products were way out of a teenager's league – but I distinctly remember the first time I laid hands on an S612, Akai's first sampler. Its modest 128 kB RAM could hold a single 1-second sample at maximum quality (32 kHz) – but none the less it was pure magic to be able to record something with a mic and instantly trigger it from the keyboard. I spotted that particular Akai sampler hooked up in my local music store, and tried it out by sampling myself strumming a chord on a Spanish guitar. My first sample...!

As the years went by, I gradually became spoiled like everyone else; there were tons of high quality sample libraries available on floppies, and soon enough the market was swamped with dedicated sample playback instruments such as the S1000PB, the E-mu Proteus, the Korg M1 and the Roland U-series to name but a few. This trend carried over into software instruments; manufacturers and others kept sampling like crazy for us so it seemed more or less superfluous to do it yourself. Propellerhead was no exception – with no sampling facilities and no hard disk recording, Reason remained a playback device for canned samples for almost 10 years – but in Reason 5 and Record 1.5, they got around to adding a sampling feature. In typical Propellerhead fashion, don't do it unless it's done right. The trick to doing it right was to bring back the simplicity and instant gratification of those early samplers – just plug in a source, hit the sample button, perform a quick truncate-and-normalize in the editor, and start jamming away.

Setting Up for Sampling

The sampling feature is part of the Audio I/O panel on the hardware interface. On the front panel there's an input meter, a monitor level control, a button for toggling the monitoring on/off and an Auto toggle switch – activate this and monitoring will be turned on automatically while you're recording a sample (figure 1).

Figure 1: Reason's Hardware interface now sports a sampling input

On the back, there's a stereo input for sampling and this will be hooked up to audio inputs 1 and 2 by default. In case you're sampling a mono source such as a mic, and the source is audio input 1, make sure to remove the cable between audio input 2 and sam-

pling input R - otherwise you'll get a stereo sample with only the left channel (figure 2).

Figure 2: The default routing is Audio Input 1/2 to Sampling Input L/R

Now, if you're sampling with a mic you may want some help with keeping the signal strong and nice without clipping, and perhaps a touch of EQ as well. In that case you can pass the mic signal through Reason devices such as the MClass suite and then sample the processed signal (figure 3).

Figure 3: You can process your audio input prior to sampling, for example with EQ

Sampling devices or the entire rack

You might wonder "hey, what's the point of sampling Reason itself when I can just export an audio loop?" Well, there are situations where that method just doesn't cut it. A few examples:

- When you want to sample yourself playing an instrument, singing, rapping or scratching, you're running the audio through Reason's effect units and you want to capture with the sample.

- When you want to edit something quickly without leaving Reason, for instance if you want to reverse a segment of your song or reverse single sounds such as a re-verb 'tail'. Just wire it into the sampling input, hit the sample button on the Kong/NN-XT/NN19/Redrum, pull up the editor, reverse the sample, done.
- When you want to sample the perfect 'snapshot' of instruments doing something that's non-repeatable, for example when a sample-and-hold LFO happens to do something you wish it would do throughout the entire song.
- When you're doing live tweaking that can't be automated, such as certain Kong and NN-XT parameters.

It's a straightforward process where you simply take any Reason devices you want to sample and plug them into the Sampling Input. In this illustration, Reason has been set up to sample from a Dr. OctoRex (figure 4).

Figure 4: Routing Reason devices to the sampling input is this simple

When you sample straight from the Reason rack, you'll want to make sure to enable the monitoring, because unlike situations where you sample an external acoustic source, you'll hear absolutely nothing of what you're doing unless monitoring is enabled (figure 5).

Figure 5: Turn on monitoring so you can hear what you're doing

Control Voltages and Gates

DISCOVERING REASON — This tutorial discusses the concepts of control voltages (CVs) and Gates in Propellerhead's software. Of course, they're not really voltages because everything is happening within the software running on your PC or Mac. But the concepts are the same so, before going on to discuss how to use them, let's first take the time to understand where CVs and Gates came from, what they are, and what they do.

We'll begin by considering one of the most fundamental concepts in synthesis: there is no sound that you can define purely in terms of its timbre. Even if it seems to exhibit a consistent tone and volume, there must have been a moment when it began and a moment when it will end. This means that its loudness is contoured in some fashion. Likewise, it's probable that its tone is also evolving in some way. So let's start by considering an unvarying tone generated by an oscillator and make its output audible by playing it through a signal modifier — in this case, an amplifier – and then onwards to a speaker of some sort. We can represent this setup as figure 1.
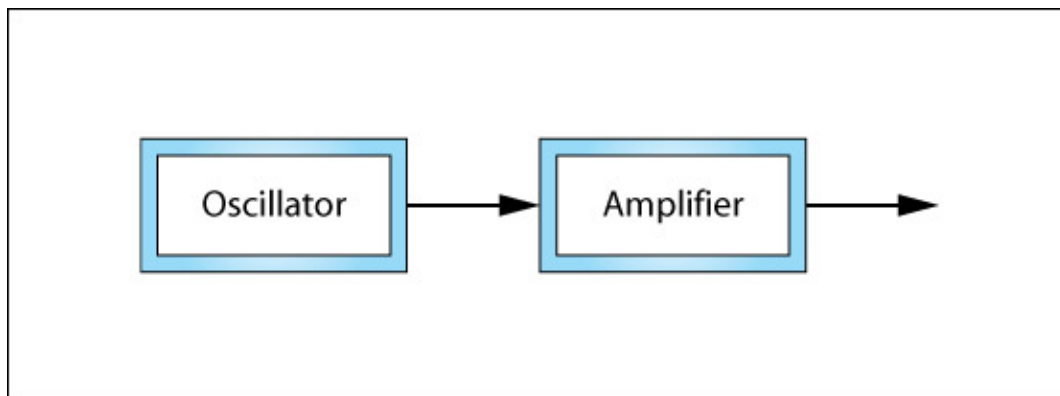

Figure 1: A simple sound generator

Now imagine that the amplifier in figure 1 is your hi-fi amp, and that the volume knob is turned fully anticlockwise. Clearly, you will hear nothing. Next, imagine taking hold of the knob and rotating it clockwise and then fully anticlockwise again over the course of a few seconds. Obviously, you will hear the sound of the oscillator evolve from silence through a period of loudness and then back to silence again. In other words, your hand has acted as a controller, altering the action of the modifier and therefore changing when you hear even though the audio generated by the source has itself remained unchanged.

Twisting one or more knobs every time that you want to hear a note isn't a sensible way to go about things, so early synthesiser pioneers attempted to devise a method that would allow them to control their sources and modifiers electronically. They found that they could design circuits that responded in desirable ways if voltages were applied at certain points called control inputs. So, for example, an amplifier could be designed in such a way that that, when the voltage presented to its control input was 0V its gain was -∞dB (which you would normally call 'zero', or 'off') and when the voltage was, say, +10V, the amplifier provided its maximum gain. Thus the concepts of control voltages (CVs) and voltage-controlled amplifiers (VCAs) were born. (Figure 2.)
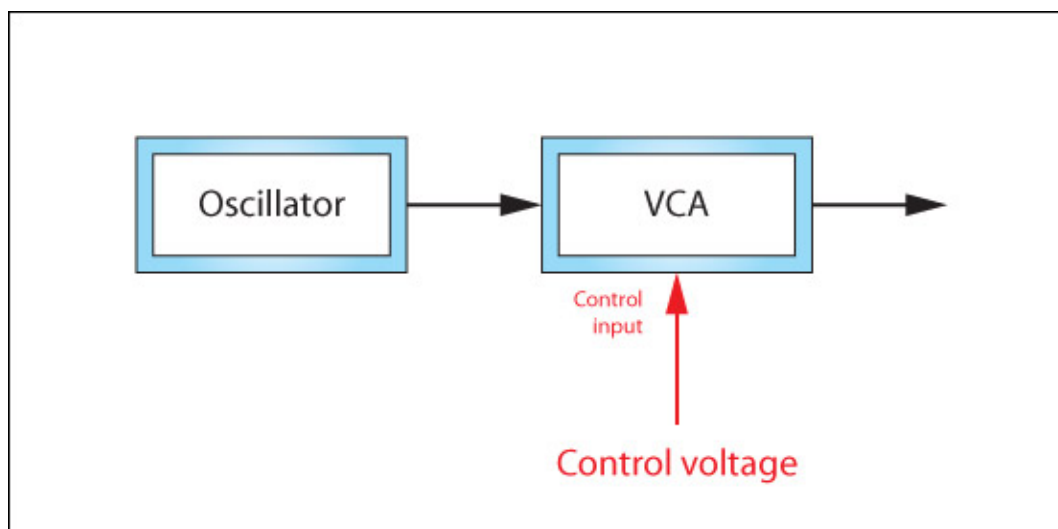

Figure 2: Shaping the output from the amplifier

The next thing that was needed was a mechanism to determine the magnitude of the control voltage so that notes could be shaped in a consistent and reproducible fashion. Developed in a number of different forms, this device was the contour generator, although many manufacturers have since called it (less accurately) an envelope generator, or "EG". The most famous of these is called the ADSR; an acronym that stands for Attack/Decay/Sustain/Release. These names represent the four stages of the contour. Three of them - Attack, Decay, and Release - are measures of time, while the Sustain is a voltage level that is maintained for a period determined by... well, we'll come to that in a moment. (Figure 3.)

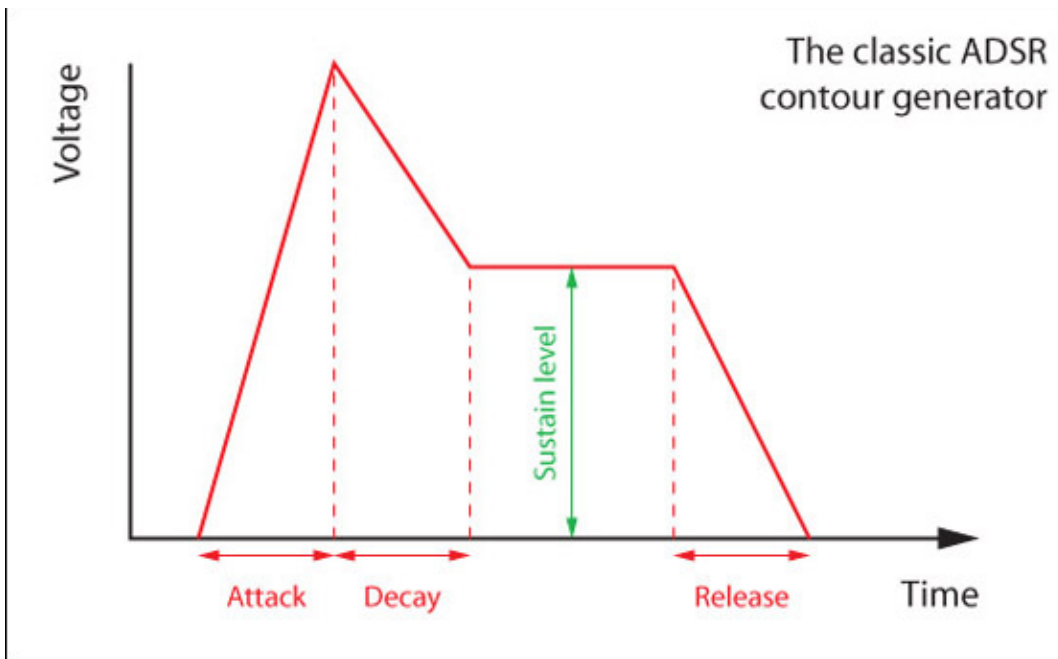The classic ADSR contour generator

Figure 3: The ADSR contour generator

If we now connect this device to the VCA in figure 2 it should be obvious that the contour shapes the loudness of the note as time passes. (Figure 4.) But how do we trigger it?
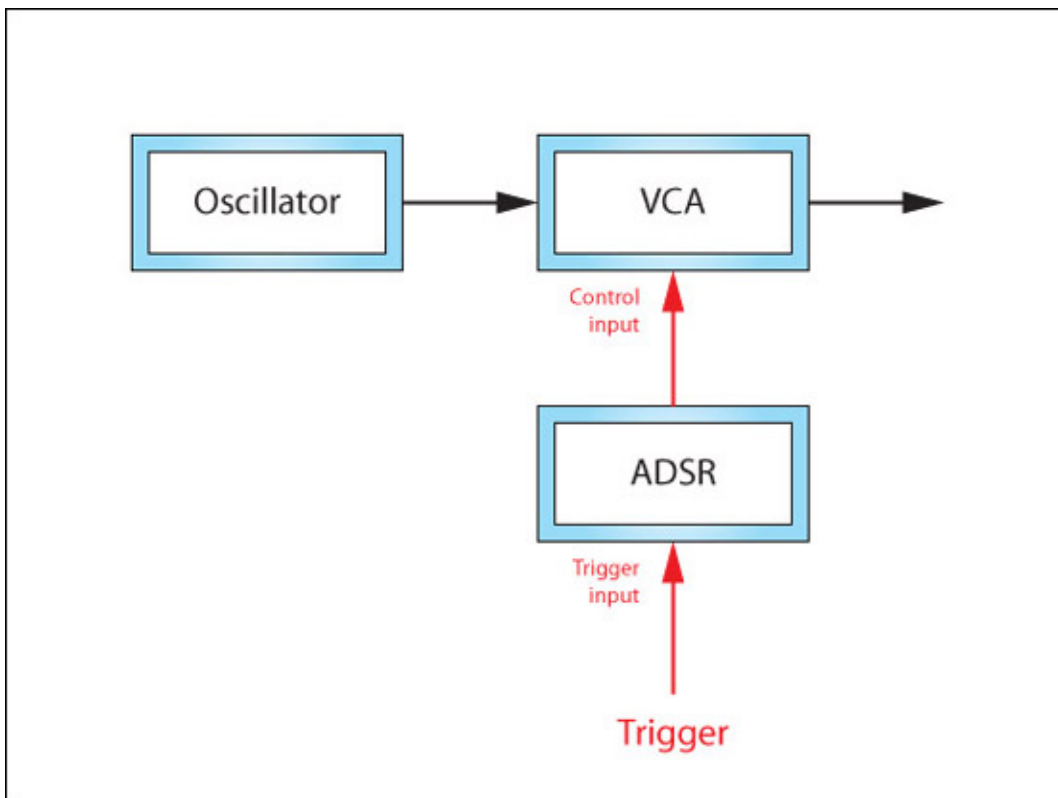
Figure 4: Amplitude control using a triggered contour generator

Let's consider what happens at the moment that you press a key on a typical analogue monosynth. Many such synths generate three control voltages every time that you do so. The first determines the pitch of the sound produced, so we can replace the concept of the oscillator in figure 1 with that of the Voltage Controlled Oscillator (VCO). The second is called a Trigger. This is a short pulse that initiates the actions of devices such as contour generators. The third is a called a Gate. Like the Trigger, the Gate's leading edge can tell other circuits that you have pressed a key but, unlike the Trigger, its voltage is generated for the whole time that you keep the key depressed, which means that it can also tell those other circuits when you release the key. (Figure 5.)
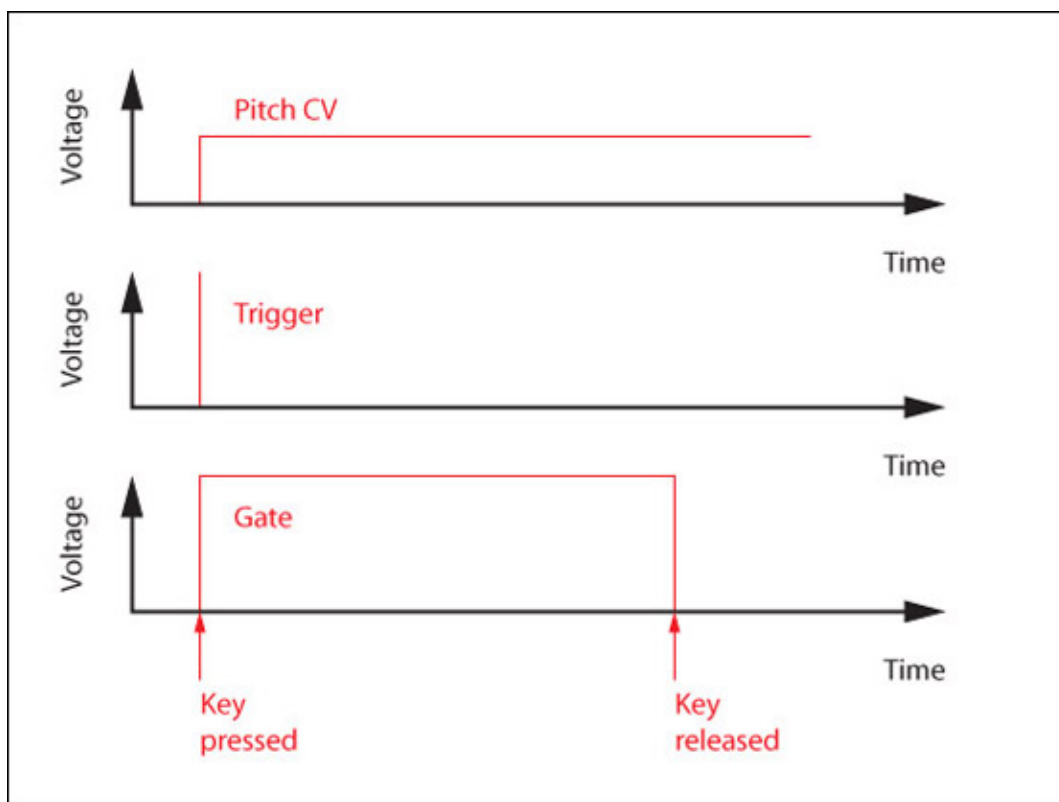

Figure 5: Pitch CV, Trigger & Gate signals

If we now return to the contour generator it's clear that the Gate is important because it tells the ADSR how long to hold the Sustain level before entering the Release phase. This means that I can redraw figure 6 to show the ADSR being triggered and how the note's shape is influenced by the duration of the Gate. (Figure 6.)
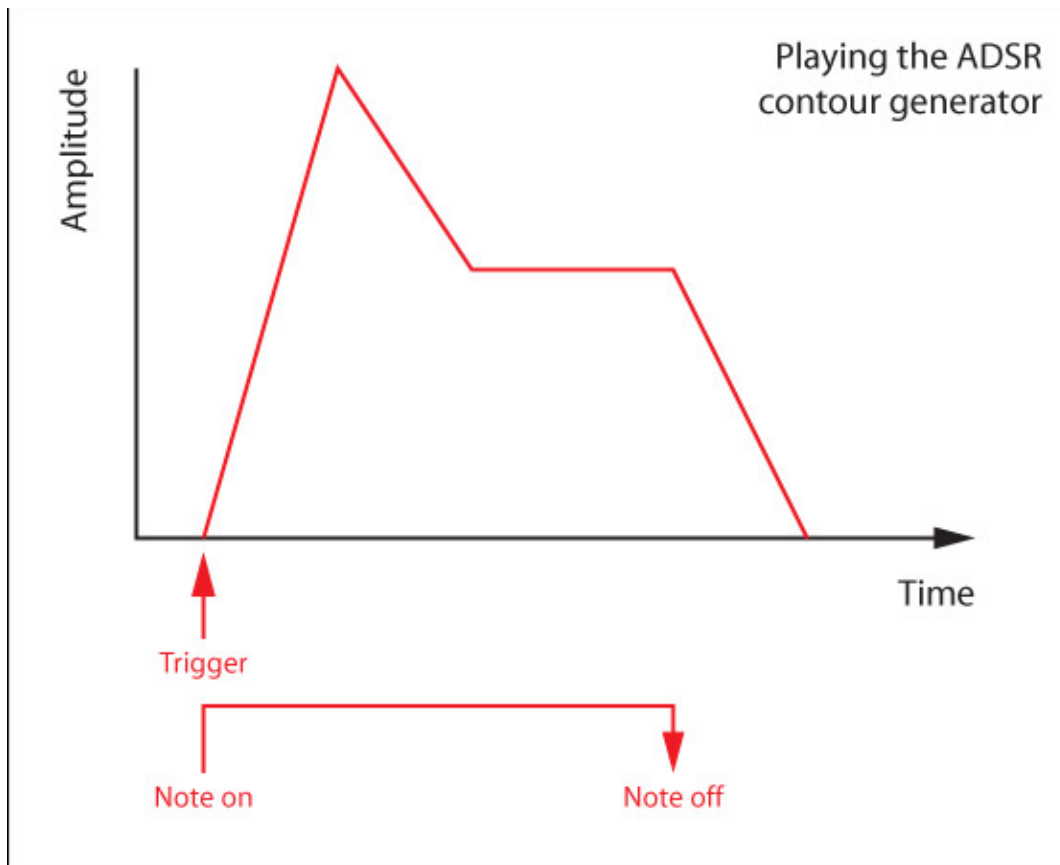
Figure 6: Controlling the ADSR

It should now be obvious why we need timing signals, but why do we need two of them? There are many synthesisers that work with just a pitch CV and a Gate, but consider what happens when you hold one key down continuously (resulting in a continuous Gate) and press other keys, perhaps to create trills or some other musical effects. If there are no subsequent Triggers, the Gate holds the ADSR at the sustain level until you release all the keys so, after the first note, none of the subsequent ones are shaped correctly. This is called "single triggering". However, if a Trigger is generated every time that a note is pressed the contour generator is re-initiated whether or not a previous note is held, and subsequent notes are shaped as intended. ("Multi-triggering".)

Putting it all together

At this point, we have a VCO controlled by a pitch CV, plus an ADSR contour generator whose initiation and duration are controlled by a Trigger and a Gate. In principle, this is all that we need to programme and play a wide range of musically meaningful sounds, but let's now extend the model by adding a second ADSR and a voltage controlled low-

pass filter (VC-LPF) to affect the brightness of the sound. Shown in figure 7 (in which the red lines are control signals of one form or another, and the black lines are audio signals) this is all we need for a basic synthesiser.
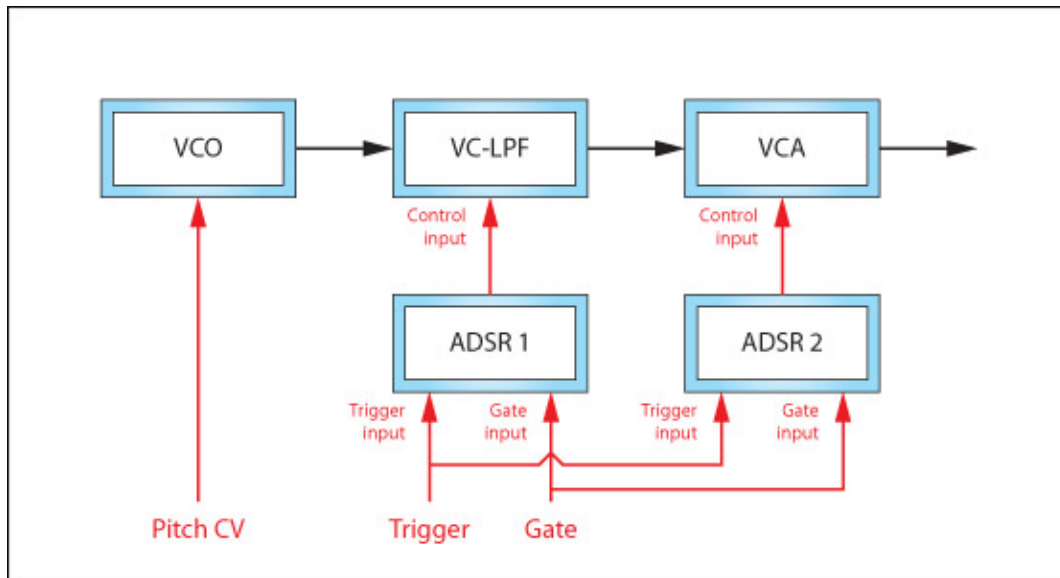
Figure 7: A basic voltage-controlled synthesiser

Of course, real synths embody a huge number of embellishments to these concepts, but we can ignore these. What's important here is to understand the differences between the three control signals and to be able to recognise one from another. However, this is not always as straightforward as it might seem. Imagine that you replace ADSR 2 in figure 7 with the triangle wave output from a low frequency oscillator. Now, instead of having an articulated note, you have one that exhibits tremolo. In other words, the oscillator's triangle wave is acting as a CV. But what happens if we simultaneously replace the Trigger in figure 7 with the pulse wave output from the same LFO? The LFO is now generating a stream of triggers that regularly reinitialise ADSR 1 to shape the brightness of the sound in a repeating fashion. As figure 8 shows, it's not WHAT generates a signal that determines whether it's a CV, Trigger or Gate, it's WHERE you apply that signal that counts.
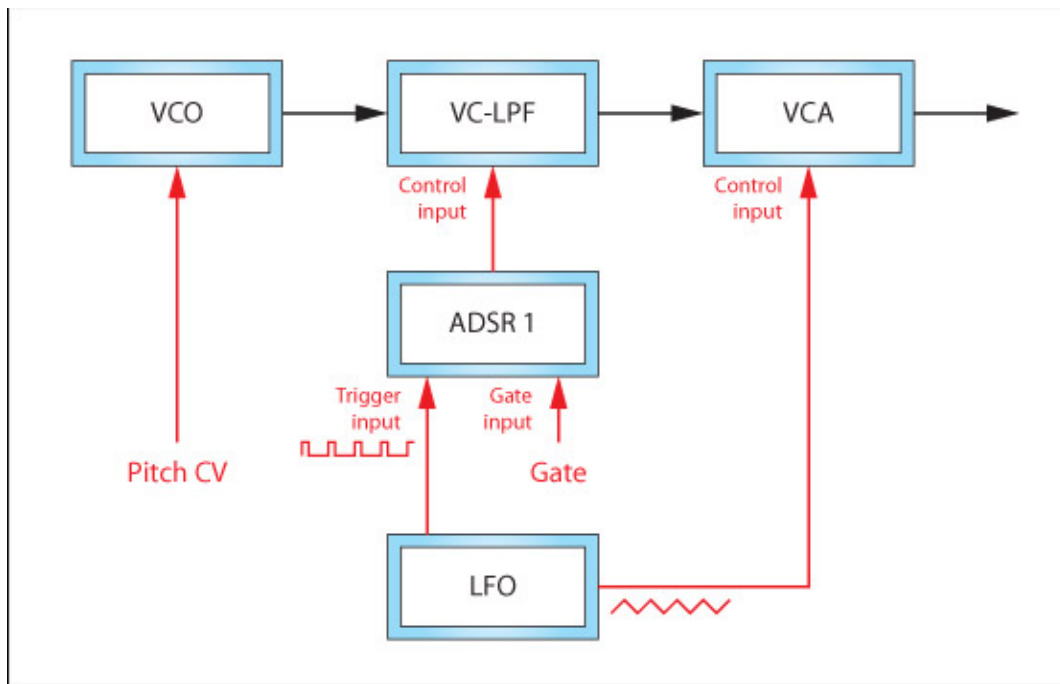
Figure 8: An LFO acting simultaneously as a CV generator and a Trigger generator

Of course, oscillators, filters and amplifiers need not be affected only by pitch CVs, contour generators and LFOs. There are myriad other voltages that can be presented to their control inputs, including velocity, aftertouch and joystick CVs, the outputs from S&H generators and voltage processors, envelope followers, pitch-to-CV converters, and many more devices. Discussing these could form the basis of an entire series of tutorials, but at this point we have to take a different direction...

So far, we've only discussed how things affect other things within a single synthesiser, but CVs and Gates also make it possible for separate instruments to talk to one another. Take the simple example of an analogue sequencer driving a monosynth. In this case, the voltages generated by the synth's keyboard are replaced by those generated within the sequencer, which sends a stream of Gates to tell the synth when to play and how to articulate the notes, and a stream of CVs to tell it what pitches those notes should be. If you're lucky, the sequencer may even be capable of producing a second CV (usually called an Auxiliary CV) that can affect other aspects of the sound while the sequence is playing. (Figure 9.)
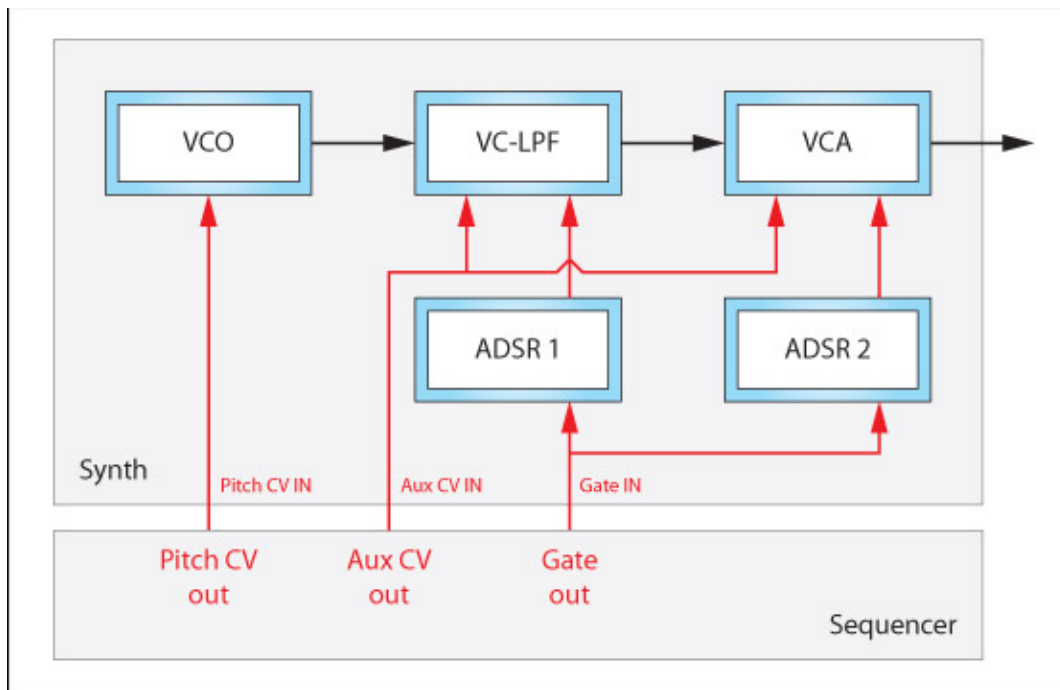
Figure 9: Connecting a synth and a sequencer

But that's not all because, with suitably equipped instruments capable of generating and receiving various control voltages, the CVs, Triggers and Gates generated by one synthesiser can control what happens on a second (or a third, or a fourth…). Furthermore, CVs generated by any of them can be mixed together (or not) and directed to control the actions of things such as effects units, sequencer controls, panning mixers and much more. (See figure 10.) It's simple, and can be wonderfully creative. The only problem is that, with a fistful of patch cables and a bit of imagination, you'll soon be on your way to creating a multi-coloured mess of electrical spaghetti.
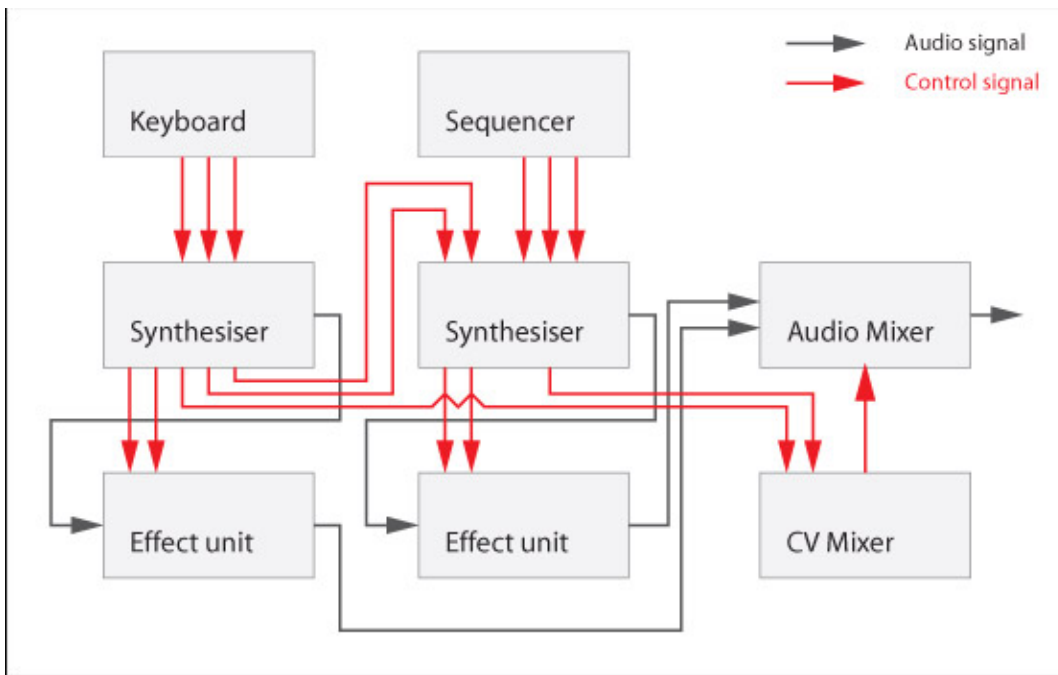
Figure 10: Connecting multiple devices using CVs and Gates

Recording Electric Guitar

By Matt Piper

Welcome to the first article in the Record U series – this article will teach basic techniques for recording electric guitar, with information about mic'ing guitar amps, and also recording directly into Reason with no microphones or amplifiers at all, using Reason's built-in Line 6 Guitar Amp device. Lets get started with some general tips for recording your guitar amp!

Use a flashlight

In many combo amplifiers, the speaker is not actually in the center of the cabinet, and may not be easily visible through the grill cloth. In this case, shining a flashlight through the grill cloth should allow you to easily see the position of the speaker so you can place your microphone accurately.

Recording in the same room as your amplifier

Recording close to your amplifier (especially with your guitar facing the amplifier) can have the benefits (especially at high gain settings) of increasing sustain and even achieving pleasant, controlled harmonic feedback. This is due to a resonant feedback loop between the amplifier, the speaker, your guitar pickups, and your guitar strings. You may also be able to achieve this effect when recording directly into the computer, if you are playing in the control room with the studio monitors (not headphones!) turned up loud enough. While it may sometimes be desirable to play guitar while separated from the amp (perhaps because the amp is in a bathroom, other separate room, or isolation box to keep it from bleeding into other instrument microphones when recording a live band), you will lose the opportunity for this particular feedback effect.

My guitar tone sounds so much brighter on the recording than it does when I listen to my amp!

Quite often (especially with combo amplifiers, where the amp and speaker are both in the same cabinet), guitarists get used to the sound of standing several feet above their

speaker, while the speaker faces straight out parallel to the floor. Because of this, much of the high frequency content coming from the speaker never reaches your ears. If you start making it a habit to tilt the amp back, or put the amp up on a stand or tilted back on a chair so that the speaker is pointed more toward your head than your ankles, you will begin becoming accustomed to the true tone of your amp, which is the tone the microphone will record, and the tone that members of a live audience will hear. This may initially require you to make adjustments to your tone—but once you have achieved a tone you like with this new setup, you can be confident that the sound you dialed in will be picked up by a properly placed microphone.

Proximity effect

When a microphone with a cardioid pattern (explained later in the Large Diaphragm Condenser Microphones section of this article) is placed in very close proximity to the sound source being recorded, bass frequencies become artificially amplified. You may have heard a comedian cup his hand around the microphone with the mic almost inside his mouth to make his voice sound very deep when simulating "the voice of God" (or something like that). This is an example of the proximity effect. This effect can result in a very nice bass response when placing a microphone close to the speaker of your guitar cabinet or combo amplifier. In fact, the Shure SM57 is designed to make use of this effect as part of its inherent tonal characteristics.

On-axis vs. off-axis

You will see these terms mentioned later in my descriptions of the mic setup examples. On-axis basically means that the microphone element is pointed directly at the sound source (sound waves strike the microphone capsule at 0 degrees). Off-axis means (when mic'ing a speaker) that the microphone element is aimed at an angle rather than straight at the speaker (so the sound waves strike the microphone capsule at an angle). On-axis will give you the strongest signal, the best rejection of other sounds in the room, and a slightly brighter sound than off-axis. I usually mic my amp on-axis, with the mic (my trusty Shure SM57) somewhere near the edge of the speaker. However, this is not "the one right way." The right way is the way that sounds best to you. I hope that the following examples will help you find your own path to the tone you are looking for.

Recording Guitar through an Amplifier with Different Mic Setups (Examples)

The following recordings were made with a Gibson Les Paul (neck pickup) played through a Fender Blues Junior combo tube amp sitting on a carpeted floor (propped back a bit). A looping pedal was used so that the exact same performance could be captured with each microphone setup. The amp was turned up to a somewhat beefy, but not deafening volume.

The Shure SM57 Microphone: A Guitarist's Trusty Friend

For recording guitar amplifiers, it is hard to find better bang-for-your-buck than the Shure SM57 dynamic microphone. It is an extremely durable microphone, and it can handle very high volume levels. When recording guitar amps, I recommend placing this mic just as close as you can to the speaker. In the following recordings, I have placed it right up against the speaker grill cloth.

— This recording was made with the SM57 facing the speaker straight on (on-axis), with the microphone placed at the outer edge of the speaker.
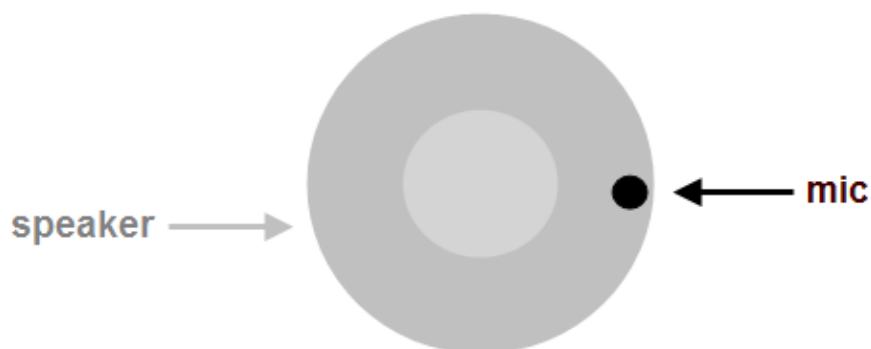
— This recording was made with the SM57 facing the speaker straight on (on-axis), with the microphone placed directly in the center of the speaker. You should be able to easily notice that this recording has a brighter tone than the recording made at the edge of the speaker. Since more high frequency content has been captured, the slight noise/hiss from the looping pedal is more noticeable.
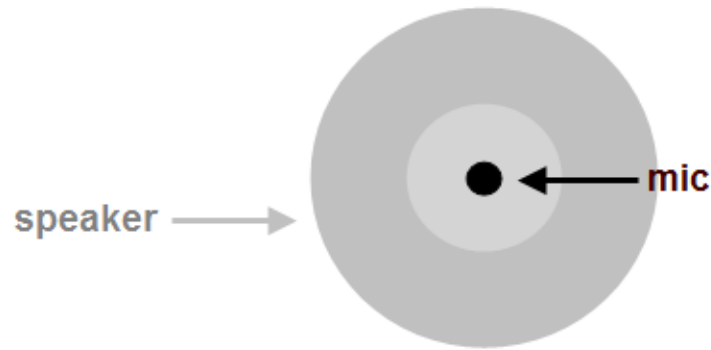
— This recording was made with the SM57 placed directly in the center of the speaker, but facing the speaker off-axis at a 45-degree angle. When compared to the preceding center/on-axis recording, this recording is a bit warmer. The noise floor is a bit less noticeable, and the high frequency content is dialed down a bit.

— This recording was made with the SM57 placed at the edge of the speaker, at a 45-degree angle (pointing toward the center of the speaker). This is the warmest of all the microphone positions recorded here. It has slightly less high-end "sizzle" than the first recording (SM57_edge_straight.mp3).

Though I have not included an example, I will mention that some people even simply hang the microphone so that the cable is draped over the top of the amp and the microphone hangs down in front of the speaker pointing directly at the floor. In this position, the type of floor makes a difference in the tone (wood, concrete, tile, or carpeted floors will result in different sounds), as well as the distance between the speaker, mic, and floor. This arrangement is likely to have the least amount of highs of anything discussed thus far, and will have less rejection of other sounds in the room (if you are recording in the same room as the rest of a band, this method would pick up more sound from the other instruments). I tend to avoid this method myself, but if you are short of mic stands, it could be helpful—and I would not discourage you from experimenting. It may turn out to be just the sound you were looking for.

Large Diaphragm Condenser Microphones

For comparison, I have also recorded the same performance with an M-Audio Sputnik tube microphone. Though not as famous as some microphones by Neumann or AKG, this is a high-quality, well-reviewed large diaphragm tube-driven condenser microphone.

M-Audio states that what they were going for was something with tonal characteristics somewhere between a Neumann U47 and an AKG C12. For the following recording, I have placed the Sputnik directly in front of the center of the speaker, on-axis, 10 inches from the front of the speaker.

— This microphone has switchable polar patterns (including omnidirectional, figure eight, and cardioid). For the recording above, I used the cardioid pattern. This means that the microphone favors sounds directly in front of it, while picking up less sound from the sides and rejecting sounds coming from behind it. The SM57, as well as the AKG C 1000 S microphones used later in this article also have cardioid pickup patterns.

What I hope will impress you here is how for this application (recording a guitar amp), the $100 SM57 compares quite favorably $800 Sputnik! For recording vocals or acoustic instruments, the Sputnik wins hands down—but for recording guitar amps, the SM57 is an amazing value and hard to beat at any price.

Room mics

Though one might often mic a room with a single microphone (perhaps a large diaphragm condenser), I have opted to use a stereo pair of small diaphragm condensers: specifically the AKG C 1000 S (an older dark gray pair). These microphones currently have a street price around $280/each USD. I attached them to a stereo mount on a single mic stand, one microphone just on top of the other, with the microphone elements arranged 90 degrees from each other (an XY pattern). This is to minimize the chance of phase problems I might otherwise encounter that could potentially cancel out some frequencies and change the tone of the recording in strange ways.

X-Y Pattern
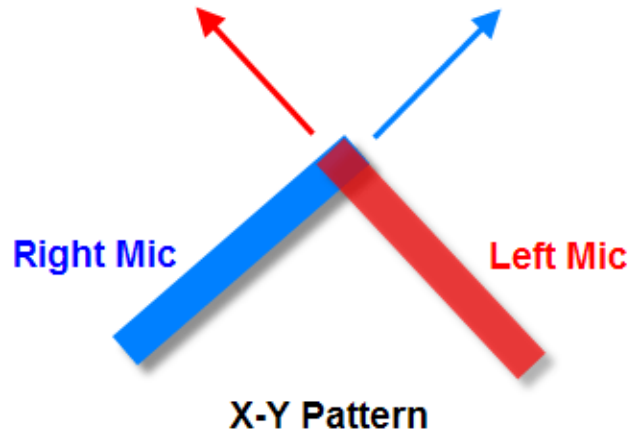
For the following recording, the mic stand was placed about 8 feet from the amplifier, at roughly the height a standing human's ears would be.

— This recording is not immediately as pleasing to my ears as the other recordings. However, when mixed with one of the close-mic'ed recordings, its usefulness becomes clearer.

— Before listening to the next MP3, I suggest prepping your ears by listening again to the SM57 45 degree edge recording

— In this recording, the SM57 recording and the stereo room mic recording are mixed almost evenly (weighted just slightly in favor of the SM57 recording). By adding the room mic pair, there is a bit of depth and also a bit of high-end definition added to the recording. There is more of a sense of space. Of course, this effect could be simulated by adding a digital room reverb (such as the RV7000 Advanced Reverb that comes with Reason). Digital reverbs allow you to simulate rooms that may have more pleasing acoustics or larger dimensions than those of a "bedroom studio" or other room in a house or apartment.

Recording guitar directly into Reason using the built-in Line 6 Guitar Amp device

This is where things get really simple, quick, and easy!

The built-in Line 6 Guitar Amp device has emulations of all types of different amps and speaker cabinets—much more variety than a single tube amp could provide. And of

course you can record at any volume level (or through headphones), so apartment-dwelling musicians won't be visited by the police in the middle of the night for disturbing the peace!

I recorded the same part (actually ditched the looping pedal and played a fresh take for this) into Reason. I plugged the same guitar directly into my audio interface. No effects pedals of any kind were used. Here are the steps I followed, which basically explain how to record a guitar track in Reason:

Here is my result:

I must say that the tone from the Line 6 Guitar Amp sounds quite good to me, even after listening to the Fender tube amp all day!

Of course, there are several totally shredding tones available, much wilder than the warm (but rather tame) tone used here. And for you knob-twiddlers out there, the presets are just starting points: The Line 6 Guitar Amp device has familiar controls just like on a standard guitar amp, so that you can easily adjust drive, bass, middle, treble, and presence just the way you like, while mixing and matching amps and cabinets to your heart's content. Experiment and find what works best for you! Of course if you create a tone you like, you can easily save it as a user preset. And later if you are mixing and decide you'd like to try something else, you can change to a different preset completely. It doesn't matter what Line 6 preset you use when recording—only your clean unaffected signal is actually recorded to your hard drive. You can change the tone anyway you like afterward.

Matt Piper is a guitarist/keyboardist and author of Reason 4 Ignite! His favorite Reason feature? "Reason has the easiest to use, best sounding audio time stretching I have ever tried."

Post tagged as: **Guitar**, **Reason**, **Record U**, **Recording**

Tools for Mixing: EQ, part 1

By Ernie Rideout

For a songwriter or a band, is there anything more exciting than having finished recording all the tracks for a new song? Hardly. The song that existed only in your head or in fleeting performances is now documented in a tangible, nearly permanent form. This is the payoff of your creativity!

Assuming that all your tracks have been well recorded at fairly full levels, without sounds that you don't want (such as distortion, clipping, hum, dogs barking, or other noises), you're ready for the next stage of your song's lifecycle: mixing.

If you haven't mixed a song before, there's no need to be anxious about the process. The goal is straightforward: Make all of your tracks blend well and sound good together so that your song or composition communicates as you intended. And here at Record U, we'll show you how to do it, simply and effectively.

Regardless of whether you've recorded your tracks in computer software or in a hardware multitrack recorder, you have several tools that you can use to create everything from a rough mix to a final mix.

Faders: Use channel faders to set relative track levels.
Panning: Separate tracks by placing them in the stereo field.
EQ: Give each track its own sonic space by shaping sounds with equalization.
Reverb: Give each track its own apparent distance from the listener by adjusting reverb levels.
Dynamics: Smooth out peaks, eliminate background noises, and bring up the level of less-audible phrases with compression, limiting, gating, and other processes.

As you learn more about mixing here at Record U, you'll learn how these tools interact. This article focuses on the most powerful and — for those new to recording, at least — the most intimidating of them: EQ.

In fact, in the course of this article we're going to create a mix using nothing but EQ, so you get comfortable using it right away. But before we go on, we must make you aware of its main limitations:

- It cannot improve tracks that are recorded at levels that are too low.
- It cannot fix mistakes in the performance.

Hopefully you'll discover that EQ can be a tremendously creative tool that can improve and inspire your music. Let's see how it works.

What is it we're mixing?

Before we start tweaking EQ, let's look at exactly what it is we're trying to blend together, which is essential to understanding how EQ works. Let's say we're going to mix a song consisting of drum, electric bass, electric guitar, organ, and female vocal tracks — a very common configuration. Let's focus on just one beat of one bar, when all that's happening is the bass and guitar playing one note each an octave apart, the organ playing a fourth fairly high up, the vocalist sustaining one note, and the drummer hitting the kick drum and hi-hat simultaneously. Here are the pitches on a piano keyboard:

Fig. 1. Here are the fundamental pitches occurring on one beat of our hypothetical multitrack session. Kick drum and hi-hat are dark blue, the bass is red, the guitar is blue, vocals are yellow, and the organ notes are green. With all the space between these notes, what could be so hard about mixing these sounds together?

Let's take a different look at the fundamental pitches of our hypothetical multitrack moment. In this diagram, the musical pitches are expressed as their corresponding frequencies.
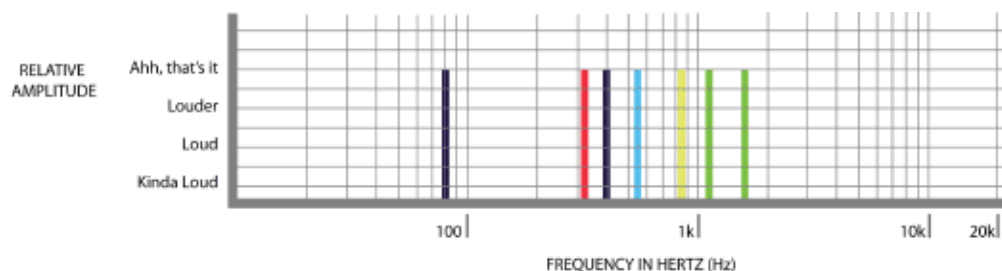

Fig. 2. Here are the fundamental pitches of our hypothetical recording session again, this time displayed as frequencies on a logarithmic display. New to logarithmic displays? The lines repre-

sent increases by a factor of 10: To the left of the 100 mark, each vertical line represents an increase of 10 Hz; between the 100 and 1k mark, the vertical lines mark off increases of 100 Hz; between the 1k and 10k mark, the increases are by 1,000 Hz; above 10k, the marks are in 10,000 Hz. This is to accommodate the fact that each note doubles in frequency with each higher octave; the frequencies add up fast over an eight-octave span. No matter how you count , it still doesn't look like this would be tough to mix. Or does it?

Ah, if only that were so. The fact is, one of the reasons that music is so interesting and expressive is that each instrument has its own distinctive tone. You can see why some instruments sound unique: Some are played with a bow drawn across strings, others have reeds that vibrate when you blow into them, some vibrate when you hit them, and others make their sound by running voltage through a series of electronics. But why would that make a group of instruments or human vocalists any harder to mix?

Instruments sound different because the notes they make contain different patterns of overtones in addition to the fundamental frequency: Each instrument has its own harmonic spectrum. Sometimes the overtones are not very loud and not numerous; with some instruments the overtones can be just as loud as the fundamental, and there can be upwards of a dozen of them. Let's take a closer look at the notes of our hypothetical recording session, this time with all of the overtones included.
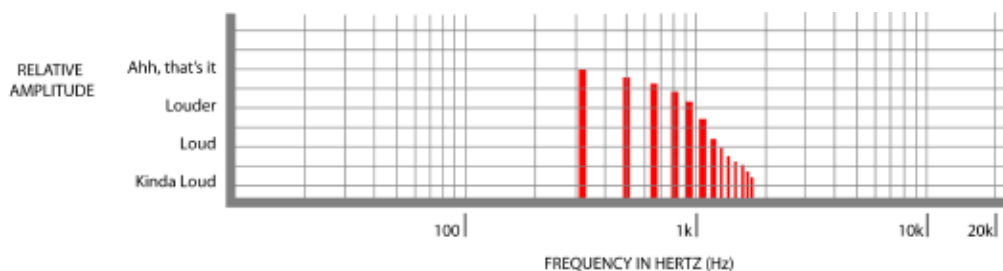
Fig 3. Here's what the harmonic spectrum of that single bass guitar note looks like.
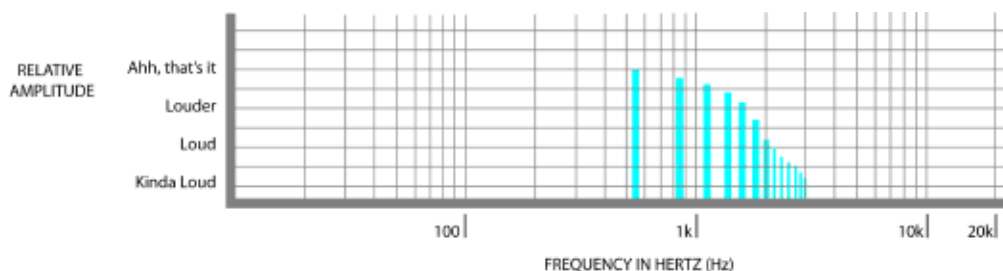
Fig 4. The harmonic spectrum of our electric guitar note might look like this — and this is through a clean amp!
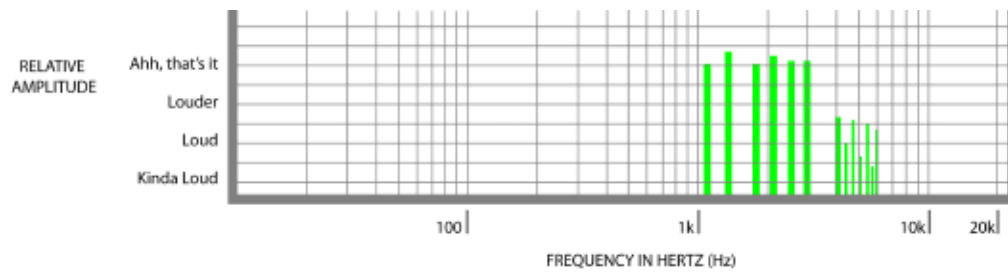
Fig. 5. The fourths the organ player is holding yield a harmonic spectrum that's even richer in overtones than the guitar.
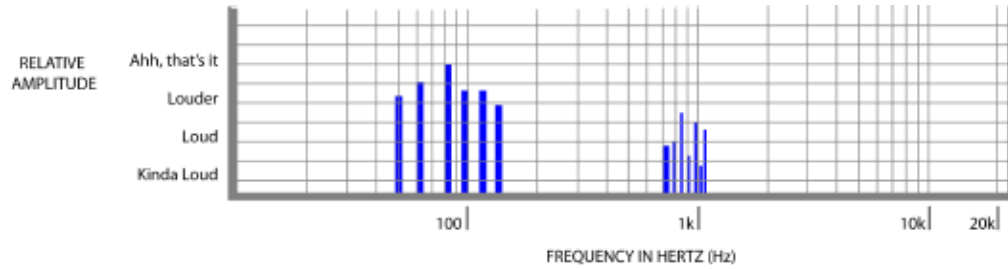
Fig. 6. Though they're not necessarily tuned to a particular pitch, the kick drum and hi-hat have a surprising number of overtones to their sound.
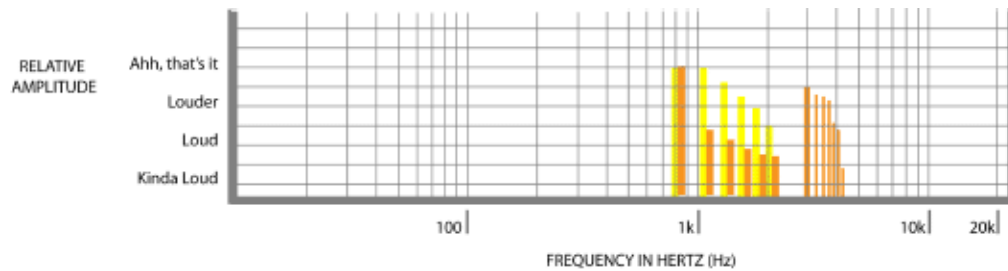
Fig. 7. If our vocalist sings an "oo" vowel, her note will have the overtones in yellow. If she sings an "ee" vowel, she'll produce the orange overtones.
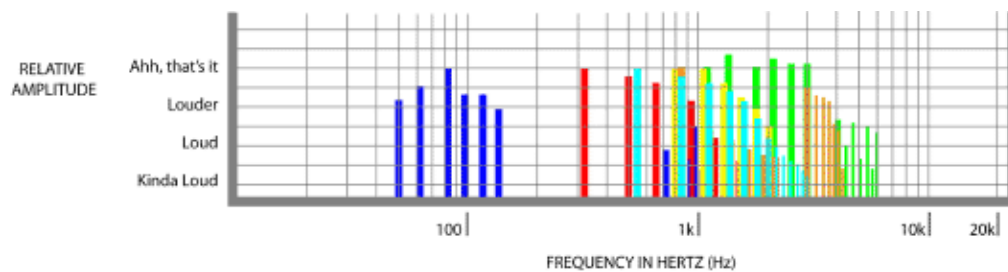
Fig. 8. Let's put the whole band together for our hypothetical one-note mixing job. Yikes. That's a lot of potentially conflicting overtones, and none of the tracks are even similar to each other in tone! It looks like this is going to be one muddy mix, unless we apply some EQ!

It seems that our simple hypothetical multitrack mix assignment might not be so simple after all. All of those overlapping overtones from different instruments might very well lead to a muddy, indistinct sound, if left alone. Fortunately, even a seemingly cluttered mix such as this can be cleared up in a jiffy by applying the right kind of EQ techniques.

There are several types of EQ, each of which applies a similar technique to achieve particular results. However, the terms that you may hear or read about that describe these results can vary widely. You'll often hear the following terms used, sometimes referring to particular EQ types, at others referring to generic EQ applications.

- Attenuate
- Bell
- Boost
- Carve out
- Curve
- Cut
- Cutoff
- Filter
- Flat
- Response
- Rolloff
- Slope
- Spike
- Sweep

As we go through the various types of EQ, we'll define exactly what these terms mean and get you acclimated to their usage. We'll also illustrate each EQ type with audio examples, harmonic spectra, and plenty of sure-fire, problem-solving applications.

Continue to part 2 —>

Post tagged as: **EQ**, **Mixing**, **Reason**, **Record U**

Tools for Mixing: EQ, part 2

EQ types: high pass and low pass filters

You're probably very familiar with the simplest type of EQ:

Fig. 9. Though an electric guitar's tone knobs can be wired to apply many different types of EQ to the sound of a guitar, at its most basic, turning the knob applies a low pass filter to the sound, gradually lowering the level of harmonics at the higher end of the frequency spectrum.

Here's a low E on an electric guitar, with the tone knob at its brightest setting. This allows all the frequencies to pass through without reduction:

Crank the tone knob down, and higher frequencies are blocked — or rolled off — while lower frequencies are allowed to pass through; hence the name, low pass filter:

Just as the low pass filter attenuates (reduces) high frequencies and allows low frequencies to pass through, there is another EQ type that rolls off (reduces) low frequencies while allowing high frequencies to pass: the high pass filter. It's not just guitars that utilize this simple EQ type. When you're mixing, usually you'll use high pass and low pass filters that are built into each channel of your mixer, which allow you to set the frequency at which the attenuation begins (also called the cutoff frequency).

Although their structure may be simple, low pass filters can be very effective at quickly providing solutions such as:

- Removing hiss in a particular track
- Isolating and emphasizing a low-frequency sound, such as a kick drum

Similarly, high pass filters can quickly give you remedies such as:

- Eliminating rumble, mic stand bumps, or low frequency hum on tracks that primarily have high-frequency energy in the music

Here's how Reason's low pass and high pass filters work.

Fig. 10. Reason provides a low pass filter (LPF) and a high pass filter (HPF) on each mixer channel, at the very top of the EQ section. You engage them simply by clicking their respective "on" buttons, and then you can set the frequency above or below which the attenuation begins.
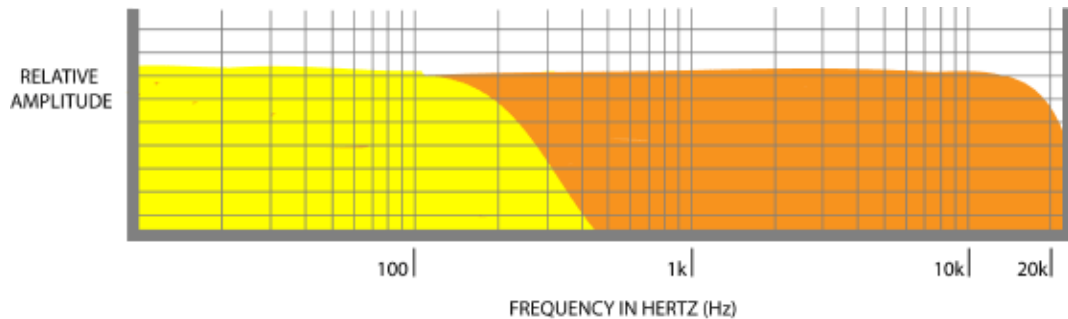
Fig. 11. Reason's low pass filter lets you set the rolloff point from 100 Hz up to 20 kHz. The angle of the roll-off has particular terminology and characteristics: The angle itself is called the "cutoff" or "knee," and the "slope" or "curve" reduces the signal by 12 dB per octave. Here you can see how much is allowed to pass at the lowest setting (yellow area only) and how much passes at the highest setting (orange and yellow).

This is the sound of a low pass filter sweeping over a drum loop:
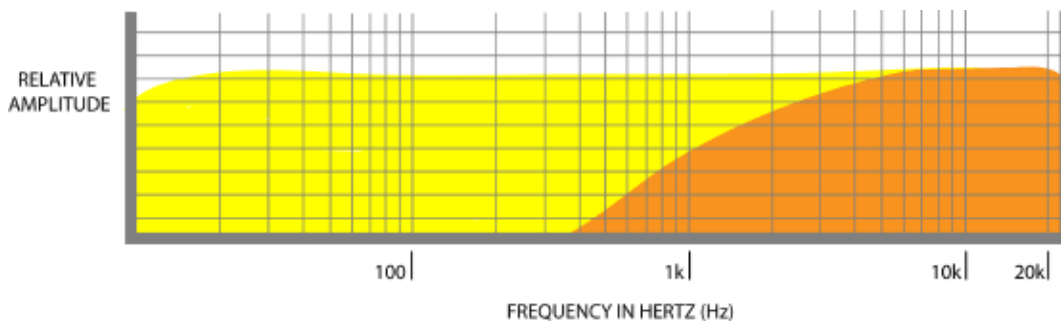
Fig. 12. Reason's high pass filter rolls off frequencies below 20 Hz at its lowest setting (allowing frequencies in the yellow and orange areas to pass) to 4 kHz at its highest (the orange area only). The slope attenuates 18 dB per octave.

This is the sound of a high pass filter sweeping over a drum loop:

EQ types: shelving

If you've ever had a stereo or car radio with bass and treble controls, shelving is another type of EQ you're already familiar with. (If all you've ever had is an iPod, we'll talk about the type of EQ you use the most — graphic EQ — in a bit.) Shelving EQ is usually

used in the common "treble and bass" configuration. It's also used at the upper and lower ends of EQ systems that have more than just two bands. Like low pass and high pass filters, shelving EQ works on the "from here to the end" model: Everything below a particular point (in the case of a bass control) is affected, and everything above a particular point (in the case of a treble control) is affected. The difference is that shelving EQ boosts or cuts the levels of the affected frequencies by an amount that you specify; it doesn't just block them entirely, which is what a pass filter does.

Shelving EQ is the perfect tool to use when a track has energy in one of the extreme registers that you want to emphasize (boost) or reduce (cut), but you don't want to target the specific frequencies or eliminate them entirely. It lets you keep the overall level of the track at the level you want compared with your other tracks in the mix, while giving you a quick way to distinguish or disguise the track. Some useful applications include:

- Percussion tracks often have energy in the extreme low and extreme high frequency areas; shelving EQ can easily bring that energy to the fore of your mix or cut it to make room for the sound of another track
- Synth bass parts make or break a dance track; a little boost with shelving EQ can quickly transform a dull track to a booty-shaker
- Adding a high shelf to a drum kit and then cutting by a few dB gives the kit a muffled, alternative mood

Let's take a look at and give a listen to the ways that shelving EQ works.

Fig. 13. The two-band EQ on the Reason's 14:2 Mixer device is a classic example of treble and bass shelving EQ. Turning a knob clockwise boosts the affected frequencies, and turning a knob counterclockwise cuts the affected frequencies.
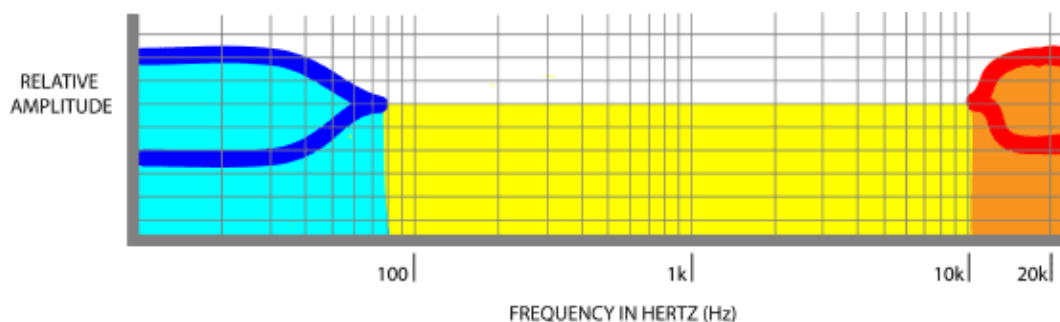

Fig. 14. This diagram gives you an idea of how shelving EQ differs from simple pass filter EQ, using the specs of the 14:2 EQ. The middle of the yellow area is the part of the track that is unaffected by the EQ; you'll still hear the frequencies in this area at the same level, even if you boost

or cut the frequencies in the shelving areas. The blue area shows the frequencies affected by the bass control: Below 80 Hz, you can cut or boost the frequencies by up to 24 dB (dark blue lines). The orange section shows the area affected by the treble control: Above 12 kHz, you can boost or cut by 24 dB (red lines).

On this drum loop, first we'll cut the high shelving EQ, then we'll boost it. Next we'll cut the low shelving EQ, then we'll boost it:

Fig. 15. Reason also gives you shelving EQ on each mixer channel. This EQ has a bit more control than that on the 14:2 mixer, as it allows you to specify the frequency above or below which the track is affected. For the high shelf, you can adjust the cutoff frequency from 1.5 kHz to 22 kHz. The bass shelf cutoff can move from 40 Hz to 600 Hz. In both cases, you can cut or boost by 20 dB.

EQ type: parametric EQ

So far, our EQ tools have been like blunt instruments: Effective when fairly wide frequency bands are your target. But what if you have just a few overtones that you need to reduce in a single track, or an octave in the mid range that you'd love to be able to boost just a tad across the entire mix?

That's where parametric EQ comes in. Parametric EQ usually divides the frequency spectrum up into bands. Some EQs have just two bands, like the PEQ-2 Two-Band Parametric EQ device. Some EQs have as many as seven or eight bands. Usually, three or four bands will give you all the power you need.

For each band of parametric EQ, you can select three parameters: the center frequency of the band, the amount of cut or boost, and the bandwidth, which is often referred to as the "Q". Adjustments in Q typically are expressed on a scale that approaches 0.0 and 3.0 at either end, with 0.0 being the widest bandwidth and the gentlest slope and 3.0 being the narrowest bandwidth with the steepest slope. Let's see what shapes parametric bands can get.

Fig. 16. This is Reason's channel strip EQ, highlighting the parametric EQ.
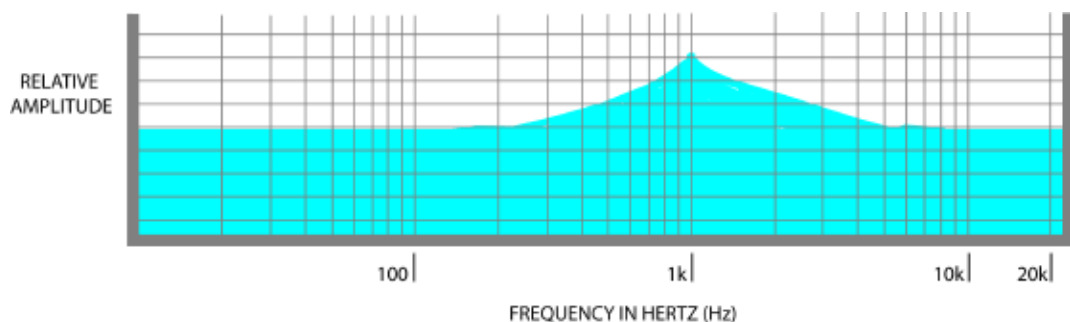
Fig. 17. This is the curve of a band of parametric EQ, centered on 1 kHz and given its highest boost with the widest Q setting. Looks like a nice, graceful old volcano.
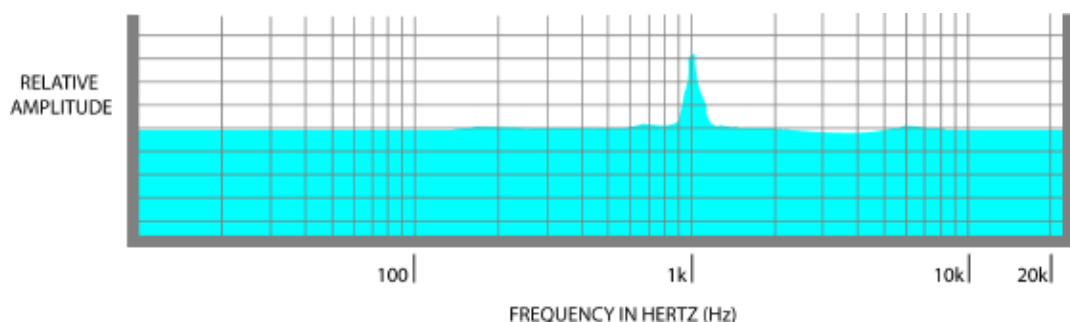
Fig. 18. This curve has the same center frequency and boost amount as the previous diagram, but with the narrowest bandwidth, or Q setting. This is what's known as a spike.

The illustrations above are meant to show you the full extent of the power that parametric EQ can offer. Normally, there's no need to use all of it. In the vast majority of cases, just a tiny bit of EQ adjustment will have a huge effect on your music, no matter what type of EQ you use.

But for now, let's use this power to explore the concept of overtones a little further. Sometimes it's hard to believe that every musical sound you hear — with the exception of the simplest waveforms — consists of a fundamental frequency and lots of overtones, each at their own frequency. An instrument playing a single note is just a single note, isn't it?

Let's take a listen to a single pitch played on a piano. We'll hit each note hard and let it decay for just a moment. While we play, we'll take one of those narrow bands of parametric EQ, the one with the spiky shape, and we'll boost it and sweep it up and down the frequency spectrum. Listen closely, and you'll hear several overtones of this single pitch become amplified.

Sweeping a piano with an EQ spike:

Sweeping the EQ spike across the single piano pitch makes it sound more like an arpeggio than a single pitch. All those "other notes" are the overtones. Even in one single note, you have a multitude of frequencies that give the sound its character. In fact, when you hear a person describe a sound as something that "cuts through in the mix," they're usually referring to a sound that's rich in overtones, more than a sound that's simply louder than all the rest of the sounds in the mix. Usually it refers to a live performance P.A. mix, but the idea applies to what we're doing here, too.

Let's experiment with cutting overtones, using the same EQ curve, but reversed. Listen to the same repeated piano note as we sweep the parametric EQ across the frequency spectrum.

Sweeping a narrow parametric cut on a piano sound:

As we swept the EQ cut up and down, you could hear the piano tone change radically. Sometimes it sounded hollow, sometimes as though it had too much bass. But there were other times when it still sounded good, even though we were reducing the sound of particular overtones.

The key to creating a great sounding mix is to know how to cut or boost overtones on each track to make each instrument have its own sonic space, so that its overtones don't interfere with other tracks, and other overtones don't interfere with it. Using parametric EQ to cut particular overtones in one track to make room for the overtones in another track is the one of the most effective ways to do it.

Let's create a mix right now, using only the types of EQ we've discussed so far. We won't touch any faders, nor will we pan any tracks. Just by manipulating overtones by cutting with EQ — we're not even going to boost anything — we'll turn a muddy mix into a clear one.

Create a mix using only EQ

Here's an excerpt of the raw, unmixed multitrack of a blues session, recorded using Reason. Check out the raw tracks.

Raw blues tracks excerpt:

Usually when you think of Muddy and the blues, you think of the great Muddy Waters. This session, on the other hand, was the other kind of muddy. The kick drum, bass, and rhythm guitar are overwhelmed by the thick sound of the horn section. The solo organ and tenor sax parts don't overlap, but their tone sure does. Sounds like we have our work cut out for us.

Where do you start? Since the bass and kick drum are the least audible, you might be tempted to bring up their levels with a shelving EQ. Don't bring any levels up! We're on a mission to cut and carve out space for each part.

In fact, just to get you comfortable with using EQ, we're not only going to cut, we're going to cut a lot, using as much as the full 20 dB of range! Normally, just a couple dB will do the trick. But let's go hog wild, just to prove that EQ can't hurt you.

Let's start with the mud. The horn section sounds great, but they're not the most important thing going on. We'll solo them along with the part they seem to be obliterating the most: the guitar.

Guitar and horn section soloed:

Yes, that's muddy. The guitar itself is boomy as heck, too. Using the low mid frequency parametric EQ band on each track's channel strip, let's employ our narrow-Q parametric EQ cut-sweeping trick. The guitar sounds much better with a big cut at 260 Hz, and the horns open up for the guitar when they're cut at 410 Hz for the tenor sax, 480 Hz for the alto sax, and 1 kHz for the trumpet. The saxes benefit from a slightly wider Q. Here's how just these four sound, with EQ cuts.

Guitar and horn section with EQ:

Fig. 19. These are the low mid frequency EQ settings for the guitar, tenor sax, alto sax, and trumpet, from left to right, skipping over the darker channel. The amount of cut is radical, but it's for educational purposes. Plus, it sounds pretty good in the mix, as you'll hear.

Overall, these four tracks sound a bit thin, especially the horns. Keep in mind we're overdoing the EQ amount. But the horn section still sounds like a horn section, the guitar is audible finally, and you can hear the bass guitar now, too. The organ and solo tenor sax are way out in front now, which gives us some room to maneuver with them.

Guitar and horn section with all tracks:

Since we want the tenor sax to be the more prominent of the two, let's solo the two tracks and then carve some space for them.

Tenor sax solo and organ fills:

It's obvious these two instruments are stepping on each other's sonic toes, so to speak. Let's apply our narrow-Q parametric EQ cut-sweeping trick to the organ, and cut it at 412 Hz or so, which sounds the best after the up-and-down sweep. That opens up the tenor sax, and yet the organ is still very audible. Since the tenor sax has such a rich sound, let's apply a low shelf filter cut to it, right at 155 Hz. Now the balance sounds right between the two solo instruments, and there's probably more room for the rest of the mix, too.

Solo sax and fill organ, EQed:

Fig. 20. These are the low mid frequency EQ settings for the organ fills on the right, and the low filter shelving settings for the tenor sax on the left.

Now the mix is really opening up. You can hear the electric bass and drums much more clearly, and though the tenor sax is definitely front-and-center, you still hear the horns, guitar, and organ.

Solo sax and fill organ with all tracks:

Since the drummer is using the ride cymbal, there is a lot of high-mid energy in the drum part. Let's see if we can cut some of that and still have the nice "ping" of the stick on the cymbal — using the narrow-Q parametric EQ cut-sweeping trick, of course. First, the drums, soloed:

Raw drums, soloed:

After sweeping up and down on the soloed drums, we didn't really find a setting that removed the "hiss" but preserved the "ping" of the ride cymbal. So we applied a low pass filter instead, with a cutoff setting of 4.9 kHz. This got us the sound we were after.

Drums with low pass filter:

Fig. 21. A low pass filter worked best on these drums; this is the setting.

Now let's listen to the mix with all of our EQ work in place:

All tracks with EQ.

All the tracks have their own sonic space. Every instrument is audible, and no track is obscuring any other track. It's a pretty good rough mix, and we didn't even make use of panning, reverb, dynamics, or level settings! That means that the distance from this mix to a final mix is much closer than it would have been if we'd started with adjusting levels — and we still have all the fader headroom to work with!

To be sure, we applied EQ in far too generous amounts. But that was just to show you how powerful yet easy it is to make a muddy mix into a transparent one.

It's also good to note that Reason's mixer is perfect for this kind of EQing: The controls are all visible at all times, and you can see which track might still have room for a center

frequency adjustment.

But even more impressive is what you can't see, but you can certainly hear: Reason's EQ section sounds fabulous. It's very smooth, and very musical.

EQ types: graphic EQ

If you're an iTunes user, you know all about graphic EQ. It's similar to parametric EQ in that you target specific bands of frequencies to cut or boost. But the frequencies are hard-wired. Some graphic EQs have as few as five or six bands, while others divide the audio spectrum up into 30 or more bands.
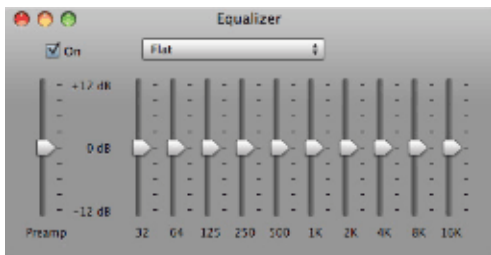
Fig. 22. This is the graphic EQ in Apple iTunes. Graphic EQ is great for applying an overall setting to an entire concert, or to a lot of songs in a particular genre.

Graphic EQ is most often used in live P.A. work, where the room is subject to resonance points that are constant. A graphic EQ lets the engineer identify those points quickly and then cut those risky frequencies from the entire mix for the rest of the night.

In mixing, you need EQ that you can tailor more to the music itself, which is why parametric EQ is what you find on great mixing boards. Like the one in Reason.

EQ types: mastering EQ

Once your mix has been perfected by your skillful use of EQ, dynamics, reverb, panning, and level settings, it's ready to be mastered — which is another way of saying it's ready for the final polish. Usually, parametric EQ is used at the mastering stage, but it's used very sparingly. It's great for bringing out some of the middle frequencies that perhaps seem to ultimately get lost in the mix after all your work.
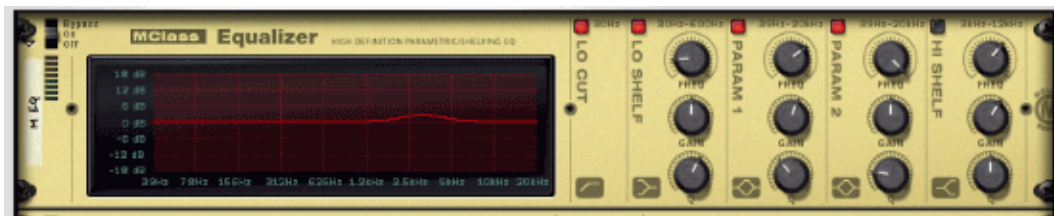
Reason has a special EQ device for mastering: the MClass Equalizer. It's automatically

patched in to the insert effects in the master channel, along with the other MClass devices: Stereo Imager, Compressor, and Maximizer. Used together with subtle settings, the MClass devices can add a magical, professional polish to your mix.

Ernie Rideout is currently writing Reason Power!, scheduled to be published in early 2010 by Cengage Learning. He grapples with words and music in the San Francisco Bay Area.

Post tagged as: **EQ**, **Mixing**, **Reason**, **Record U**

Tools for Mixing: Essentials of Dynamics Processors

By Ernie Rideout

Mixing a song that you've lovingly composed, arranged, performed, and recorded can be a tremendously satisfying experience. But if you're unfamiliar with the tools and techniques available for creating a mix, you might feel a bit of anxiety as you sit at your computer or in front of your mixer, staring at all those faders, knobs, and processors, wondering what to do.

Be anxious no more! You've come to the right place. Record U will provide you will clear explanations and practical advice for making your music sound amazing. The good news is that whether you've recorded your tracks in computer software or in a hardware multitrack recorder, you have all the tools you need to create everything from a rough mix to a final mix. Let's survey these tools briefly:

- **Faders:** Use channel faders to set relative track levels.
- **Panning:** Separate tracks by placing them in the stereo field.
- **EQ:** Give each track its own sonic space by shaping sounds with equalization.
- **Reverb:** Give each track its own apparent distance from the listener by adjusting reverb levels.
- **Dynamics:** Smooth out peaks, eliminate background noises, and bring up the level of less-audible phrases with compression, limiting, gating, and other processes.

There are a few important things to keep in mind about all these tools:

1. They can't fix poorly recorded material
2. Any change you make with one tool can affect the results of the other tools
3. The tools were originally designed to do a simple job — but they're actually full of fun creative potential, too!

In this article we're going to focus on the tools that many musicians consider to be boring: dynamics processors. But when used intelligently and in tandem with careful listening — that's the most important tool you have, those ears of yours— dynamics processors can turn an ordinary mix into an extraordinary one. And they're full of tricks that you can use to make amazing sounds you never imagined.

A word of warning: In the course of this article we'll be taking dynamics processing to

extreme degrees to demonstrate the effects that are possible. For most music, the best application of dynamics processing is the most subtle and gentle. A little can go a long way. Always check your dynamics processing work by listening to the effect in the context of the entire mix. Use the "bypass" switch on the device to compare the processed effect with the non-processed sound. This will help make sure that you keep the intent of your music in mind as you mix!

Let's take a tour of what kinds of processors fall under the "dynamics" heading. Then we'll fiddle with the knobs!

What are dynamics?

It's the same as in your music: Dynamics are changes in loudness for expressive purposes. This category of processing is called "dynamic" because that's the aspect of sound it's designed to address — relative loudness and the ways we perceive it. The operative part of that definition is "perception." Our brains have very funny ways of processing differences in loudness. Louder is better, for example — have you ever felt that way? Most musicians have, especially when they're in front of a mixing board. But it's not necessarily so; any increase in loudness in one track can mask more serious problems in other tracks that can continue to cause problems in the mix, even more so once they're harder to pinpoint. Among other potential problems.

The point of this cautionary tale is this: Dynamics processors use numerical values for their parameter settings, but they require your careful listening to do their jobs properly. They're not magic.

Well, some are. We'll get to those in a bit.

So what exactly do these things do with the dynamics of your music? All kinds of things, such as:

- Making bits that are too loud softer
- Making bits that are too soft louder
- Making unwanted noise between notes and phrases disappear
- Making just one part of the frequency spectrum louder or quieter
- Allowing note attacks to pass through unaffected while making the sustained parts louder
- Increasing the sustain of certain sounds

- Allowing certain sounds to increase or decrease in loudness only when other sounds occur
- Emphasize certain parts of the audio spectrum so that you say, "Wow, that just sounds . . . better!"
- Change their behavior depending on the performance dynamics of the music

Dynamics processors come in several flavors. Interestingly, depending on how you set the parameters on some types, they automatically change to a similar corresponding type. Here are the main types:

Compressor

What gets "compressed" by this process is the range of volume between the softest and loudest levels in a track. At its most basic, a compressor makes the loud parts softer. This process was invented to make some types of material that naturally have a wide range of dynamics — such as singing — more consistently audible compared with its accompaniment, especially when its target was for radio broadcast. It's often used for vocals, drums, and entire mixes. Most compressors analyze a sound only by its dynamics. Some compressors — called multiband compressors — divide up the sound by frequency, and then apply the compression separately for each frequency band.
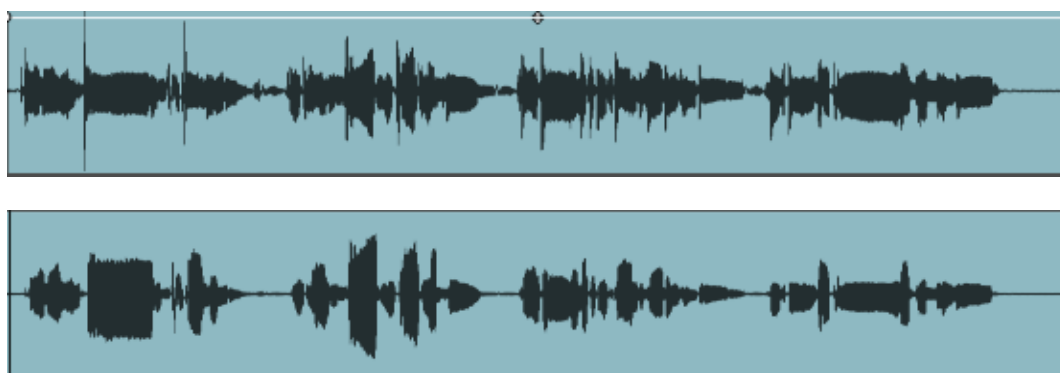

Fig. 1. These two waveforms show the effect of compression. The top waveform is without compression; notice the distance between the transients (the beginning of the notes) and the sustained portions. On the bottom is the same phrase compressed. You can see that the attacks and peaks are much softer compared to the original waveform. We'll listen to examples of this in a bit.

Limiter

Compressors and limiters are often one and the same device; the difference is in how you set up the controls. Limiting affects only the loudest parts of a performance and reduces their volume. Sometimes this is used on the beginnings of notes with very loud transients. Sometimes it's used to reduce the volume of entire sections of music. Generally the goal is to prevent clipping or distortion in the signal chain.

Gate

Another aptly named process, gating prevents unwanted sounds that occur during pauses between notes or phrases from being audible. Often this is used to eliminate bleedthrough, which is the sound of other instruments that were being played in rooms or isolation booths nearby, and were therefore picked up at a low level during pauses.
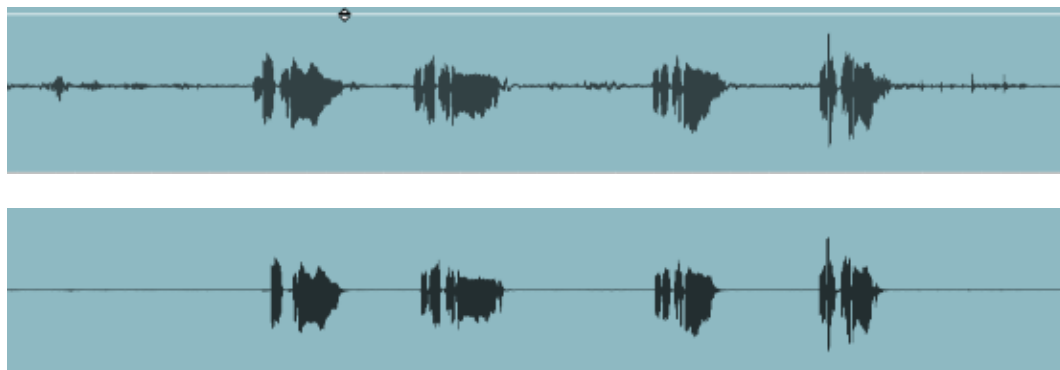
Fig. 2. In the un-gated top waveform, you can see the low-level background noise in between the instrumental performance. In the bottom waveform, the background noise has been eliminated by a gate processor.

Expander

Gates and expanders go together. Where a gate will close entirely to prevent soft sounds or noise from being audible, expanders simply reduce the volume of a track during pauses. An expander does the opposite of a compressor: The compressor finds loud sounds and decreases their volume, making the peaks closer to the softer parts in dynamic range. An expander finds the soft sounds and reduces their volume, making the peaks farther away from the softer parts in dynamic range. The goal of expansion is to get the best "signal to noise ratio" in a track: You'll see this written as "S/N" or "SNR" in specification lists, which means "sounds you want/background noise you don't want." This ratio is usually expressed as a measurement in dB (decibels) such as 100 dB, which means the signal is 100 times stronger than the noise. The bigger that number, the

better the S/N, and the better the sound quality. The good news is that you don't need to know any of this tech stuff to use an expander!

Maximizer

A maximizer is one of those processor types that at least seems like magic. Maybe that's overstating the case a bit, but the fact is, processors such as maximizers and enhancers use compression and limiting processes that are optimized to achieve particular results, without forcing the user to fiddle with the entire range of compressor or limiter settings. You put your track or full mix through them, and they just sound . . . better. Specifically, maximizers limit the peaks and increase the perceived loudness of a track or song.

How do dynamics processors work?

Dynamics processors do a variety of tasks, but they all work on the same basic principles:

1. They sort sounds out according to whether a sound exceeds a threshold or not
2. They apply their effect to sound that's above or below the threshold
3. The timing of the effect depends on settings for attack and release
4. The amount of the effect is determined by settings for a ratio (for compression/limiting) or a range (or range of attenuation; gating/expansion)

Let's look at how a threshold setting affects the way dynamics processors work.
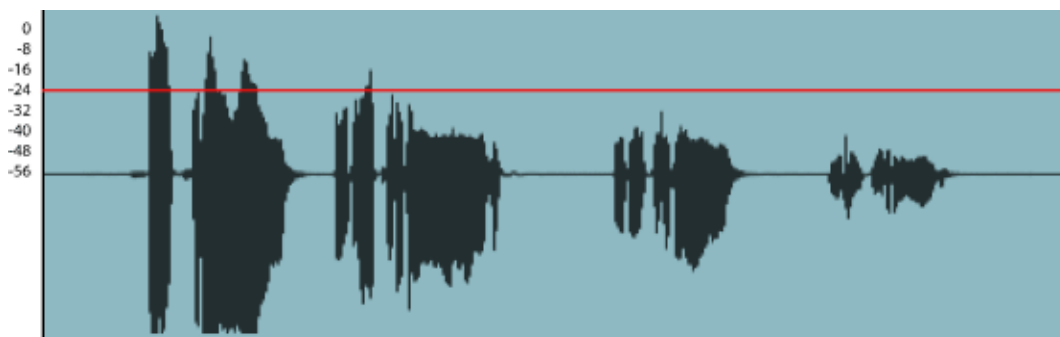
Fig. 3. Here are four sounds at a variety of dynamic levels. We've set the threshold for around -24 dB; our volume scale indicates the relative volume that corresponds with that of the channel faders in Reason. Sounds A and B are above or are mostly above the threshold, whereas sounds C and D are below it.
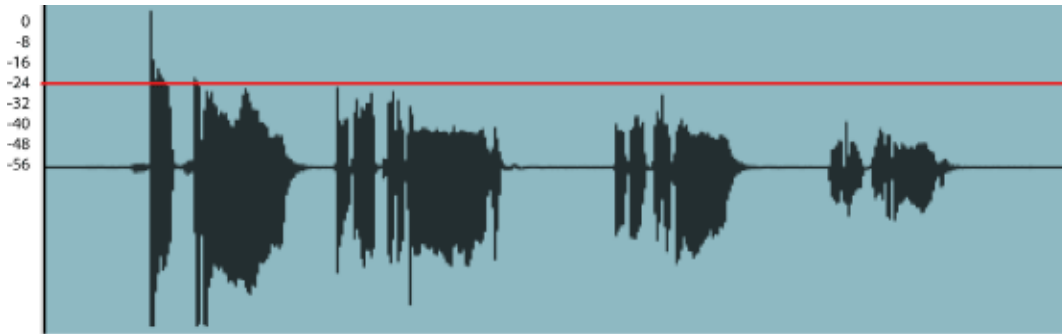
Fig. 4. If we apply compression, it affects the sound above -24 dB. You can see that the volume of sound A has been reduced, and that the beginning of sound B ha, as well. Sounds C and D are not affected. The result is that the range between the loudest sounds and the softest has been compressed.
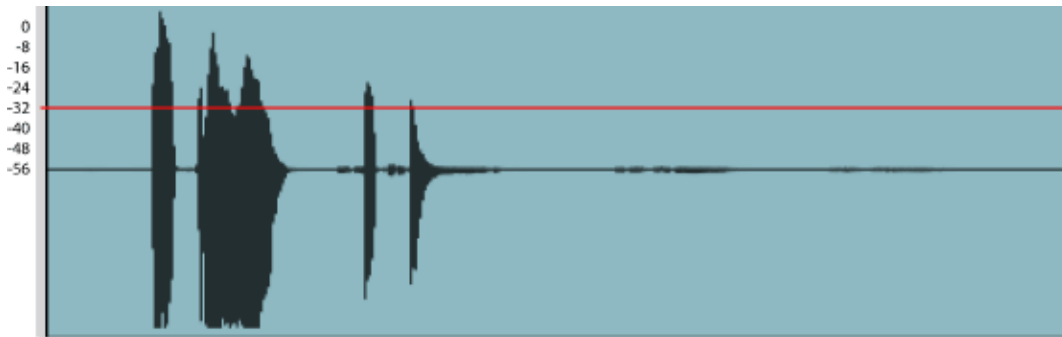
Fig. 5. When we apply gating to the same sounds with the same threshold setting, we get some dramatic results. Sound A passed through entirely unaffected. Sound B got cut off when its volume fell below the threshold. Sounds C and D have been gated completely, since they were entirely below the threshold. There are ways to use gating so that it doesn't chop off notes, and there are ways to use it so that the chop becomes a cool effect. More on these in a bit.

Attack and release settings let you adjust dynamics processors to work best with your music. A fast attack time means the effect is applied fully the moment that the sound exceeds the threshold. A slow attack time tells the processor to apply the effect gradually (relatively speaking), ramping up to the full effect. Let's look at how these settings can affect your music.

Fig. 6. This sound has a loud, fast attack transient and a long decay.

Fig. 7. With the attack and release set for fast response, the attack transient gets compressed right off the bat, as does a portion of the sound that follows the transient. When the signal dips below the threshold, the decay actually seems to jump up slightly in volume.

Fig. 8. With a slow attack, the compressor effect doesn't take hold until just after the attack transient. Then the sustained part of the sound is compressed, until it falls below the threshold. At that point, the compressor doesn't just stop working suddenly, it eases off gradually. The result is much smoother, and the attack transient sounds natural.

The ratio setting on a compressor is where you dial in how much compression occurs once the volume exceeds the threshold. Usually this is demonstrated with a diagram that illustrates how loudness changes as compression is applied above a threshold. But we'd rather show you how it looks on your music. Here is a series of identical snare hits, repeated over and over at precisely the same volume. The only thing that changes in Figure 10 is the compression ratio, which we'll dial as the example plays from 1:1 (no compression) all the way up to infinite compression (maximum; also known as brick wall compression).

Fig. 9. Here are the snare hits, with a compression ratio of 1:1, that is, no compression.

Fig. 10. The only thing that changed for this waveform was that we gradually increased the compression ratio from 1:1 to infinite. As you can see, higher ratio settings can have a dramatic ef-

fect.

Using dynamics effectively in your music is a matter of trying out combinations of these settings. You may find that for your music subtle settings work great, making your mixes sound polished and professional. Or you may enjoy using dynamic processors as an extreme sound design tool, making such effects as compression pumping and dramatic gating part of your sound. In the next section, we'll have some listening examples that demonstrate a variety of dynamics processing applications, from basic to wacky. We'll use the excellent dynamics processing available in Reason, both within the mixer channel strip and in the included devices.

How do I use compressors in my music?

One of the most common applications of compression is for vocal tracks. The human voice is incredibly expressive, and part of that expressivity is the wide range in volume we can employ within a phrase or even within a single word. Consider the following phrase.

This solo female singer uses a full range of expression, including dynamics:

Why would you want to change that performance at all? It's beautiful the way it is, in the context of a solo. The decision of whether to apply dynamic effects should always be based on your musical sensibility, and on what you hear. Once you know all there is to know about dynamics processing, let your music tell you what it needs, rather than using these fabulous tools just because they're there.

That said, consider the following context for the same vocal

Here's the same solo female, this time backed up by a smokin' rhythm section:

This track isn't just telling us the vocal needs compression — it's yelling for it! The softer words and the ends of the phrases get lost in the sound of the band. Let's apply some gentle compression from the channel strip in Reason. Here are the settings:
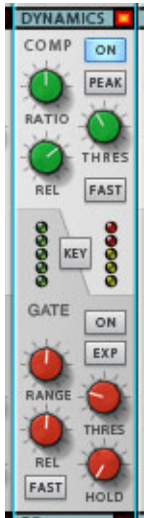
Fig. 11. In Reason, the channel strip features a very smooth, fine-sounding compressor/limiter and gate/expander. Let's turn on the compressor, set the threshold for around -24 dB, leave the attack slow, set the release on the long side, say around 700ms, and dial in a fairly substantial ratio of 4:1, since we've got a lot of sonic competition in the track.

Ahh. Much better. The subtle compression we added in Reason's channel strip makes the vocal sound nice and tight, and perfectly audible — without any adjustment to faders or EQ.

Another common application of compression is to take background parts and squish 'em to make them sound more like a single instrument or voice, even though there may be very thick chords being sung or played. Here's an excerpt from a blues track.

In this blues track, notice how the horn section gets a little lost in the mix, and how it sounds like three individual sounds:

If your artistic intent is to communicate the individuality of the three horn players in the section, then let them sound independent. But if you really wish they sounded more cohesive, try some excessive compression.

This time, we're going to set up Reason's MClass Compressor device as an insert effect, and we'll send the three horn tracks through this compressor. The attack on each note is important, especially on the trumpet, so we'll use a slow attack on the compressor to preserve them. The notes are warm and breathy, so we don't want really obvious compression artifacts; the soft knee setting lets the compression build gradually, rather than suddenly, so we'll use this as well. To really get a squashed sound, we'll dial up the ratio to nearly the highest setting. Then we'll have a gentle release. Finally, we'll add some output gain to make up for volume lost by the heavy compression. Here's how to accomplish this.

Fig. 12. To set up a background section as a group, add a 12:4 mixer to your rack, then add an MClass Compressor to the rack; the compressor is automatically wired into the aux sends and returns. Then take the direct outs of each instrument or background voice channel, and connect them to their respective channels in the 14:2 mixer. Finally, set all three instrument Aux 1 sends on the 14:2 to maximum, and crank the Aux 1 return above the 14:2's master fader. Now all these tracks can be controlled with one fader, and they all go through the same compressor settings.

Fig. 13. We wanted to really squash the horns, but also to give their note attacks some room. Threshold: -30 dB, soft knee. Ratio: 60:1. Attack, 36 ms; release, 100 ms. The MClass Compressor has a very smooth sound, perfect for this type of application.

The massive compression we applied to the horn section as a subgroup helped bring it together as a single sound:

Dynamics processors usually perform their duties based on the audio that's passing through their circuitry. However, there is a groovy feature that allows you to control the effect a compressor has on a track — by using the audio from a separate track. It's called **sidechaining**, and it has a variety of uses, from dance tracks to voiceover work.

The devices in Reason are designed to take advantage of the musical power inherent in sidechaining, and it's very easy to set it up. If you have Reason installed as well, sidechaining opens up entire worlds of creative possibilities, as you can use signals from just about any instrument or device to control the sound of many other instruments.

Let's say you're doing the voiceover for a podcast, and you want to have music playing in the background while you speak. Normally, every time you start to talk, you have to pull down the fader for the music bed. With sidechaining, you can trigger a compressor with your voice, but have the effect apply to the music bed, lowering its volume while you speak. This technique is called "ducking," and with Reason's Audio Track Devices, it's very easy to set it up.

Fig. 14. Each Audio Track Device has inputs expressly designed for sidechain input from another device. On the back of the voiceover track, take the Insert FX "To Device" output and connect it to the Sidechain Input on the back of the music bed Audio Track Device. Then turn on the compressor in the music bed's channel strip and tweak it until you get the ducking effect you want.

Easy and effective ducking:

With exactly the same simple little setup, you can create grooving synth parts by triggering the synth channel's gate instead of the compressor. Just click on the gate's "On" button and adjust the threshold.

Drum loop and synth pad with gate off:

Drum loop and grooving synth stutters with gate on:

Maximizing is generally added after the mixing process, when you're putting the final mastering touches on your tune. Reason's MClass Maximizer operates much like the compressors and limiters we've worked with to this point. But it's got an element of magic to it, too. It's one of those things that can just make a mix sound better. The Maximizer is automatically patched in to the master channel effects; bypass these effects as you're doing your rough mix, then add them in once you've got your EQ and track balance where you want it. Check it out.

Fig. 15. The MClass Maximizer's settings are familiar to you from your compressor work. This is the place to actually indulge in the feeling that "louder is better."

Rough mix of our bluesy horns, before maximizing:

The Maximized Blues. Sounds better already!

The "Good" Button

So far, we've talked about how to use compression to solve problems with individual tracks and groups of tracks. There's a very special part of the mixing process that applies dynamics to all your tracks: mastering. Here at Record U, we'll be providing you with tons of information on the mastering process, which involves using dynamic processing a great deal, and in very specific and groovy ways.

But even before we get into mastering, let's check out a very special compressor that's integrated in to Reason's mixer. At the top of the master channel, above even the mastering suite and insert effects, is the Master Compressor. Reason's mixer is a meticulous software model of the classic SSL analog consoles, and the Master Compressor is a shining example of this attention to detail.

The Master Compressor not only looks just like the famous mix buss compressor on SSL analog consoles, it makes everything sound good at the touch of a button, just like the original.

The Master Compressor is a faithful recreation of the mix buss compressor on the SSL mixing boards, from the look of the VU meter to the circuitry. It was designed and built like a typical mastering compressor to give an entire mix the "radio sound" as soon as it was turned on — even though it does have the standard compression controls, including sidechain keying and a "makeup" gain for increasing the volume following compression. You engaged the original by pressing on the "In" button; many people took to calling this the "Good" button because it simply made things sound better when you clicked it. You can do the same with your music by clicking on the "On" button on the Master Compressor. Try it out!

Dynamics 'til daylight

Reason's dynamic processors sound fantastic, and they're extremely flexible. On every channel of the mixer you'll find compression and gating that not only gets the job done, but sounds smooth and warm, too. The MClass effects are very effective. And we didn't even get to the Comp-01 half-rack compressor!

It's incredibly easy to try out all the dynamic processing effects Reason has to offer. Experiment often, and soon you'll have a good sense of what kind of dynamics processing works best for your music.

Use dynamics processing sparingly; even if you can't hear the compressor doing its work when you solo a track, check the track in the mix. Nothing messes up a mix faster

than dynamics processing that goes overboard. Use the "bypass" switch on the dynamic effects to make sure you're not straying too far from where your music wants to go!

Currently editor-at-large for Keyboard magazine in the U.S., Ernie Rideout writes and performs in the San Francisco Bay Area.

Post tagged as: **Dynamics**, **Mixing**, **Reason**, **Record U**

Blog

Tools for Mixing: Levels & Panning

Posted by **Mattias** in **Tutorials**

By Ernie Rideout

It feels great to finish writing a song, right? It feels even better when your band learns the song well and starts to sound good performing it. And it's even more exiting to get in a studio and record your song! What could possibly be better?

Mixing your song, of course. Nothing makes you feel like you're in control of your creative destiny as when you're in front of a mixing board — virtual or physical — putting sounds left, right, and center, and throwing faders up and down.

Yeah! That's rock 'n' roll production at its best!

Except for one or two things. Oddly enough, it turns out that those faders aren't meant to be moved all over the place. In fact, it's best to move them as little as possible; there are other ways to set the track levels, at least initially. And those pan knobs are handy for placing sounds around the sound stage, but there are other ways to get sounds to occupy their own little slice of the stereo field that are just as effective, and that should be used in conjunction with panning.

Here at Record U, we're committed to showing you the best practices to adopt to make your recorded music sound as good as it possibly can. In this series of articles, we draw upon a number of tools that you can use to make your tunes rock, including:

Faders: Use channel faders to set relative track levels.
Panning: Separate tracks by placing them in the stereo field.
EQ: Give each track its own sonic space by shaping sounds with equalization.
Reverb: Give each track its own apparent distance from the listener by adjusting reverb levels.
Dynamics: Smooth out peaks, eliminate background noises, and bring up the level of less-audible phrases with compression, limiting, gating, and other processes.

In this article, we're going to focus on two of these tools: gain staging (including setting levels with the faders) and panning. As with other tools we've discussed in this series, these have limitations, too:

1. They cannot fix poorly recorded material.
2. They cannot fix mistakes in the performance.
3. Any change you make with these tools will affect changes you've made using other tools.

It's really quite easy to get the most out of gain staging and panning, once you know how they're intended to be used. As with all songwriting, recording, and mixing tools, you're free to use them in ways they weren't intended. In fact, feel free to use them in ways that no one has imagined before! But once you know how to use them properly, you can choose when to go off the rails and when to stay in the middle of the road.

Before we delve into levels, let's back up a step and talk about how to avoid the pitfall of poorly recorded material that we at Record U keep warning you about.

Pre-Gain Show

Before your mixing board can help you to make your music sound better, your music has to sound good. That means each instrument and voice on every track must be recorded at a level that maximizes the music, minimizes noise, and avoids digital distortion.

If you're a guitarist or other tube amp-oriented musician, you're likely to use digital distortion every day, and love it. That's modeled distortion, an effect designed to emulate the sound of overdriven analog tube amplifiers — which is a sound many musicians love.

The kind of digital distortion we don't want is called clipping. Clipping occurs when a signal is overdriving a circuit, too, but in this case the circuit is in the analog-to-digital converters in your audio interface or recording device. These circuits don't make nice sounds when they're overdriven. They make really ugly noise — garbage, in fact. It can be constant, or it can last for a split-second, even just for the duration of a single sample. If that noise gets recorded into your song, you can't get rid of it by any means, other than to simply delete the clipped sections. Or re-record the track.

The way to avoid clipping is to pay close attention to the sound as it's coming in to your recording device or software, and then act to eliminate it when it occurs. There are several things that can indicate a clipping problem:

1. Your audio interface may have input meters or clipping indicators; if these go in the red, you've got a problem. Clip indicators usually stay lit up once they're been triggered, so you know even if you've overloaded your inputs for a split second.

Fig. 1. Having multi-segment meters on an audio interface is handy, like these on the MOTU Traveler; if they look like this as you're recording tracks, though, you probably have a serious clipping problem.

Fig. 2. Many audio interfaces have a simple LED to indicate clipping, as on the Line 6 Toneport KB37; here the left channel clip indicator is lit, indicating clipping has occurred. Bummer.

2. Your recording device or software probably has input meters; if the clipping indicator lights up on these, you've got a problem. In Reason, you have two separate input meters with clip indicators.
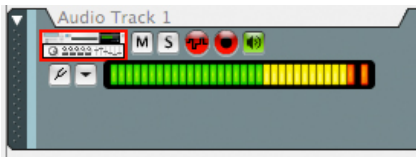
Fig. 3. In Reason, each track in the Sequencer window has an input meter. As with clip indicators on your recording hardware, these clip indicators stay lit even if you've gone over the limit for a split second — this input looks kind of pegged, though.
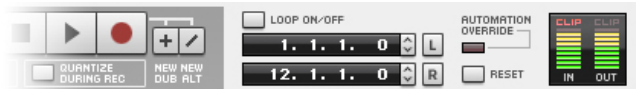
Fig. 4. The Transport Panel in Reason has a global input meter with a clip indicator as well.

3. The waveform display in your recording software's track window can indicate clipping. If your waveforms look like they've gotten a buzz haircut, you may have a problem.

Fig 5. If you see a waveform in a track that resembles the one on the left, you probably have a clipping problem. But if it's this bad, you'll probably hear it easily.

These are helpful indicators. But the best way to avoid clipping is to listen very carefully to your instruments before you record, or right after a recording a sound check. Sometimes clipping can occur even though your input meters and audio waveforms all appear to be fine, and operating within the boundaries of good audio. Other times you may see your clip indicators all lit up, but you might not be able to detect the clipping by ear; this can happen if the clipping was just for an instant. It's worth soloing the track to see if you can locate the clipping; if you don't, it may turn up as a highly audible artifact when you're farther along in your mixing process, like when you add EQ.

How do you crush clipping? If you detect clipping in a track during recording, eliminate it by doing one of the following:

1. Adjust the level of the source. Lower the volume of the amplifier, turn down the volume control of the guitar or keyboard.
2. If the tone you're after requires a loud performance, then lower the levels of the input gain knobs on your audio interface until you're getting signal mixer that is not clipping.
3. Use the pad or attenuator function on your audio interface. Pads typically lower the signal by -10, -20, or -30dB, which might be enough to let you get your excellent loud tone but avoid overloading the inputs. Usually the pad is a switch or a button on the front or back panel of the audio interface.
4. Sometimes the overloading is at the microphone itself. In this case, if the mic has a pad, engage it by pushing the button or flipping the switch. This will usually get you a -10dB reduction in signal.
5. Sometimes the distortion comes from a buildup of lower frequencies that you may not need to make that particular instrument sound good. In this case, you can move the mic farther away from the source, which will bring down the level. If the mic has a highpass filter, engage it, and that will have a similar effect.

The reverse problem is just as bad: an audio signal that's too quiet. The problem with a track that's too soft is that the ratio between the loudest part of the music and the background noise of the room and circuitry of the gear isn't very high. Later, when you're running the track through the mixer, every stage that amplifies the track will also amplify the noise.

The fix for this is simpler: Move the mic closer to the source, turn the source up, turn off the pad and highpass filters on the mic, or turn up the gain controls on your audio interface.

The goal is to make the loud parts of each track as loud and clean as possible as you record, while avoiding clipping by any means. That doesn't mean the music has to be loud; it just means that the loudest part of each track should get into the yellow part of the input meters.

Gain Staging

Now that you've spent all that time making sure each track of your song is recorded properly, you'd think the next thing we'd tell you is to start adjusting the relative levels of your tracks by moving those gorgeous, big faders. They're so important-looking, they practically scream, "Go on, push me up. Farther!"

Don't touch those faders. Not yet. You heard me.

At this point in the mixing process, those big, beautiful faders are the last things you need. What you need is far more important: the gain knob or trim control. And it's usually hidden nearly out of sight. It certainly is on many physical mixers, and it is on the mixer in Reason as well. Where the heck is it?

When setting your gain stages, use this knob to set track levels . . . (scroll down)

. . . not this. Using the faders comes later.

Fig. 6. The gain knob or trim control is often way at the top of each channel on your mixer. In Reason, it's waaaaaaaay up there. Keep scrolling, you'll find it.

This little dial is usually way the up at the top of the channel strip. Why is way the heck up there, if it's so important?

It has to do with signal flow, for one thing. When you're playing back your recorded tracks, the gain knob is the first stage the audio passes through, on its way through the dynamics, EQ, pan, insert effects, and channel fader stages.

You use the gain control to set the levels of your tracks, prior to adding dynamics EQ, panning, or anything else. In fact, when setting up a mix, your first goal is to get a good static mix, using only the gain controls. A static mix is called that because the levels of all the track signals are static; they're not being manipulated as the song plays, at least not yet.

Those beautiful big channel faders? They should all be lined up on 0, or unity gain. All tidy and shipshape.

Instead of using the faders, use the gain controls to make the level of each track sound relatively equal and appear in the channel meter as though it's between -5 dB and -10 dB below 0. Using your ears as well as the meters, decrease or increase the gain of each track until most of its material is hitting around -7 dB. You can do this by soloing tracks, listening to tracks in pairs or other groups, or making adjustments while you listen to all the tracks simultaneously.

The gain control target is -7 dB for a couple reasons. Most important, as you add EQ, dynamics, and insert effects to each track, the gain is likely to increase, or at least has the potential to increase. Starting at -7 dB gives each track room to grow as you mix. Even if you don't add any processing that increases the gain, the tracks all combine to increase the gain at the main outputs, and starting at this level helps you avoid overloading at the outputs later.

Why shouldn't you move the faders yet? After all, they sure look like they're designed to set levels!

Hold on! The faders come in later in the mixing process, and we want them to all start at 0 for a couple reasons. The scale that faders use to increase or decrease gain is logarithmic; you need larger increases in dB at higher volume levels to achieve apparent increases that equal the apparent increases made at lower levels. In other words, if your fader is down low, it's difficult to make useful adjustments to

the gain of a track, since the resolution at that end of the scale is low. If the fader is at 0, you can make small adjustments and get just the amount of change you need to dial in your mix levels.

The other reason is headroom. You always want to have room above the loudest parts of your music, in case there are loud transients and in case an adjustment made to EQ, dynamics, or effects pushes the track gain up. Plus, moving a fader up all the way can increase the noise of a track as much as the music; using EQ and dynamics on a noisy track can help maximize the music while minimizing the noise; the fader can stay at 0 until it's needed later.

Once you have each track simmering along at -7 dB, you're ready to move on to the other tools available for your mix: EQ, dynamics, effects, and panning. As you make changes using any of these tools, you may want to revise your static mix levels. And you should; just keep using the Gain control rather than the faders, until it's time to begin crafting your final mix.

It's more than a phase

As you're checking the level of each track, you may find the little button next to the gain control useful: It's the invert phase control. In Reason, this button says "INV," and by engaging it, you reverse the phase of the signal in that channel. It's good that this button is located right next to the gain knob, because it's during these first step that you might discover a couple tracks that sound too quiet, softer than you remember the instrument being. Before you crank the gain knob up for those tracks, engage the INV, inverting the phase, and see if the track springs back to life.

Fig. 7. It's small, but it comes in handy! The invert phase control can solve odd track volume problems caused by out-of-phase recording.

If so, it's because that particular sound was captured by two or more mics, and because of the mic location, they picked up the waveform at different points in its waveform cycle. When played back together, these tracks cancel each other out, partially or sometimes entirely, since they play back the "out of phase" waveforms. The INV button is there to make it easy to flip the phase of one of the tracks so that the waveforms are back in phase, and the tracks sound full again.

Perspective

All of the tools available to you for mixing your song are there to help you craft each track so that it serves the purpose you want it to for your song. For some tracks, this means creating a space just for that particular sound, so it can be heard clearly, without the interference of other sounds. For other tracks, that might mean making the sound blend in with others, to create a larger, aggregate sound.

Just as with EQ, dynamics, and effects, panning is one of the tools to help you achieve this. And just as when you use EQ, dynamics, or effects, any change you make in panning to a track can have a huge effect on the settings of the other tools.

Unlike the other tools, though, panning has just one control, and it only does one thing. How hard could it be to use?

As it happens, there are things that are good to know about how panning works, and there are some things that are good to avoid.

The word "panning" comes from panorama, and it means setting each sound in its place within the panorama of sound. It's an apt metaphor, because it evokes a wide camera angle in a Western movie, or a theater stage. You get to be the director, telling the talent — in this case, the tracks — where to stand in the panorama; left, right, or center, and downstage or upstage.

Panning handles the left, right, and center directions. How can two-track music that comes out of stereo speakers have anything located in the center? It's an interesting phenomenon: When the two stereo channels share audio information, our brains perceive a phantom center speaker between the two real speakers that transmits that sound. Sometimes it can seem to move slightly, even when you haven't changed the panning of any sounds that are panned to the center. But usually it's a strong perception, and it's the basis of your first important decisions about where to place sounds in the stereo soundscape.

Once you've established the sounds you want to be in the center, you'll walk straight into one of the

great debates among producers and mixing engineers, which is where to place sounds to the left and right. The entire controversy can be summed up in this illustration courtesy of the amazing Rick Eberly:

Fig. 8. This is controversial? Whether the drums face forward or back? You're kidding, right?

Couldn't be more serious.

Well, more to the point, the debate is about whether you want your listeners to have the perspective of audience members or of performers. This decision is kind of summed up in the decision of where to place the hi-hat, cymbals, and toms in your stereo soundscape. Hi-hat on the left, you're thinking like a drummer. Hi-hat on the right, you're going for the sound someone in the first row would hear.

You really don't need to worry about running afoul of the entire community of mixing engineers. You can do whatever you want to make your music sound best to you. But it's good to keep in mind the concept of listener perspective; being aware of where you want your audience to be (front row, back row, behind the stage, on the stage, inside the guitar cabinet, etc.) can help you craft the most effective mix.

Balance

Just as important as perspective is the related concept of balance. In fact, many mixing engineers and producers refer to the process of placing sounds in the stereo soundscape as "balancing," rather than "panning." Of course, they include level setting in this, too. But for now, let's isolate the idea of "balance" and apply it to the placing of sounds in the stereo soundfield.

Here's the idea. In a stereo mix of a song or melodic composition, the low frequency sounds serve as the foundation in the center, along with the main instrument or voice. On either side of this center foundation, place additional drums, percussion, chordal accompaniment, countermelodies, backing vocals, strumming guitars, or synthesizers. Each sound that gets placed to one side should be balanced by another sound of a similar function panned to a similar location in the opposite side of the stereo field.

Here's one way this could look.
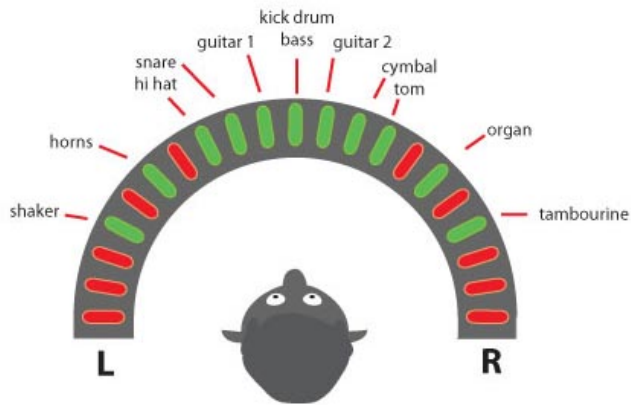
Fig. 9.

There are many ways to diagram sounds in a mix. This simple method mimics the pan pot on the mixer channels in Reason. At the center: you (sorry about the nose). In front of you are the foundation sounds set to the center. To either side of you are the accompanying sounds that have been placed to balance each other. In this song, we've got drums, bass, two electric guitars playing a harmonized melody, organ, horns, and a couple of percussion instruments. Placing sounds that function in a similar way across the stereo field equally (snare and hi-hat — cymbal and tom; horns — organ; shaker — tambourine) make this mix sound balanced from left to right; when we get to setting levels, we might choose to reinforce this by matching levels between pairs.

And now you know where we stand on the perspective debate . . . at least for this clip.
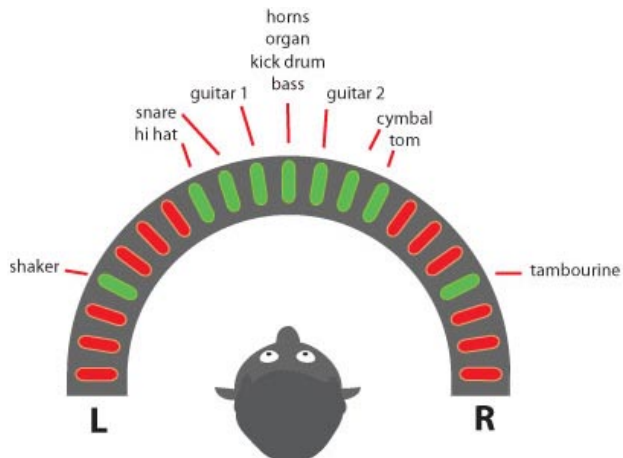
Fig. 10.

Here's another approach to a mix. We've put the horns and organ in the center. This is still balanced, but this approach may not give us one critical thing we need from everything we do in a mix: a clear sonic space for each instrument. We'll hear how this sounds in a bit.
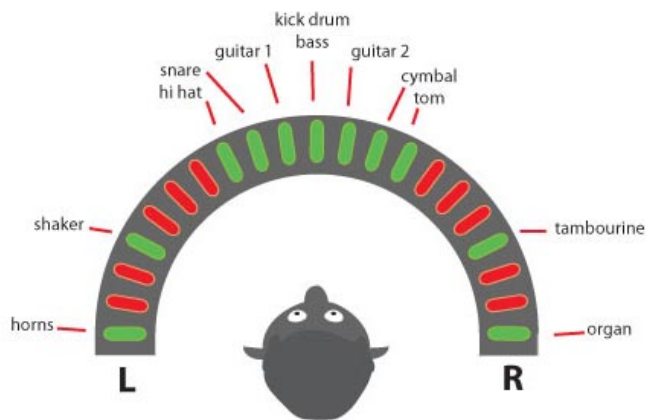
Fig. 11.

Here's yet another balanced approach, this time putting the horns as far to the left and right as possible. Though valid, this also presents certain problems, as we'll hear shortly.
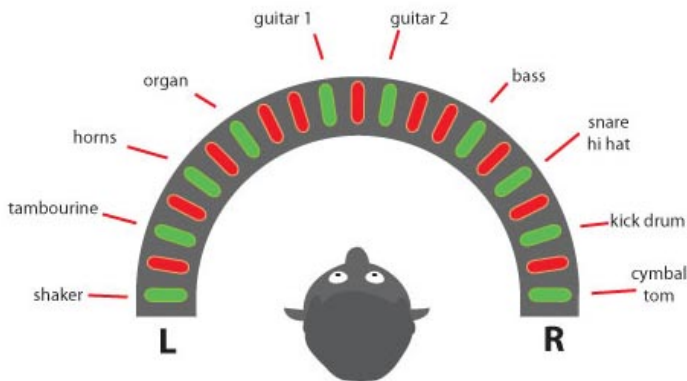
Fig. 12.

This diagram represents a mix that looks balanced, but when you listen to it, you'll hear that it's not balanced at all. The foundation instruments are not centered, for one thing, and this has a tremendous impact. For most studio recordings, this approach might be disconcerting to the listener. But if your instrumentation resembles a chamber ensemble or acoustic jazz group and you're trying to evoke a particular relationship between the instruments, this could be just the approach your composition needs. We'll see how it works out in the context of a rock tune a little later.

Set up a static mix

Let's go through the process of setting up a static mix of a song, using the steps and techniques we've talked about to set the gain staging, the levels, and the balance. As it happens, the song we'll work on is just like the hypothetical song on which we tried several different panning scenarios.

A couple of interesting things to know about this song. All the instruments come from the ID8, a sound module device in Reason, except for the lead guitars, which are from an M-Audio ProSessions sample library called Pop/Rock Guitar Toolbox. The drum pattern is from the Record Rhythm Supply Expansion, which is available at no charge in the Downloads section of the Propellerhead website —click here to get it.

The Rhythm Supply Expansion contains Reason files each with a great selection of drum patterns and variation in a variety of styles, at a range of tempos. The really cool thing about the Expansion files is that they're not just MIDI drum patterns, they include the ID8 device patches, too — just select "Copy Device and Track" from the Edit menu when in Sequencer view, then paste the track and ID8 into your song file. With the Rhythm Supply Expansion tracks, your ID8 becomes a very handy songwriting and demoing tool.

All right. We've completed our tracking session, and we're happy with the performances. We're satisfied that we have no clipping on any of the tracks, since we had no visual evidence of clipping during the tracking (e.g., clip LEDs or pegged meters on the audio interface, clipped waveforms in the Sequencer

view) and we heard no evidence of clips or digital distortion when we listened carefully. Now let's look at our mixer and see what we need to do to set the gain staging and levels.

 Fig. 13. First things first, though: Bypass the default insert effects in the master channel to make sure you're hearing the tracks as they really are (click on the Bypass button in the Master Inserts section).
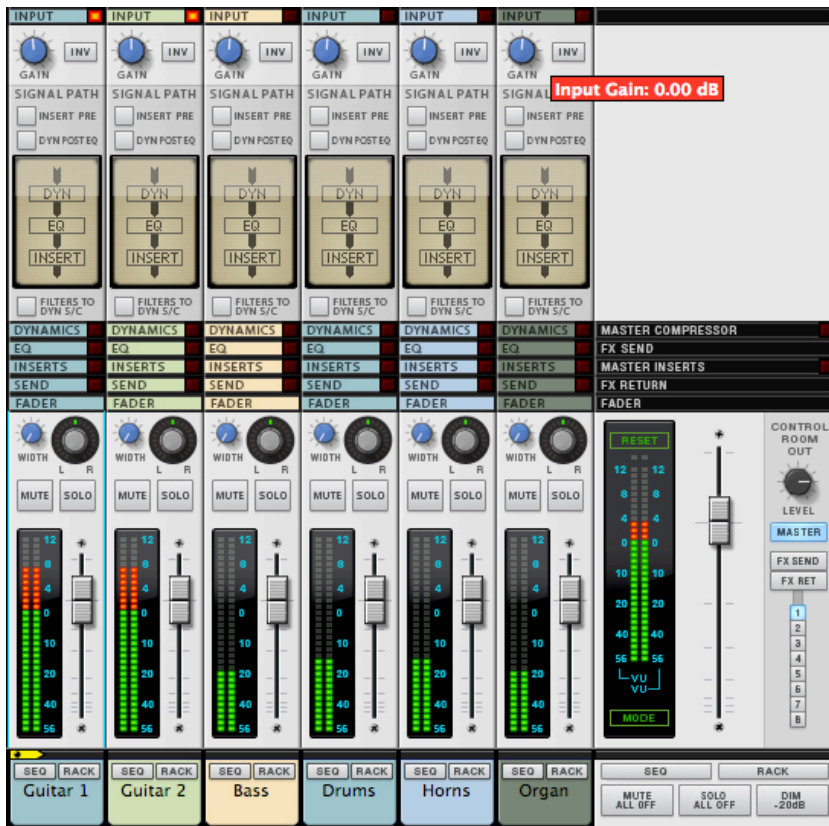
Fig 14. While the levels of each track seem to be in the ballpark, it's clear that there is some disparity between the guitars (too loud, what a surprise) and the rest of the instruments. Quick! Which do you reach for, the faders or the gain knobs? Let's collapse the mixer view by deselecting the dynamics, EQ, inserts, and effects sections of the mixer in the channel strip navigator on the far right of the mixer. Now we can see the Gain knobs and the faders at the same time. Still confused about which to reach for to begin adjusting these levels? Hint: Leave the faders at 0 until the final moments of your mixing session!

Okay, that was a little more than a hint. Have a listen to the tracks at their initial levels:

00:00   00:00

Fig. 15. Using only the Gain knobs, we've adjusted the level of each track so that, a) we can hear each instrument equally, and b) the levels of each track centers around -7 dB on its corresponding meter. Even though we brought up the levels of the non-guitar tracks, the overall master level has come down, which is good because it gives us more headroom to work with as we add EQ, dynamics, and effects later. Ooh, and look at how cool those faders look, all centered at 0! Let's hear the difference now that the levels are all in the ballpark:

00:00   00:00

Since a bit part of this process is determining exactly where each drum and percussion sound is to go, let's take that Rhythm Supply Expansion stereo drum track and explode it so that each instrument has its own track. This is easy to do: Select the drum track in Sequencer view, open the Tool window (Window > Show Tool Window), click on Extract Notes to Lanes, select Explode, and click Move. Presto! All your drum instruments are now on their own lanes (watch out, your hi-hat has been separated into two lanes, one containing the closed sound, and the other containing the open sound, just something to keep in mind or combine them into a single track). Copy each lane individually, and paste them into their own sequencer tracks. Now each drum instrument has its own track, and you can pan each sound exactly as you want.

Let's listen to the process of balancing. We'll build the foundation of our mix first, starting with the drums, then adding the bass, then the lead instruments, which are the guitars. Let's mute all tracks except the drums and then pan the drum tracks to the center:

00:00   00:00

Sounds like a lot of drums crammed into a small space. There is a shaker part that you can't even hear, because it's in the same rhythm as the hi-hat. Let's pan them, as you see in Fig. 9. Hold on tight, we're taking the performer's perspective, rather than the audience's:

00:00   00:00

That opens up the sound a great deal. You can hear the shaker part and the hi-hat clearly, since they're now separated. Even a very small amount of stereo separation like this can make a huge difference in the audibility of each instrument. Now let's add the bass, panned right up the center, since it's one of our foundation sounds:

00:00   00:00

The bass and kick drum have a good tight sound. Now let's un-mute the two lead guitar tracks. We'll pan these a little to the left and right to give them some separation, but still have them sound clearly in the center:

00:00   00:00

So far, we've got a nice foundation working in the center. All the parts are clearly audible. Sure, there's a lot of work we could do even at this stage with EQ, dynamics, and reverb to make it sound even better. Let's resist that urge and take the time to get all the tracks balanced first. Now we'll un-mute the organ and horn tracks. These two instruments play an intertwining countermelody to the lead guitars. They're kind of important, so let's see what they sound like panned to the center, as in Fig. 10:

00:00   00:00

Wow. There is a lot going on in this mix. The parts seem to compete with each other to the point where you might think the horn and organ parts are too much for the arrangement. Let's try panning the horns and organ hard left and hard right — in other words, all the way to the left and right, respectively:

00:00   00:00

Well, we can hear the horns and organ clearly, along with all the other parts. So that's good. But they sound like they're not really part of the band; it sounds unnatural to have them panned so far away. Let's try panning them as you see in Fig. 9:

00:00   00:00
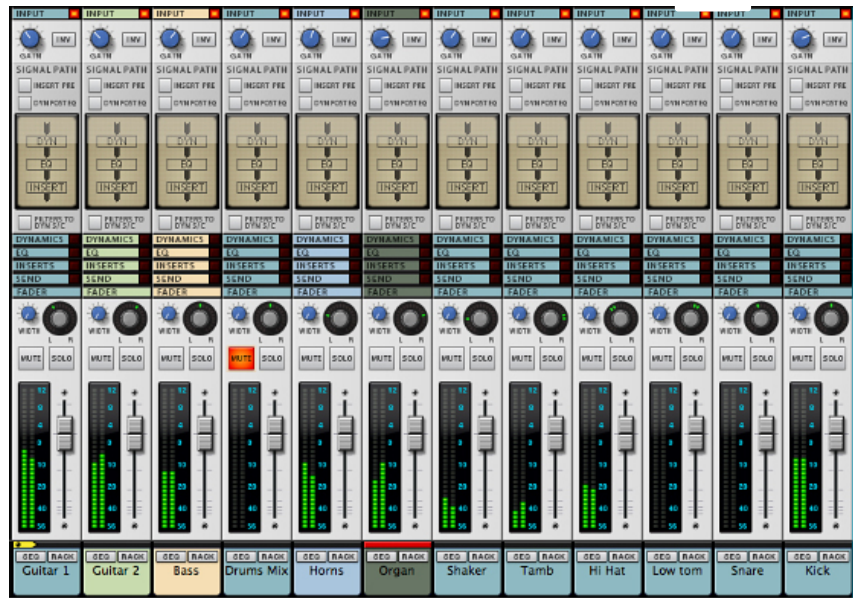
Fig. 16. This screenshot shows our static mix, with all track levels adjusted, all sounds balanced, and all faders still at 0. Now we're ready to clarify and blend the individual sounds further with EQ, dynamics, reverb, and level adjustments, which you can read about in the other articles here at Record U.

Wait! What about the balancing example that was out of balance, in Fig. 12? How does that sound? Check it out:

00:00   00:00

The big problem is that the foundation has been moved. In fact, it's crumbled. The sound of this mix might resemble what you'd think a live performance would sound like if the performers were arranged across the stage just like they're panned in this example. But in reality, the sound of that live performance would be better balanced than this example, since the room blends the sounds, and the concert goers would perceive a more balanced mix than you get when you try to emulate a stage setup with stereo balancing.

That's not to say you shouldn't take this approach to balancing if you feel your music calls for it. Just keep in mind what you consider to be the foundation of the composition, and make sure to build your mix from those sounds up.

And don't touch those faders yet!

Based in the San Francisco Bay Area, Ernie Rideout is Editor at Large for Keyboard magazine, and is writing Propellerhead Record Power! for Cengage Learning.

Post tagged as: **Mixing**, **Reason**, **Record U**

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Recording Vocals and Selecting a Microphone

By Gary Bromham

Performing a lead vocal is arguably the toughest job in the recording studio. This in turn puts more emphasis on the importance of capturing and recording the vocal performance as perfectly as possible. Vocalists often tire easily and generally their early takes tend to be the best (before the thinking and over-analyzing takes over!)

Usually, and in a very short space of time, an engineer has to decide which mic, signal path (preamp, compressor eq etc) to use, set the correct level for recording and headphone balance, create the right atmosphere for singing and generally be subjected to, at best, minor grunts, at worst verbal abuse until the penny drops! Vocalists are a sensitive bunch and need nurturing, cuddling and whatever else it takes to make them feel like a supertar!

During this article I shall attempt to set out a strategy for accomplishing these goals and maybe throw in a tip or two I've picked up along the way to assist in capturing the perfect take.

Selecting a Microphone

Microphones come in all shapes and sizes but a basic understanding of how they work will help in any assessment of which one we choose.

All microphones work in a similar way. They have a diaphragm (or ribbon), which responds to changes in air pressure. This movement or vibration is in turn converted into an electrical signal, which is amplified to produce a sound. This is very simplistic but essentially the basic science behind making a sound with a mic.

There are three main types of microphone to choose from.- Dynamic, Ribbon and Condenser.

Dynamic microphones are generally used for more close miking purposes such as drums or guitar cabinets, their sound is usually more mid-range focused and they can cope with higher sound pressure levels (SPL's)

Condenser, or Capacitor mics, as they are also known, are more sensitive to sound pres-

sure changes. They also tend to have a greater frequency response or dynamic range than dynamic mics. For this reason they tend to be the de-facto choice for vocals. Condenser microphones require a power source, called phantom power, to function. This is needed to power the built in preamplifier and also to polarize (power) the capsule. However this may not always be the choice. Bono from U2 for example likes to use a Shure SM-58 dynamic mic as it allows him more freedom to move around or perform the vocal as if in a live environment! A condenser mic, due to its sensitivity, might prohibit being held in the hand due to noises or indeed it may have too greater frequency range!

Ribbon mics are the curved ball here, so to speak, as they are often richer in tone to both our alternatives. They are softer or more subtle; they tend not to have the hyped top-end of condenser mics and unlike dynamic mics are very sensitive to SPL changes. For this reason they have to be treated with care as the ribbon will not tolerate excessive movement from either loud sound sources or being thrown around! They also generally have a much lower output level to the other two and subsequently need more gain from a pre-amp.

Here's a short snippet of a vocal I'm currently working on recorded with a Shure SM58:

The same recorded with a Neumann U-87:

And with a Coles 4038 ribbon mic:

When comparing models there are a number of important specifications we need to consider.

Frequency Response

When we look at the frequency response of the vocal mic we select will it sound flat or natural or will it boost certain frequencies? It is often preferable, particularly with vocals, for a microphone to enhance or accentuate certain frequencies, which suit a particular singer. Check out www.microphone-data.-com for a detailed look at different mics specs. I love this site and spend hours trawling

through the pages…does this mean I'm sad and need to get a life?

Sound Pressure Level (SPL)

How much dynamic range or level can the microphone cope with? This is the difference between the maximum sound pressure level and the noise floor or in basic terms the range of usable volume without distortion at high level or noise at low level. Dynamic mics are generally much better at dealing with loud source material than the condenser or ribbon variety.

Noise Floor or Noise level

How loud is the background noise created by the microphone itself? Obviously for someone who sings rock music this is less of an issue than some body who sings ambient jazz. As a rule of thumb Capacitor mics are more adept at capturing subtleties and nuances than dynamic mics.

Sensitivity

Scientifically this is a measure of how efficiently a microphone converts sound pressure changes to control voltage or electrical signals. This basically is how loud the microphone is capable of being. You remember I mentioned earlier that Ribbon mic require a preamplifier with lots of gain to get the correct level to feed the mixer or recorder-in our case 'Reason'.

Polar Patterns

Don't worry…nothing to do with global warming!

Our final consideration in choosing a microphone is the pickup pattern or as it is more commonly known the Polar Pattern. On a circular graph this is a representation of which direction a mic picks up sound. The diagram illustrates how this works.

Fig 2. The diagram shows three basic polar patterns. All other patterns are variations of these. The blue circle is an Omni pattern, the red circles show a figure of eight and the green line shows
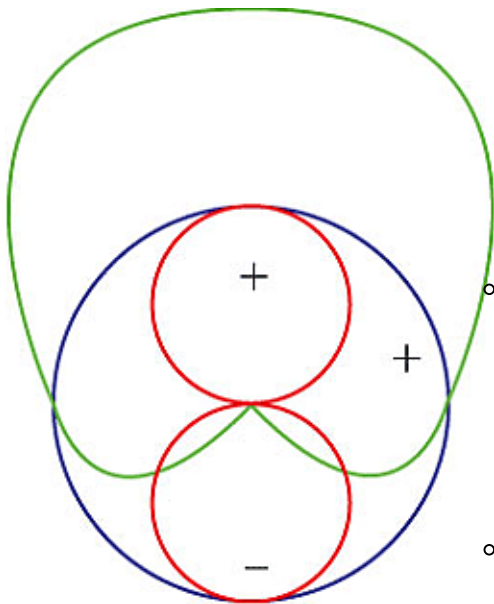
the cardiod.

There are essentially three basic patterns for us to consider when understanding where the mic will pickup sound:

○ Omni-directional.

As its name suggests, the microphone will pickup sound equally from all directions. Useful if you want to record all the ambience or space around the source.

○ Cardiod

Otherwise known as (and as its name suggests) 'heart-shaped' Picks up the source mainly from the front, while rejecting most sound from the sides and rear. The advantage here is that the microphone captures only the source that it is pointing at. Hyper-cardiod is similar and often cast in the same category. These simply have a narrower field of pick up than the normal cardiod and are very well suited for singers where more isolation is required or else where feedback is a problem.

○ Bi-Directional

Also known as 'figure of eight'. Here sound is picked up equally from the front and rear, whilst signal from the sides is rejected.

Generally speaking there is no rule as to which type of microphone or which pattern we should use when recording vocals although most engineers tend to veer towards a condenser mic and use a cardiod pattern. There is good argument to suggest that an omni pattern is the ultimate setting but this poses a further question relating to the recording environment, which I shall touch on, later in this article.

In summary our checklist when choosing a suitable microphone should look something like this:

1. Consider the frequency response. Is it flat and will it therefore produce a more natural result or does it boost particular frequencies and thus enhance our vocal sound?
2. Check the polar pattern. Does it have the pick up pattern we require?

3. Check the sensitivity. How much gain will we need on our preamp to get the required level for recording?
4. Check the dynamic range and the noise level. Can the mic handle the softest and loudest levels for capturing the vocal performance?

Practical tips and further considerations

Shock treatment

It is generally beneficial to use a shock-mount when recording vocals. This prevents low frequency sounds from getting in to the microphone.

Pop treatment

Fig 3. Hi-pass filter in the Reason eq section, set to take out any unwanted noises below 80Hz.

It is also a good idea to use a pop shield. Often when a singer stands too close to the microphone sudden puffs of air such as 'p' and 'f' sounds produce unwanted noises. Pop filters can be bought ready-made to prevent this but the more resourceful amongst us have sometimes resorted to the DIY approach and stolen a pair of our wife's (or husbands!!!) stockings and stretched them over a wire coat hanger to achieve a similar result.

To help with both of the above problems we might also try a hi-pass filter. If the microphone does not have a dedicated switch then we could use the filter in the eq section such as the one in the channel strip in Reason.

The Proximity Effect

Nothing to do with over-use of garlic in cooking and need for breath fresheners!

As we get nearer or further away from a microphone the bass frequencies increase or decrease accordingly. Typically a cardiod microphone will boost or cut frequencies around 100Hz by as much as 10-15dB if we move from 25cm to 5cm. and back again. This phenomenon is known as the 'Proximity Effect'. We can use this to our advantage if the singers mic technique is good, producing a richer, deeper and often more power-

ful sound that pop or rock singers really like! Radio DJ's or announcers have been using this technique for years, particularly on the late night luuuurrrrv show!

Unfortunately the use of this effect requires the vocalist to maintain a fairly consistent distance from the mic and for this reason it is often more desirable to select a mic which has less of this proximity effect. As a generalisation condenser microphones are better at this than dynamic mics. Here are a few audio examples demonstrating this principle.

Vocal recorded using a Neumann U-87 recorded at a distance of 3cm:

At a distance of 18cm:

Finally at 60cm:

The Tube Effect

A 'tube' or 'valve' microphone uses a valve as the preamplifier for gain rather than a conventional solid-state (usually FET) circuit. Most early condenser microphones such as the Neumann U-47 or the AKG C-12 employed this circuitry, at least until transistors were invented. The tonal characteristic is often warmer and more pleasing to the ear, the sound is however coloured and not always suitable for every singer. In reality the tube is adding a small level of distortion and if overused can sound muddy and unfocused!

The sound of the classic AKG C-12 Valve microphone:

Recording Environment

Something people often overlook and underestimate is the effect of the room or recording environment on the sound of the vocal. Indeed you can be using the best microphone in the world and still obtain an awful sound if the acoustic space is reflective or badly treated. To an extent our ears are able to block out or ignore deficient room acoustics whereas the microphone only records what's there! An omni-directional microphone will accentuate this whereas a hyper-cardiod will, to an extent, minimise it. Unfortunately, not all of us are blessed with perfect recording environments all of the time and often have to adapt or improvise with the set of conditions we have.

Reflection filters have become very trendy these days with the advent of bedroom studios. The SE reflection filter is one I use personally at home and highly recommend.

Failing on this, duvets, carpets or anything absorbent, will help in alleviating the situation. In summary it is often not your mic that is the problem but the environment in which it is being recorded which is the real problem. Save some money, get down to Ikea, and buy a couple of new duvets before spending another $1000 on a bespoke microphone!

Trade Secrets

Headphone balance

The headphone mix, after microphone selection, is probably the most critical part of the recording process for a vocal. We can save ourselves hours, not to mention several tantrums, if the balance is good for our singer. Some vocalists also like to sing with one side of their headphones off. In my experience this can be an indication that their headphone mix isn't completely right. Singers also tend to have their 'cans' too loud because they believe they can't hear themselves when in reality they are not relaxed or are hearing themselves incorrectly. When pitching is flat this can also be a tell tale sign of headphones being too loud, the reverse being true when the singer is performing sharp! (' I should point out at this stage that 'cans' is on the list of banned word in certain studios!)

It is often a good idea to setup a separate headphone mix or, cue mix, as it is sometimes called for the singer. This is useful for a number of reasons. If we want to tweak the vocal whilst the singer is singing, but without them hearing us do so, we need to set up a separate balance from the one we are listening to in the control room. For this we simply use one of the sends in the mixer in Reason. We route the output from one of the sends in the master section to the hardware interface, which in turn feeds our headphones via dedicated outputs on our soundcard. Provided we have the Pre button engaged, to the left of the Send knob, when we solo a channel in the mixer the vocalist will still hear only their individual cue mix.

Fig 4. Shows Send 8 being used as a cue send for the vocalists headphone mix. Note the send is sent Pre-fade so that whatever changes we make to the channel (level, solo mute etc) do not affect what is heard in the cans. Also note that in the Master section we can monitor the FX send via the control room out.

Fig 5. Here we see the back of the Master section where send 8 is routed out of Outputs 3-4 of our soundcard via Reasons hardware interface.

Time is of the essence

Capture that take before it's too late! Singers have a tendency to over-analyse or be over critical of their performance. Often the first thing they sing, before the inhibitions set in, are the best things they sing. My strategy is to record everything. It is after all much easier to repair a less than perfect vocal sonically, even if the compressor and pre-amp weren't set up perfectly, than it is get the vocalist to repeat that amazing performance.

Reverb

Singers often like to sing with reverb. This is in itself okay but not if it's at the expense of pitching and timing. It's harder to find the pitch of a note if all you're hearing is a wash of reflective sound. Isn't that what we spent all that time trying to eliminate when we created the recording environment? Not exactly, no, sometimes vibe is an important factor but there is a happy medium here!

Vocal chain

Selecting a preamplifier, and if necessary a compressor, is often almost as difficult as choosing the right mic. Whether you are using a Neve 1073 or an Apogee Duet as your preamplifier the principle of setting up and recording a vocal is the same. Increase the gain on the pre-amp until you start to hear a small amount of audible distortion or see slight clipping and then reduce the level by 5-10dB.

I have often heard it said that if we select a valve microphone for our vocal then the pre-

amp and compressor might be better suited to being solid-state. Too many valves in the chain can often add too much colour! Personally, in my chain, I like a valve mic with solid-state preamp followed by a valve compressor.

We only really need a compressor (or more likely a limiter) if we have a particularly dynamic vocalist. An LA-2A opto or 1176 FET style is ideal if the budget allows! Be very sparing as compression if set incorrectly cannot be undone! The M-Class compressor can be set up to behave in a subtle way, conducive to controlling level fluctuations but not squashing the sound.

Fig 6. A typical compressor set-up for recording vocals. Moderate attack and release settings ensure a relatively inaudible effect on the input when recording a vocal.

The Hardness Factor!

An interesting exercise when evaluating different microphones for different vocalists is to rate them on a scale of 1-10 on a hardness factor. A Shure SM-58 dynamic mic might get an 8 while a Rode NT2 condenser mic might get a 4. When selecting the microphone, we also give our singer a rating as well. A hard sounding voice gets a softer microphone whilst a more subtle vocal may require a harder sounding mic.

Summary

Whilst writing this article I have been conscious of not being too preachy! These are only guidelines to recording a vocal and often the great thing with recording is breaking rules. A basic understanding of how the microphone works is helpful but the single most important thing is getting the atmosphere right for the vocal to happen in the first place. Many great vocal performances have been captured with strange microphone selections and singers insisting on stripping off in the vocal booth to get the vibe! Don't question it, just remember…record everything!

Gary Bromham is a writer/producer/engineer from the London area. He has worked with many well-known artists such as Sheryl Crowe, Editors and Graham Coxon. His favorite Reason feature? "The emulation of the SSL 9000 K console in 'Reason' is simply amazing, the new bench-

mark against which all others will be judged!"

Post tagged as: **Reason**, **Record U**, **Recording**, **Vocals**

Preparing a Space for Recording (often on a budget)

By Gary Bromham

When preparing a space for recording and mixing we enter a potential minefield, as no two areas will sound the same, and therefore no one-solution-fits-all instant fix is available. There are, however, a few systematic processes we can run through to facilitate vastly improving our listening environment.

When putting together a home studio, it is very easy to spend sometimes large sums of money buying equipment, and then to neglect the most important aspect of the sound; namely the environment set up and used for recording. No matter how much we spend on computers, speakers, guitars, keyboards or amps etc., we have to give priority to the space in which they are recorded.

Whether it be a house, apartment, or just a room, the method is still based on our ability to soundproof and apply sound treatment to the area. It is extremely difficult to predict what will happen to sound waves when they leave the speakers. Every room is different and it's not just the dimensions that dictate how a room will sound. Assorted materials which make up walls, floors, ceilings, windows and doors – not to mention furniture – all have an effect on what we hear emanating from our monitors.
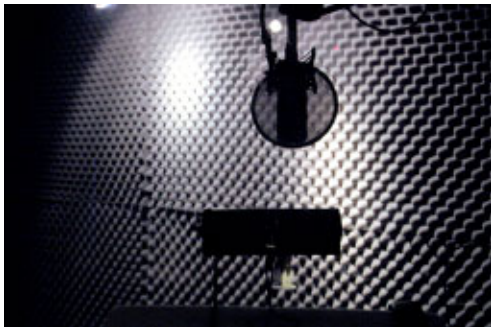
Fig 1. A vocal booth with off the shelf acoustic treatment fitted.

Whether we have a large or a small budget to sort out our space, there are a number of off-the-shelf or DIY solutions we can employ to help remedy our problem. It should be pointed out at this stage that a high-end studio and a home project studio are worlds apart. Professional studio design demands far higher specification and uses far narrower criteria as its benchmark, and therefore costs can easily run in to hundreds of thousands!

Why do we use acoustic treatment?

An untreated room – particularly if it is empty – will have inherent defects in its frequency response; this means any decisions we make will be based on the sound being

'coloured'. If you can't hear what is being recorded accurately then how can you hope to make informed decisions when it comes to mixing? Any recordings we make will inherit the qualities of the space in which they are recorded. Fine if it's Abbey Road or Ocean Way, but maybe not so good if it's your bedroom.

No matter how good the gear is, if you want your recordings or mixes to sound good elsewhere when you play them, then you need to pay attention to the acoustic properties of your studio space.

Begin with an empty room

When our shiny new equipment arrives in boxes, our instinct is always to set it up depending on whether it 'looks right', as if we are furnishing our new apartment.

Wrong!

Beware. Your main concern is not to place gear and furniture where they look most aesthetically pleasing, but where they sound best. The most important consideration is to position the one thing that takes up zero space but ironically consumes all the space. It is called the sound-field, or the position in the room where things sound best.

One of the things I have learned is that the most effective and reliable piece of test equipment is – surprise surprise – our ears! Of course we need more advanced test equipment to fine-tune the room but we need to learn to trust our ears first. They are, after all, the medium we use to communicate this dark art of recording.

Listen!

Before you shift any furniture, try this game.

Ask a friend to hold a loudspeaker, and whilst playing some music you are familiar with, use a piece of string or something which ensures he or she maintains a constant distance from you of say 2-3 metres. Get them to circle around you whilst you stand in the centre of the room listening for the place where the room best supports the 'sound-field'. The bass is usually the area where you will hear the greatest difference. As a guide listen for where the bass sounds most solid or hits you most firmly. Why am I focusing on bass? Because, if you get the bass right, the rest will usually fall into place.

Also, avoid areas where the sound is more stereo (we are after all holding up just one

speaker, a mono source); this is usually an indication of phase cancellation. Beware of areas where the sound seems to disappear.

Finally, having marked a few potential positions for speaker placement, listen for where the speaker seems to sound closest at the furthest distance. We are looking for a thick, close, bassy and mono signal. When we add the second speaker this will present us with a different dilemma but we'll talk about speakers later.

Remember: Though you may not have any control over the dimensions of your room, you do have a choice as to where you set up your equipment, and where you place your acoustic treatment. As well as the above techniques there are other things to consider.

- It is generally a good idea to set up your speakers across the narrowest wall.
- As a rule, acoustic treatment should be as symmetrical as possible in relation to your walls.
- Ideally your speakers should be set up so that the tweeters are at head height
- The consistency of the walls has a huge bearing on the sound. If they are thin partition walls then the bass will disperse far easier and be far less of a problem than if they are solid and and prevent the bottom end from getting out. (This is a Catch-22 as thin walls will almost certainly not improve relations with neighbours!)

Audio 1.'Incredible' Front Room:

Audio 2.'Incredible' Center Room:

Audio 3.'Incredible' Back Room:

Three audio examples demonstrating the different levels of room ambience present on a vocal sample played 0.5 m/2.5m/5m from the speakers in a wooden floored room.

The Live Room

If you are lucky enough to have plenty of space and are able to use a distinct live area the rules we need to observe when treating a listening area don't necessarily apply here. Drums, for example, often benefit from lots of room ambience, particularly if bare wood or stone make up the raw materials of the room. I've also had great results recording guitars in my toilet, so natural space can often be used to create a very individual sound. Indeed, I've often heard incredible drum sounds from rooms you wouldn't think

fit to record in.

Fig 2. Reflexion Filter made by SE Electronics.

'Dead Space'

It is often a good idea to designate a small area for recording vocals or instruments which require relative dead space. It would be unnatural (not to mention almost impossible) to make this an anechoic chamber, devoid of any reflections, but at the same time the area needs to be controllable when we record. Most of us don't have the luxury of having a separate room for this and have to resort to other means of isolating the sound source like the excellent Reflexion Filter made by SE Electronics. This uses a slightly novel concept in that it seeks to prevent the sound getting out into the room and subsequently cause a problem with reflections in the first place. Failing this, a duvet fixed to a wall is often a good stopgap and the favourite of many a musician on a tight budget.

Time for Reflection

Every room has a natural ambience or reverb, and it should be pointed out at this stage that it is not our aim to destroy or take away all of this. If the control room is made too dry then there is a good chance that your mixes will have too much reverb, the opposite being true if the room is too reverberant.

The purpose of acoustic treatment is to create an even reflection time across all – or as many as possible – frequencies. It obviously helps if the natural decay time of this so called reverb isn't too excessive in the first place.

Higher frequency reflections, particularly from hard surfaces, need to be addressed as they tend to distort the stereo image, while lower frequency echoes, usually caused by standing waves, often accent certain bass notes or make others seem to disappear. High frequency "flutter echoes", as they are known, can often be lessened by damping the ar-

eas between parallel walls. A square room is the hardest to treat for this reason, which is why you generally see lots of angles, panels and edges in control room designs. Parallel walls accentuate any problems due to the sound waves bouncing backwards and forwards in a uniform pattern.

Standing waves
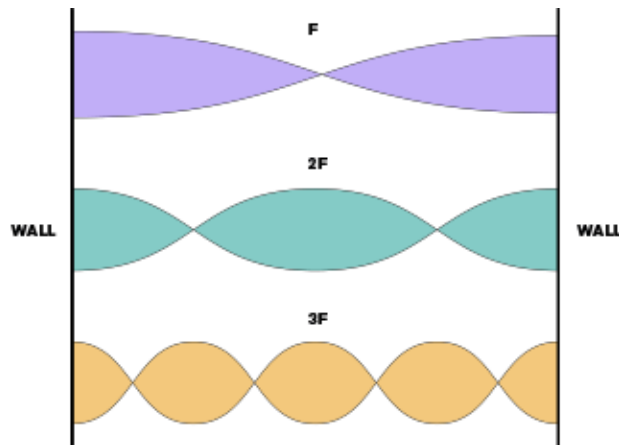
Fig 3. A graph showing different standing waves in a room

Standing, or stationery, waves occur when sound-waves remain in a constant position. They arise when the length of this wave is a multiple of your room dimension. You will hear an increase in volume of sounds where wavelengths match room dimensions and a decrease where they are half, quarter or eighth etc. They tend to affect low end or bass (because of the magnitude of the wavelength). For this reason they are the hardest problem to sort out, and because of the amount of absorption and diffusion needed generally the costliest to sort out. Further explanation is required.

Suppose that the distance between two parallel walls is 4 m. Half the wavelength (2m) of a note of 42.5 Hz (coincidentally around the pitch of the lowest note of a standard bass guitar-an open 'E') will fit exactly between these surfaces. As it reflects back and forth, the high and low pressure between the surfaces will stay constant – high pressure near the surfaces, low pressure halfway between. The room will therefore resonate at this frequency and any note of this frequency will be emphasized.

Smaller rooms sound worse because the frequencies where standing waves are strong are well into the sensitive range of our hearing. Standing waves don't just happen between pairs of parallel surfaces. If you imagine a ball bouncing off all four sides of a

pool table and coming back to where it started; a standing wave can easily follow this pattern in a room, or even bounce of all four walls, ceiling and floor too. Wherever there is a standing wave, there might also be a 'flutter echo'.

Next time you find yourself standing between two hard parallel surfaces, clap your hands and listen to the amazing flutter echo where all frequencies bounce repeatedly back and forth. It's not helpful either for speech or music.

Audio 4. Subtractor in Reason:

Here's an ascending sequence created in Reason using Subtractor set to a basic sinewave. Whilst in the listening position play back at normal listening level. In a good room the levels will be even but if some notes are more pronounced or seem to dissapear this usually indicates a problem at certain frequencies in your room.
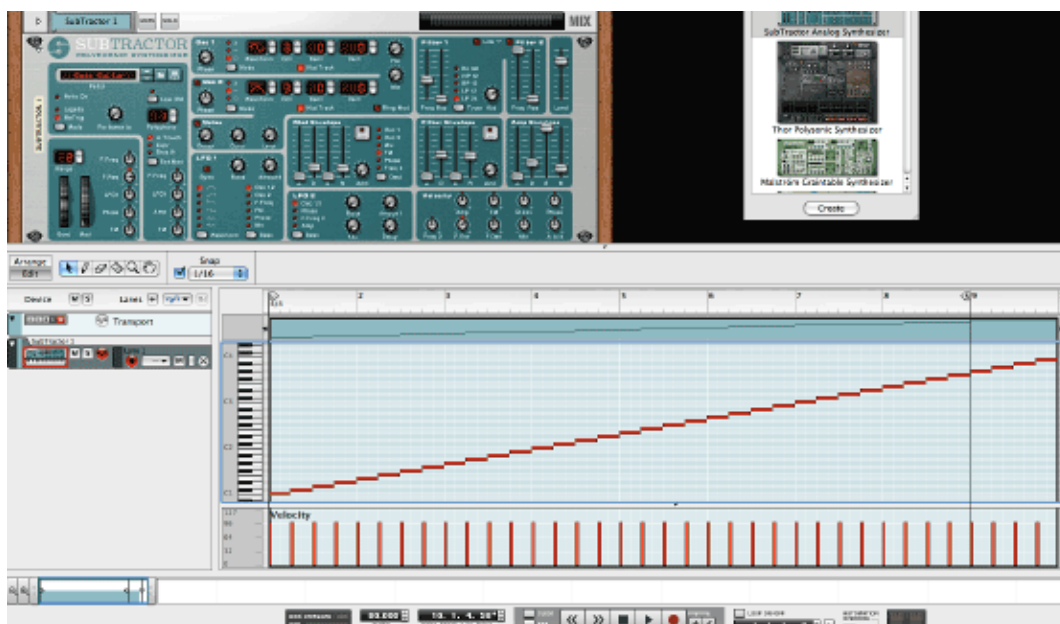
Fig 4. A chromatic sequence using Subtractor created in Reason.
Download this as a Reason file and convert the notes in the file to frequency and wave length.

Absorption or Diffusion…that is the question?

The two main approaches when sorting out sound problems are finding the correct balance between absorption and diffusion. While absorbers, as their name suggests, absorb part of the sound; diffusers scatter the sound and prevent uniform reflections bouncing back into the room.

Absorbers tend to be made of materials such as foam or rockwool. Their purpose is to soak up sound energy. Foam panels placed either side of the listening position help with mid or high frequencies or traps positioned in corners help to contain the unwanted dispersion of bass.

Diffusers are more commonly made of wood, plastic or polystyrene. By definition they are any structure which has an irregular surface capable of scattering reflections. Diffusers also tend to work better in larger spaces and are less effective than absorbers in small rooms.

Off-the-Shelf solutions

Companies such as Real Traps, Auralex and Primacoustic offer one-stop solutions to sorting out acoustic problems. Some even provide the means for you to type in your room dimensions and then they come back with a suggested treatment package including the best places to put it. These days I think these offer excellent solutions and are comparatively cheap when you look at the solutions they offer. What they won't give you is the sound of a high end studio where huge amounts of measurement and precise room tuning is required but leaving science outside the door they are perfect for most project studios.

DIY

The DIY approach can be viewed from two levels. The first, a stopgap, where we might just improvise and see what happens. The second, a more methodical, 'let's build our own acoustic treatment because we can't afford to buy bespoke off the shelf tiles and panels' approach.

- This could simply be a case of positioning a sofa at the back of the room to act as a bass trap. Putting up shelves full of books which function admirably as diffusers. Hanging duvets from walls or placing them in corners for use as damping. I even know of one producer who used a parachute above the mixing desk to temporarily contain the sound!
- Build your own acoustic treatment. I personally wouldn't favour this as it is very time consuming and also presumes a certain level of ablity in the amateur builder department. The relative cheapness of 'one solution' kits where all the hard work is done for you also makes me question this approach. However, there are numerous

online guides for building your own acoustic panels and bass traps which can save you money.

Speakers

Though speakers aren't directly responsible for acoustic treatment their placement within an acoustic environment is essential. I've already suggested how we might find the optimum location in a room for the speakers; the next critical thing is to determine the distance between them. If they are placed too close together the sounds panned to the centre will appear far louder than they actually are. If they are spaced too far apart then you will instinctively turn things panned in the middle up too loud. The sound is often thin and without real definition.

Finally, speaker stands or at least a means of isolating the speaker from the surface on which it rests is always a good idea. The object is to prevent the speaker from excessive movement and solidify the bass end. MoPads or China Cones also produce great results

Headphones

The role of headphones in any home studio becomes an important one if you are unsure of whether to trust the room sound and the monitors. This, in essence, removes acoustics from the equation. Though I would never dream of using them as a replacement for loudspeakers, they are useful for giving us a second opinion. Pan placement can often more easily be heard along with reverb and delay effects

Summary

With only a small amount of cash and a little knowledge it is relatively easy to make vast improvements to the acoustics of a project studio. A science-free DIY approach can work surprisingly well, particularly if you use some of the practical advice available on the websites of the companies offering treatment solutions. Unfortunately, most musicians tend to neglect acoustic treatment and instead spend their money on new instruments or recording gear. When we don't get the results we expect it is easy to blame the gear rather than look at the space in which they were recorded or mixed. Do yourself a favour – don't be frightened, give it a go. Before you know it you'll be hearing what's actually there!

Gary Bromham is a writer/producer/engineer from the London area. He has worked with many well-known artists such as Sheryl Crowe, Editors and Graham Coxon. His favorite Reason feature? "The emulation of the SSL 9000 K console in 'Reason' is simply amazing, the new benchmark against which all others will be judged!"

Post tagged as: **Reason**, **Record U**, **Recording**

Tools for Mixing: Reverb

By Ernie Rideout

Of all the tools we talk about in the Tools for Mixing articles here at Record U, reverb is unique in that it's particularly well suited to make it easy for you to create clear mixes that give each part its own sonic space.

Reverb derives its uniqueness from the very direct and predictable effect it has on any listener. Since we humans have binaural hearing, we can distinguish differences in the time between our perception of a sound in one ear and our perception of the same sound in our other ear. It's not a big distance from ear to ear, but it's enough to give our brains all they need to know to immediately place the location of a sound in the environment around us.

Similarly, our brains differentiate between the direct sound coming from a source and the reflections of the same sound that reach our ears after having bounced off of the floor, ceiling, walls, or other objects in the environment. By evaluating the differences in these echoes, our brains create an image accounting for the distances between the sound source, any reflective surfaces, and our own ears.

The good news for you: It's super easy to make your mixes clearer and more appealing by using this physiological phenomenon to your advantage. And you don't even need to know physiology or physics! We'll show you how to use reverb to create mixes that bring out the parts you want to emphasize, while avoiding common pitfalls that can lead to muddiness.

All the mixing tools we discuss in this series — EQ, gain staging, panning, and dynamics — ultimately have the same goal, which is to help you to give each part in a song its own sonic space. Reverb is particularly effective for this task, because of the physiology we touched on earlier. As with the other tools, the use of reverb has limitations:

1. It cannot fix poorly recorded material.
2. It cannot fix mistakes in the performance.
3. Any change you make to your music with reverb will affect changes you've made using the other tools.

As with all songwriting, recording, and mixing tools, you're free to use them in ways they weren't intended. In fact, feel free to use them in ways that no one has imagined before! But once you know how to use them properly, you can choose when to go off the rails and when to stay in the middle of the road, depending on what's best for your music.

Before we delve into the details of using reverb in a mix, let's back up a step and talk about what reverb is.

Reverb: Cause and Effect

At its most basic, a reverberation is an echo. Imagine a trombonist standing in a meadow, with a granite wall somewhere off in the distance. The trombonist plays a perfect note, and an echo follows:

Fig. 1. This is a visual representation of a basic type of reverb: a single echo. The listener hears a trombonist play a note, and then the subsequent echo. Even with eyes closed, the listener can picture how far away the reflecting wall might be, based on how long the sound took to reflect, which direction it seemed to come from, and its loudness.

Now lets put the trombonist on the rim of a large canyon. Once again, the trombonist plays a perfect note, and this time several echoes come back, as the sound reflects off of stone walls at differing distances and of differing angles.

Fig. 2. The trombonist plays the note again, this time from the rim of the Grand Canyon. The listener is also on the rim of the canyon, and hears the original note, followed by the subsequent

echoes. With the diminished volume of each echo, the listener can easily picture how far away the canyon walls are. Even if each echo is a perfect copy of the original sound, as long as it diminishes and volume and seems to come from a location other than that of the original sound, the listener's mind places the trombonist in an imagined space.

Trombonists being highly sought after in Sweden, even to the point of being an imported commodity, let's put our trombonist in the Stockholm Konserthuset, one of the finest concert halls in Europe. This time, rather then producing a series of individual echoes, the note our trombonist plays generates numerous echoes that overlap in time, ultimately creating a wash of sound that decays gradually.
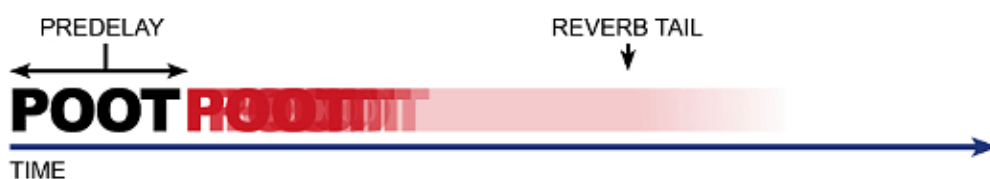
Fig. 3. Once onstage at the Konserthuset, the trombonist plays the note again. This time, the echoes are so numerous as to blend into a wash of sound. To the listener in the front row, still with eyes closed, the length of the predelay (the time from the initial note attack until the time the first reverberation occurs) and the length of the reverb tail (the gradual decay of the wash of echoes) provide enough information for them to imagine the size of the stage, the location of the walls, the height of the ceiling, and other characteristics.

Heading upcountry a bit, we'll put our trombonist in the Uppsala Cathedral, one of the largest medieval cathedrals in Scandinavia. Standing right in the middle of the cathedral, the trombonist blows a note and is immediately surrounded in a wash of reverberation that seems to last forever.

Fig. 4. In a cathedral, the note the trombonist plays seems to expand and reverberate endlessly as

the sound reflects off of the many stone surfaces to cross and re-cross the vast space. The mind of the listener can picture not only the dimensions of the space, but also the material with which it's constructed, based on which overtones reverberate the longest.

Though simple, the reverb scenarios above represent aspects of how you can use reverb and delay to create sonic space around your tracks — and they explain why these effects are, well, effective. Plus, they also represent the real-world phenomena that inspired the creation of the reverb effects that are the basis of all studio reverbs. Let's check out some notable milestones in reverb development, as this knowledge will also make it easier for you to dial up the exact reverb effects you need.

Man-made Reverb: Chasing the Tail

For recording orchestral and chamber music, the simplest way to get a great reverb is to put the ensemble in a space that produces a great reverb, such as a concert hall or cathedral, and record the performance there. Of course, there are aspects of this that make it not so simple, such as the cost, the delays due to unwanted sounds caused by passing trucks or airplanes, and the lack of availability of such venues in general.

In the mid-20th Century, many recording studios were built with rooms big enough to hold large orchestras, in the hopes of re-creating that naturally occurring reverb. In some cases, these rooms definitely had a sweet sound. In many, however, though they could hold an orchestra, the sound was not reverberant enough.

There are many reasons that recording engineers are called engineers, and one of them is their resourcefulness. To overcome the reverberation situation, engineers would convert or build rooms in the same building as the studio, sometimes adjacent to it, and sometimes underneath it. These rooms would have a certain amount of reverberation caused by the surface material, the angles of the walls, and objects placed within the room to diffuse the sound. By placing a speaker in such a room and sending a recorded signal to the speaker, the signal would have the reverberation characteristics of the room. By placing a microphone in the room to return this reverb-processed sound to the mixing desk in the control room, the engineers could then add the processed signal to the original recording to give it the extra reverberative quality. This is the basis of the effects send and return capability found on almost all mixers, real and virtual. And when you see chamber or room listed as the type of reverb in a reverb processor, it's this kind of room they're trying to emulate.

Some studios succeeded in creating reverberation chambers that created a convincing reverb, such as those at Abbey Road in London and at Capitol Records in Los Angeles, which were used on the recordings of the Beatles and Frank Sinatra, respectively. But these didn't work for every kind of music, and you couldn't vary the amount of reverb time. There was definitely a market for some kind of device that would let engineers add reverb to a track without having to build an addition to their studio. Since steel conducts sound and transmits it well, plate reverbs were developed, in which a steel plate would be set to vibrate with sound introduced by a transducer at one end of the plate, and the processed sound would be captured by a pickup at the other end of the plate.

A German company called EMT produced the most popular of these in the late 1950s, which featured up to six seconds of reverb, a movable fiberglass panel that cold vary the decay time, a mono send input, and stereo return outputs. Their sound was smoother than that of a real room, but also darker. Though these were attractive attributes when compared to the cost and inflexibility of a reverb chamber, they were far from convenient: In their cases they were eight feet long, four feet high, and one foot thick! Consider that the next time you dial up a plate reverb preset on your processor.

What recording studios did have on hand were lots of tape recording machines. By sending a signal from the mixing desk to a dedicated tape recorder, recording the signal with the record head, and then returning the signal via the playback head to the mixing desk, a single echo of the signal was created, due to the distance between the record and playback heads on the tape recorder. This delayed signal could then be blended with the original. This is called a **slapback echo**, and it's prevalent on lots of rock and roll recordings from the 1950s. Even though the effect was just of one echo, it still imparted a sense of space to the instrument or voice to which it was applied, setting it apart from the other parts.

This poor man's reverb was improved when engineers figured out how they could take the tape-delayed signal from the playback head and route it back to the record head, creating multiple delays. This became known as **tape delay**, and crafty engineers developed ways to keep the sound from building up too much (feedback), so the effect would be of three or four quick echoes that got quieter with each iteration. This added another dimension to the spatial effect, and when engineers started sending this multiple-delayed signal into their dedicated reverb chambers, they discovered yet another useful reverb application.

Fortunately for you, there is Propellerhead Reason, so you don't have to build underground rooms or rewire reel-to-reel tape recorders. In fact, you don't need anything except your computer and Reason! No matter what hardware or software recording devices you work with, keep in mind the technology behind these historical developments, as well as our travelling trombonist, as we work with reverb applications in the next section.

It is Better to Send than to Insert

In each of the other Tools for Mixing articles, we created rough mixes using just the single tool featured in the article. We did this purely to explore the power of each of the tools brings to your music, not to suggest that you create a final mix using only panning, or EQ. In fact, in creating the rough mixes, we applied the tools sometimes to extreme levels, which you normally wouldn't do when crafting a mix. Normally, you'd use all your tools in equal amounts to make each track stand out just the way you want.

We're going take a similar approach with reverb, though in some cases, we'll actually grab the channel faders and make adjustments to achieve the full effect of placing sounds in the soundstage.

Another difference between reverb and the other mixing tools is the point at which it's best to apply it in the signal path: as an **insert effect** or as a **send effect**. Here's the difference between the two.

*Fig. 5. **Insert effect signal path:** When you add an insert effect to a mixer channel, the entire signal for that track is processed by the effect, whether you've chosen a reverb, EQ, compressor, or any other effect. The only control you have over the amount of signal processing is with the effect's wet/dry mix, which is the mix between the unprocessed and processed signals. This method is best for effects that have more to do with the sound design of an individual track than with the sound of your overall mix. While you're mixing, it's a pain to go into the controls of individual insert effects to change the wet/dry mix. This limits your flexibility when mixing.*
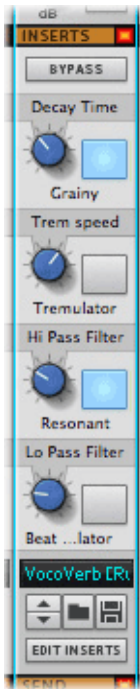
*Fig. 6. **Insert effect section in Reason:** Here is a mixer channel in Reason that has an insert effect applied to it, in this case a very wacky vocal processor effect. It's easy to tweak the four available controls, two of which are active here, but it's not easy to adjust the wet/dry mix from the mixer.*
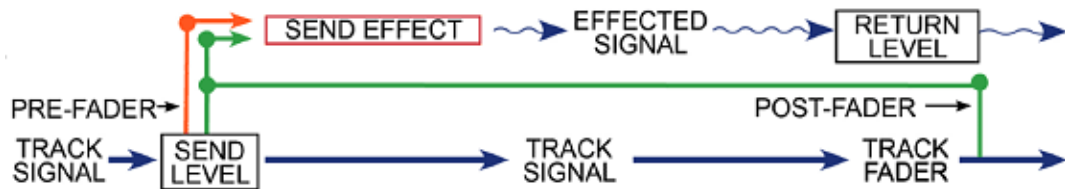
*Fig. 7. **Send effect signal path:** To process a track with a send effect, you engage the send effect, which splits the signal. The **Send Level** knob lets you set the amount of signal that gets sent to the effect. The processed signal goes through the **Master Effects Return** control, which lets you set the amount of processed signal you want to mix with the unprocessed signal via the effects return bus — a key element when it comes to mixing. Using the channel effects send controls in their default state, the send occurs **post-fader** (green arrow). In this mode, you set your send level with the **Send Level knob**. The channel fader then boosts or cuts the send level as you move the fader up or down, respectively, relative to the setting of the send knob. Any adjustments you make with the channel fader will affect both the track volume and the send level. The **Return Level** knob on the master channel determines the global level of the return signal mixed in to the master bus. In other words, the balance between the effect and dry signal remains proportional as you move the channel fader. If you choose **pre-fader** by clicking on the **Pre** button, then the channel fader has control over the effects send level as determined by the position of the Send Level knob (orange arrow). The level of the processed signal is determined only by the settings of the Send Level knob and the Master Return Level knob. Having these two options gives you a lot of control over how you blend processed and unprocessed sounds in your mix.*
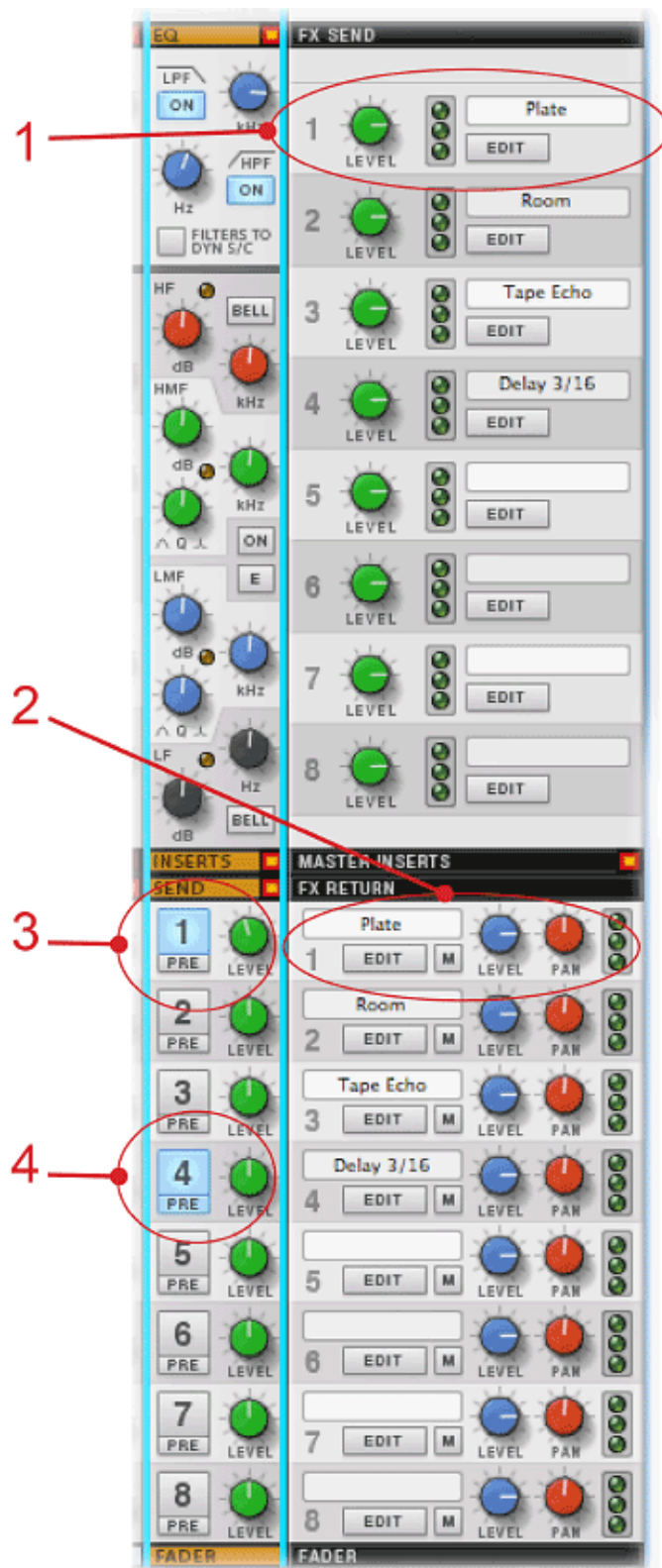
*Fig. 8. **Send effects in Reason:** This shows the overall effect send levels and effects in the Master Channel (1), the return levels in the Master Channel (2), an individual track send button and send level (3), and an individual track send button and send level with the pre-fader button en-*

gaged (4). In the examples that follow, we'll be making most of our adjustments just with the individual channel controls.

Most engineers use reverb as a send effect, not as an insert effect. This allows much more flexibility and control during mixdown. You can easily achieve a more unified sound by sending multiple tracks to the same reverb, create individual locations for tracks by adjusting send levels, or distinguish different groups of tracks by applying different reverbs to all tracks in each group.

Since we're committed to giving you the best practices to adopt, we'll focus on using reverbs as send effects.

Create a Mix Using Reverb: Give a Single Track Its Own Space

This brief excerpt features a great Redrum pattern from the ReBirth 808 Mod Refill and a meandering Malström synth line triggered by an RPG-8 random arpeggiator pattern. The drum part is busy, to say the least. The arpeggio covers four octaves, and varies the gate length as the pattern progresses, resulting in staccato sections followed by legato sections. The basic levels are comparable. Give a listen:

The synth is certainly audible, but it gets lost in the ReDrum. Let's see if we can create some sonic space for it. We'll activate the default RV7000 plate reverb in the Master FX Send section by clicking on the first send button in the Malström channel strip.

Wow. That made a huge difference in the presence of the synth part. Even the low staccato notes stand out, and the smooth plate reverb seems to reinforce not only the individual pitches, but also the loping random melody overall. And that's just with the default settings, not even with any adjustment to send or return levels! Let's try the next default send effect by activating the second send on the Malström channel strip, which feeds a room reverb also on the RV7000.

The room reverb definitely gives the synth notes some space and makes them more present. But it doesn't have the smooth sustain of the plate reverb. Let's tweak the send level on the channel strip by cranking it up about 10 dB and see what that sounds like.

Increasing the send level had two interesting effects: It gave the synth its own sonic space, but that space sounded like it was way behind the drums! This is the basic idea of how you can use reverb to make one track sound like it's toward the back of the sound-

stage, and another sound like it's toward the front. More effect = farther away from the listener. We'll experiment more with this a little later. Now let's try out the next send effect, which is a tape echo created with a Combinator patch that uses two DDL-1 delay instruments and the tape emulation effects from a Scream 4 sound destruction unit.

The multiple echoes reinforce the sound while giving it a very distinct sense of space. This kind of effect isn't for all types of music, but it works great with a nice melodic synth patch like this. Let's try out the fourth and last default send effect, which is a very simple 3-tap delay from a single DDL-1 instrument.

Very interesting. There is no reverb per se applied to the Malström track, yet it sounds like it has reverb on it. It's more present in the mix as well. This is because on the sustained notes, even though we can't hear the echoes distinctly, they definitely create a sense of sustained reverb. On the staccato notes, you can still hear the delayed echoes, but since they're rapid and they decay quickly, they continue the apparent effect of reverberation. This is why engineers use both reverb and delay to help give each track its sonic space; even when used by itself, delay can create a very convincing sense of space.

This also explains why we had our trombone player go to the Grand Canyon earlier: To demonstrate that echo, delay, and reverb are effective variations of the same tool. What else have we picked up on?

- Increasing the send to a reverb makes a track recede from the listener
- Delay can have the same overall effect as reverb in a mix

Create a Mix Using Reverb: Make Tracks Come Forward and Others Recede

This track is a wacky bit of big jazz that has a brief feature for the saxophone section, which consists of soprano, alto, tenor, and baritone sax. Each part is angular and the harmonies are crunchy, to say the least. Let's have a listen.

You can hear that there are individual parts, but it's difficult to discern them, even though the tone is distinct for each instrument. They definitely don't blend, and there is not a clear sense of which instrument has the lead. Let's start by adding some room reverb to all four instruments, using the second send effect controls in each channel strip.

Putting the entire section in the same reverb space does help make them sound more distinct. It also makes the section sound like it was in the same room at the same time as

the rhythm section, though they're obviously closer to the listener. It's a good start. Let's assume the soprano sax has the lead, and bring it to the fore a little bit more by increasing the send levels of the other three saxes.

Well those three saxes sure sound like they're in the background. But they're still just as loud as the lead soprano sax. Let's adjust their respective levels just a bit using the channel faders, to see if we can create the sense that the soprano sax is stepping forward.

By reducing the three lower saxes by 3 dB and boosting the soprano 1 dB, we've created a pretty convincing audio image of the lead player standing closer to the listener than the rest of the section. Musically, we've probably gone a bit overboard, as this might not be the best mix. But what the heck, let's switch things up and make the tenor player step out front.

Tenor players unite! With the tenor at 1 dB on the fader and -12 dB at the send, while its colleagues are all at -3 dB at the faders and 0 dB at the sends, we've created a clear picture of one instrument coming forward toward the listener and others moving away, even though it's not playing the highest part.

You can use these methods on background vocals, rhythm section tracks, and any tracks that you want to sound unified, yet at different distances away from the listener. The takeaway:

- Sending a group of tracks to the same reverb gives a unifying sound
- To bring a track forward, bring up the fader and reduce the effect send
- To send a track away from the listener, bring down the fader and increase the effect send

Create a (Rough) Mix Using Reverb

This blues track you may recall from other Tools for Mixing articles. We'll try the X-Games mix on it, using reverb and a little bit of level adjustment only to create a rough mix. As we said before, this is not a "best practice" for creating a rough mix. It is, however, a good way to learn what best practices are for using reverb and delay to create sonic space for each track in your mix. The track levels are consistent, and the panning is all right up the center. Let's give a listen.

We can hear all the parts, but the individual tracks are very dry. There's no sense of

blend, and all the instruments are right up front. Let's start by putting the horns and drums in the back of the soundstage by increasing the channel strip send levels to the default room reverb and bringing down the channel faders a bit on the horns.

All right, now we've got a stage going on. The drums are still very present, but their sound is defined by the size of the stage. The horns sound like they're behind the guitar and organ, right where we want them for now. The guitar needs its own space, let's try some delay to see if that sets it apart.

Sending the guitar track to the tape echo certainly sets it apart. We also sent the organ to the room reverb, though not so much as to make it recede. It's starting to sound like a band. But there might be a couple more things we can do to make it sound better. One danger to sending groups of instruments to the same reverb is that the muddy low-mid frequencies can start to build up, and this might be the case with this rough mix. Let's see if we can't bring those down by editing the EQ on the RV7000 that's producing the room reverb.

Fig. 9. To edit the EQ on the RV7000, go to the Rack View, click on the EQ Enable button, and click on the triangle to the left of the Remote Programmer. This brings opens up the Programmer. Click on the Edit Mode button in the lower left corner until the EQ mode light is lit. The RV7000 gives you two bands of parametric EQ to work with, and for our needs, the Low EQ is all we need. Set the Low Frequency to its highest setting, which is 1 kHz, and crank the Low Gain all the way down. This creates a highpass filter that removes all the low and mid frequencies that were bouncing around our virtual stage due to the effect of the room reverb.

Ah, cutting the low EQ on the RV7000 helped a lot to open things up, but it still leaves us a sense of space. For fun, we also put some plate reverb on the guitar along with the tape echo, which really put it in a unique space that doesn't interfere with the other instruments. We sent the organ to the room reverb, to make it seem like it's part of the session, but still up front.

Perhaps not the mix we'd send to the mastering studio, but certainly one that shows how easy it is to use reverbs and delays to set up a virtual soundstage! The takeaway from this mix:

- Cut the mids and lows out of the reverb using the built-in EQ on the reverb itself; this will help keep your mix from getting muddy
- Send tracks to the back by increasing the send and bringing down the level
- Blend tracks where possible by sending to the same reverb
- Help individual tracks find their own space by sending them to a reverb or delay different from most other tracks

Combined with your mastery of the other mixing tools, your knowledge of how to use reverb and delay in a mix will help you get the mixes you want in a minimum amount of time!

Based in the San Francisco Bay Area, Ernie Rideout is Editor at Large for Keyboard magazine, and is writing Propellerhead Record Power! for Cengage Learning.

Post tagged as: **Mixing**, **Reason**, **Record U**, **Reverb**

Tools for Mixing: Insert Effects

By Ernie Rideout

In the Tools for Mixing series here at Record U, we discuss a number of useful types of effects and processing in other articles: dynamics such as compression and gating, EQ types such as shelving and parametric, send effects such as reverb and delay, and master effects such as maximizing and stereo imaging. Most importantly, these other articles cover how you can use these effects and processors to make your mixes sound great.

What's different about insert effects, the subject of this article? Well, in some ways, absolutely nothing. Insert effects can be compressors, EQs, reverbs, delays, and any other kind of processor. Like these effects and processors, insert effects are very effective tools to give each track its own sonic space and to make your mix sound better — it's these goals that we'll focus on in this article.

In addition to their fundamental assignment as mix improvement devices, insert effects can be used as sound design and arrangement tools as well. Sound design means altering the sound of something, such as taking the sound of a guitar and making it sound a little different, significantly different, or even unrecognizable. Doing so can have a profound effect on the emotion of a track and even an entire song. In an arrangement, you can use insert effects to create subtle and not so subtle changes to the sound of an instrument or voice as the song progresses from one section to another. Using automation, you can have an effect get more intense in the choruses when the instruments are playing at full volume, then less intense in the quieter verses.

What?? You haven't used automation before? It's easy as pie. It's very similar with all multitrack recording software, but it's particularly easy in Reason. Let's digress for a quick lesson on automation.

To record automation data for an effect parameter, simply alt-click or option-click on the knob, dial, or slider you want to automate. In this illustration, we alt-clicked on the Feedback knob on the DDL-1 Digital Delay device. A green box outlines the control, as you see here, to let you know that the control is now automated. At the same time, a Feedback parameter automation lane has been created for this device in the sequencer, as you can see at the bottom of the illustration. The parameter record button is red, indicating the parameter lane is record-ready.

Next, all you have to do is hit the record button in the Transport Controls, and then move the Feedback knob with your mouse to change the sound as your song progresses. You can see the cursor has been dragged up, and a parameter display appears indi-

cating that the Feedback parameter has been increased to 44. In the sequencer, you can see automation curves being recorded.
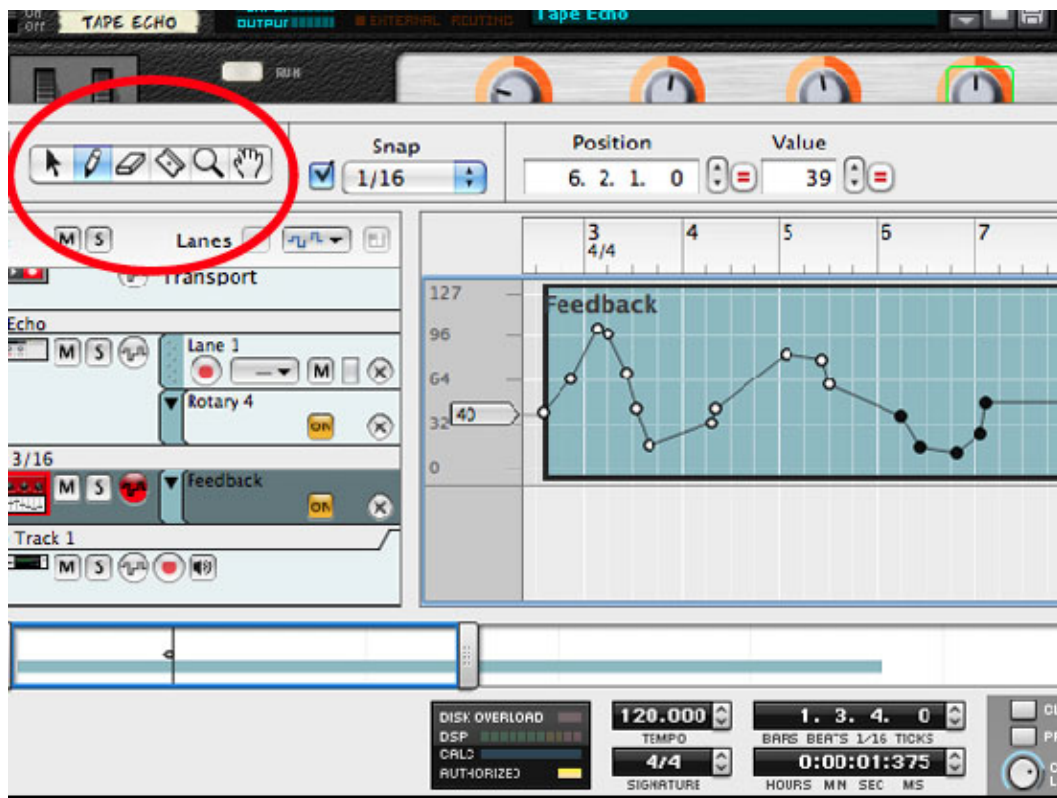
When you finish a pass, you can easily edit and change the automation data you've recorded. Just double-click on the automation clip, and whether you're in arrangement mode or in edit mode, you can edit the data points with the editing tools, such as the Pencil tool shown here.

There. Couldn't be simpler.

For our purposes as budding mix engineers, the most significant differences between insert effects and other effect applications are the following:

- Insert effects work only on one channel, not across multiple channels.
- Insert effects process the entire channel; there is no un-effected track signal once the effect is active.

This means that though you have tremendous power over the fate of a single track with insert effects, the worst thing that could happen to your mix is that only one track might

sound crappy. And you can always delete the effect and go back to your unprocessed track.

Adding insert effects is usually one of the last things you do when you're mixing — unless you think like a sound designer, in which case adding insert effects to your tracks is one of the first things you do. With a song featuring acoustically recorded instruments, the order of business in a mix session usually flows like this:

1. Set gain staging.
2. Apply basic EQ to individual tracks.
3. Apply basic dynamics.
4. Set panning.
5. Apply send effects.
6. Apply insert effects.

Sometimes you'll apply more than one tool simultaneously. Sometimes you'll do things in a different order. The point is that insert effects are usually what you add to fine-tune your mix, to give individual tracks something special. As with all of the tools we discuss in Tools for Mixing, you could achieve a decent basic mix using insert effects alone; they're that powerful.

Do This Before You Apply Insert Effects

No matter how you like to approach your mix, before you apply insert effects, make sure you do two things to your entire mix:

1. Using a high pass filter on the channel EQ, roll off all frequencies below 150 Hz on all your tracks, especially those tracks that you don't think have any energy at those frequencies. The exceptions are the kick and the bass. Even on the bass, roll off everything below 80 Hz. Doing this will make your mix sound much more open right away, and it's critical to remove unnecessary energy in these frequencies before your insert effects amplify them.
2. Listen to each track carefully, and remove any sounds that you don't want, including buzzes, bumps, coughing, fret noise, stick noise, or any sound made by a careless musician. Any unwanted sounds may get amplified by the application of insert effects.

Signal Routing

Now that you're at this stage of your mastery of mixing, it's time to reveal what the deal is with signal routing. Most mixing boards and recording programs let you choose the order in which each track signal passes through the various processors and effects. At the top of each channel in Reason's mixing board, you can see a little LCD diagram, entitled Signal Path. Just above the LCD diagram, there are two buttons: Insert Pre and Dyn Post EQ. When engaged, the Insert Pre button places the insert effects prior to the dynamics and the EQ; when disengaged, the insert effects come after the dynamics and the EQ. When the Dyn Post EQ button is engaged, it means the compressor and gate follow the EQ; when disengaged, the dynamics are in front of the EQ.

Fortunately, the wonderful mixer gives a very clear picture of how the signal path options work. Let's look at this LCD picture as we discuss the musical reasons for choosing one signal path over another. And keep in mind that when you change the signal path, the compressor and gate don't jump to the bottom of the channel strip, and the insert effects don't slide on up to the top. In the channel strips themselves, the controls stay in their place. It's just the audio that follows its various courses behind the scenes.
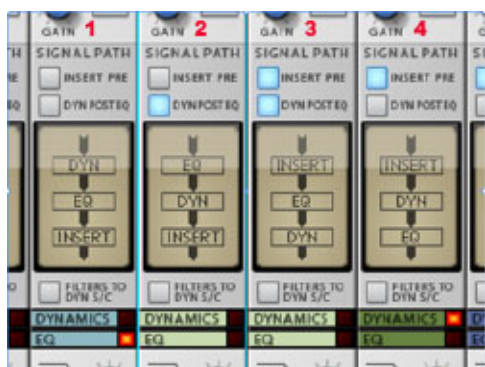
When you're gaining experience as a mix engineer, it's always nice to have a compressor at the end of your signal path to attenuate any extreme boosts in the signal you may inadvertently cause. Without the compressor in that catchall position, a severe peak might get all the way to the main outputs, where it could cause clipping — and it might take a long time to figure out what's causing the overage. If you're an experienced mix engineer, you're likely to be vigilant for such accidents, and can therefore choose any signal path option for any reason you like. Here are musical reasons for choosing each of the four options for signal routing in Reason's mixer over the others.

1. In this configuration, the compressor comes first, so it tames the peaks in the track. The EQ is next, opening up the options for sound sculpting — but with no compressor or limiter following it to compensate for any boosts that might lead to clipping farther down. The most musical choice of an insert effect here would be another compressor, in order to get a super-compressed sound that moves forward in the mix. You'll see this in action in Audio Example 20, later on.
2. Similar to configuration No. 1, this puts the EQ first, and then the compression. The

most musical choice for an insert effect here would be another EQ, in order to get a very specific frequency curve dialed in.

3. This is the best configuration for less experienced mixers. You can make bold choices in the insert effect slot and really get some fun into the track, be aggressive with dialing in the frequencies you want with EQ, and then the compressor will be there to even out anything that gets out of hand.

4. This configuration puts the compressor after the insert effect, which is good for safety, but with the EQ at the end, this configuration is best for an experienced engineer who uses EQ as their primary mixing tool even when fine-tuning a mix. This means the artistry such an engineer puts in to their EQ won't get squashed by a compressor farther down the signal path.

And Now, the Insert Effects

Insert effects themselves are simple as can be. But their interaction with other processors and mixing tools can have surprising results from seemingly subtle changes. That's why the preceding introduction is important: Any changes you make to your music with an insert effect will have an impact on the changes you've made using other tools. The difference, as noted before, is that insert effects impact the individual tracks more than the overall mix.

Insert effects come in a several types. **Dynamics (loudness) processors:** You're already familiar with compressors, limiters, maximizers, and gates. They can be used as insert effects, too. **Timbre effects** are also familiar to you already, as they include EQ and filters. **Modulation effects** include tremolo, ring modulation, chorus, flanging, vibrato, and vocoders. These effects use variations in pitch, loudness, and time to get their sound. **Time-based effects** include reverb and delay. **Distortion effects** include overdrive, vinyl effects, exciters, tube distortion, distressing, downsampling, and bit conversion. **Pitch correction effects** include Auto Tune and vocoders.

Combo effects: Amp simulators use a combination of effects in one package, including distortion, delay, compression, and EQ. The Scream 4 device uses a combination of distortion, EQ, formant processing, and compression to make its impressive sounds.

As you explore the effects available in your recording software or hardware multitrack recorder or mixer, experiment like crazy with insert effects. Try out anything and everything, and don't be afraid to just delete the effect and start over. In Reason, it's quite easy to explore effects that have already been designed for particular musical applica-

tions; just click on the folder icon (Browse Insert Effect Patch) at the bottom of the Insert Effect section and find the Effects folder in the Reason Sound Bank.

Click this early, and often, to discover what insert effects can do for your music.

In the examples that follow, we're focusing on acoustically-recorded tracks (that is, tracks that aren't created from virtual instruments) that are intended to support a song, since creating good mixes of acoustic tracks is one of the biggest challenges that the majority of songwriters face. The examples here illustrate applications of a wide variety of effects to solve common mix problems: They're designed to show the results various types of insert effects can have on the way the affected track sits in the mix. In most cases, we're applying the effects in excessive ways, going a bit over the top to make the results obvious. It's possible to solve problems in a mix using only insert effects. But in practice, you should use all the tools available to you — EQ, dynamics, send effects, panning, and inserts — to make your mix sound the way you want it to.

Vocals

This example features a chicken-pickin' rhythm section of drums, bass, and electric guitar, with a solo female vocal. No effects have been applied yet, but in addition to just setting the basic gain staging, we've carved out a little space for the vocal with a bit of EQ on the instrument channels. The vocal is audible enough to begin working more on the mix.

Make a vocal thicker by doubling

When we apply a simple doubling of the vocal track and a little detuning to the double using the UN-16 Unison device, the vocal suddenly has expanded to occupy a much more prominent place in the mix without adding any perceptible increase in level. Well, it's twice as many as doubling, as we used the Unison's lowest voice count of four (it also does eight and 16!). We set the detuning fairly low and the wet/dry mix so that the

original signal is the most prominent.

Adding a UN-16 Unison device to a vocal track is easy. Just select the track, channel, or audio device and select UN-16 Unison from the Create menu. All connections are made automatically. All you have to do is decide if you want four, eight, or 16 voices.

Thicken up a vocal with delay

Running the vocal through a basic delay with just two repeats and a fairly low feedback setting makes a huge difference in the spread and level of the vocal. When you add delay, you should always watch that the volume doesn't get carried away. This example uses the DDL-1 Digital Delay device with the wet/dry balance set rather dry, so the original signal comes through clearly. Even with just two repeats, the effect is very much like a reverb.

Ducking a vocal delay

The trouble with putting a delay on a vocal is that the repeats can get in the way of the clarity of the lyrics. You could adjust the wet/dry mix by hand, riding it towards the wet signal when each phrase ends. Or you could put a compressor on the delay and trigger it from the vocal signal. This ducks (lowers the volume of) the delay effect as long as the original vocal signal is above the threshold on the compressor. When the vocal dips below the threshold, the full delay signal comes back up in volume. This gives

the vocal a bigger place in the mix, while keeping the effect out of the way of the lyrics.
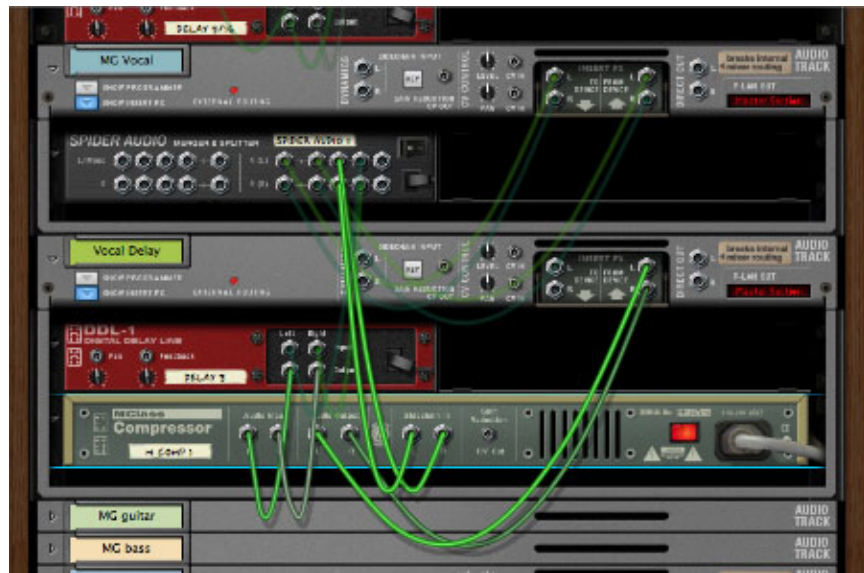
It's easy to set up a ducking delay. In this excerpt, we want the delay to be softer while the vocal is present, but to come back up when the vocal phrase ends. First, create a Spider Audio Merger & Splitter on the vocal channel. Run the Insert FX outputs into the input on the splitter (right) side of the Spider and connect one set of Spider outputs back to the Insert FX inputs. Create a new audio track for the delay, and create a DDL-1 Digital Delay and an MClass Compressor for this track. Take a second set of outputs from the Spider, and connect them to the inputs of the delay. Take a third set out outputs from the Spider and connect them to the Sidechain In on the compressor — this is what will trigger the ducking effect. Connect the delay outputs to the compressor inputs, and run the compressor outputs to the delay channel Insert FX inputs. Presto! Your ducks are now in a row.

Thicken a vocal with chorus

Another way to fatten up a vocal track is to run it through a chorus effect. This doesn't necessarily make the track more prominent, but it does seem to take up more of the frequency spectrum. For this example, we used the CF-101 Chorus/Flanger, with very little delay, and just enough modulation to make the effect obvious while not overwhelming the vocal.

Extreme reverb effects for setting off vocals

Adding reverb to a vocal track is a sure way to give it some of its own space, even apart from the reverb you might apply with the send effects. You can even get some extreme sound design effects with reverb, such as this reverse reverb algorithm on the R7000 Advanced Reverb. This may not be the most appropriate use of reverb for this particular track, but you can easily hear how the reverb imparts its own space and EQ to the vocal, setting it apart from the instruments.

Use multieffects to bring out vocals

For vocals, the best approach to insert effects often involves a subtle combination of reverb, delay, chorus, compression, and EQ. Here is our track again, this time with a blend of two reverbs, delays set to different timing in the left and right channels, a mono delay set to yet a third delay rate, and a chorus, all routed through a submixer in a handy effects preset called "Vox Vocal FX." The four channel insert effects controls have been programmed to control the various delay times and levels, the reverb decay, the delay EQ, the dry/wet balance. The vocal sounds sweeter, all the lyrics are clear, and the track seems to float in its own space in the mix. Sweet!

Distressing effects for vocals

At the other end of the spectrum, there are distortion effects. For the vocal track we've been working on, distortion may be rather inappropriate — but we won't tell the vocalist if you won't. The result is powerful. We inserted a Scream 4 device, set the Destruction mode to Scream, set the EQ to cut the low end, and set the tweaked the Body size and resonance to get a very distressed bullhorn sound. It'd be more effective if it were Tom Waits singing something that would tear your heart out, but you can hear how this may or may not be just the ticket for your own songs.

Guitar

Let's listen to the excerpt we'll be working with for the next few examples, before we start mangling the guitar. This track has mono electric guitar and mono electric bass, both recorded direct, with no amps and with no effects. The drums are in stereo, and they have a bit of room reverb on them. Although the guitar has a decent bit of sonic real estate in which to sit, it sounds kind of thin. Let's see what we can do to beef it up a bit.

Spread out a guitar with phaser

Adding a phaser to the guitar pushes it back in the mix a bit. But the gentle swirling of the phased overtones gives it a new frequency range to hang out in. The PS-90 is a stereo effect, so our mono guitar is now a stereo guitar. Even though the track is panned straight up the center, the phaser gives it a wider dimension.

When vocals or lead instruments collide with guitar

What if there's a vocal or lead instrument that shares some of the same frequency range as the phased guitar? Is the phase effect enough to make the guitar distinct from the lead? As you can hear in this example, without further adjustment the guitar and trumpet are competing. Does this mean more work with EQ and dynamics to solve the problem? Check out the next example to find out.

Making room for a vocal or lead by spreading a phased guitar

Since our phased guitar track is now stereo, let's see what happens when we widen the stereo spread. As you can hear, the two sides of the PS-90 effect are panned hard right and left, leaving the center of the track virtually guitar-free, and the trumpet part now sounds like it's all by itself. This is a great technique to use to make room for a vocal or lead instrument; sometimes just putting a time-based effect such as a phaser on a guitar or keyboard part and then spreading the sound wide is all you need to do to make the vocal stand out clearly.

Add presence to guitar with chorus

Chorus is another modulation effect that gets a similar result to a phaser when applied to a guitar in a mix, in that the guitar seems pushed back in the mix We've taken the stereo channels of the CF-101 Chorus/Flanger and spread them wide, left and right. The guitar definitely has more presence, and seems to float in a shimmery kind of way above the drums and bass.

Re-amping for sonic flexibility

Re-amping is a great technique for working with direct-recorded guitar tracks. In a nutshell, you send the signal of a direct-recorded guitar to an amp, then you record the

sound of the amp. Re-amping gives you a lot of flexibility in guitar tone. If your recording program has amp models, you can use them in a similar way. Reason features guitar and bass amp models from Line6, which you can add to a track just like you would with any other effect, and it can make a huge difference to the presence and tone of the guitar. Here, we've selected a very clean amp model, and without boosting the bass, the low strings are a lot more audible. The guitar now has a much better location in the mix.

Distortion and overdrive to bring out the guitar

Then there's the time-honored tradition of making more room for a guitar in the mix by using an amp that's overdriven and distorted like crazy. Here's the same track with the Line6 guitar amp inserted and the "Treadplate" preset selected. The correct phrase is, "My goodness, that certainly cuts through the mix now, doesn't it?" But it doesn't overpower the drums and bass, either.

Multiple amps for huge yet flexible guitar sounds

Amp modeling plug-ins such as the Line6 amps give you lots of sonic flexibility and options for getting a guitar track to sit in a mix. But sometimes you want even more from a guitar sound. In this example, we're running the same exact guitar track through three separate Line6 amp simulators. To get the maximum flexibility, we inserted a Line6 amp on each of three mixer channels, then split the dry signal using the Spider Audio Splitter & Merger device and sent it to each of the other two amps. This setup lets us use the channel dynamics and EQ on each of the amps, which allows us to roll off the low end of one amp, to just use the highs, and then dial out the high and low end on the third amp so it just projects middle frequencies. It's a great way to build up a massive guitar sound while giving you more options for making it all work in the mix.

Here are the three Line6 amps, each in their own track.
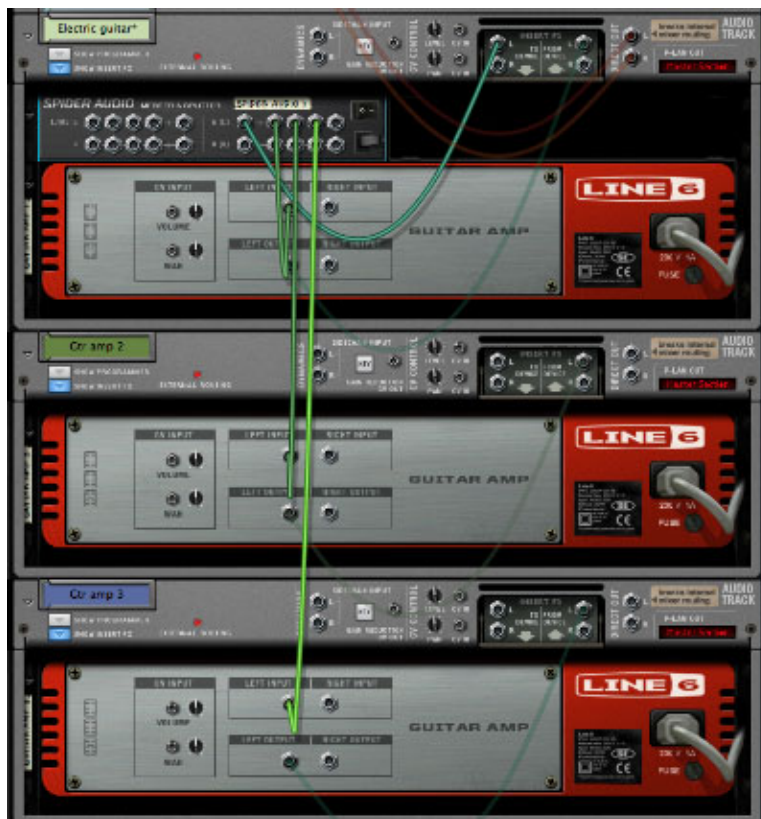
To connect the three amps in series, create a Spider Audio Merger & Splitter on the guitar track. Since we're dealing with a mono signal, run the Insert FX Left out into the Spider Left input. Run one Left output to each of the amps. Run the amp outputs back to the Insert FX inputs. Now you're ready for some serious guitar sound sculpting — as well as some powerful mix crafting.

Set the guitar apart with tremolo

Tremolo is a classic modulation effect that not only helps give a track its own sonic space, but also imparts a whole new character to the performance. This tremolo effect is created by a combination of effects in a preset called "Wobble," which you can find in the Reason Sound Bank just by clicking on the Browser Insert FX Patch button in the Insert section of any channel strip. "Wobble" uses a combination of limiters, EQ, and compression, the latter of which is controlled a CV signal triggered by the track volume. Reason has tons of effect patches that are designed to give you the effect you're looking for while helping to make the track fit into the mix. Just browse some of the effect patches to discover more.

Drums

Beef up the drums with delay

If your drum track isn't quite as full sounding as the rest of the instruments in your track, you can increase its sonic girth by adding a delay to it. The delayed signal should have no more than one or two repeats, the repeats should be so soft as almost inaudible, and the repeats are most effective when timed with the music. Start with the delay timed with the quarter-notes, then try eighth-notes, then 16ths, 32nds, 64ths, and even smaller values. If done well, the drums will just sound fuller, and you won't be able to distinguish the delay. For this example, we inserted a DDL-1 Digital Delay device on the drums, and set its dry/wet ratio to 8, so that the repeats were almost subliminal.

Give cymbals a psychedelic shimmer

Cymbals lend themselves to certain modulation effects. Applying a flanger to a drum track with lots of cymbals results in a trippy, swirling sound that also has the benefit of beefing up the track. For this example, we split the stereo drum track using the MClass

Stereo Imager, setting the crossover frequency to around 1.8 kHz, and then sending the high band output to a CH-101 Chorus/Flanger device. We adjusted the flanger to get a slow swirl, then mixed the flanged sound back in with the direct signal using a Micromix submixer device. If you have multitracked drums, then you can just slap a flanger right on the overhead channel.

Bass

Put the bass in your face

One of the more common inserts to apply to a bass track is compression, though any modulation or delay effect can sound great, too, depending on the material. Now that we've got the guitar tremolo and the drum cymbals swirling from the previous examples, the bass is sounding flabby and getting lost. Compression is a good way to attack this. We cranked up the channel compression, but it was still not quite enough. So we flipped the signal path so the insert effects came after the dynamics in the channel. We applied an insert effect preset named "Super Bass Comp" to the bass track, cranked the ratio (which was mapped automatically to the insert effect knobs), and presto! A bass sound that's solid as a rock, and seems to be coming towards you as you listen, rather than hanging back. This is the litmus test for a hard-compressed track: If you've done it right, the track comes forward in the mix. This preset utilizes three MClass devices, Compressor, Equalizer, and Maximizer.

Based in the San Francisco Bay Area, Ernie Rideout is Editor at Large for Keyboard magazine, and is writing Propellerhead Record Power! for Cengage Learning.

Post tagged as: **Mixing**, **Reason**, **Record U**

Recording Drums in your Home Studio

By Giles Reaves

Drums are probably the oldest musical instrument in existence, as well as being one of the most popular. Drums are also one of the most basic instruments, having evolved little in concept through the years: at their most basic, drums are anything you strike which makes a sound!

As simple as they are, drums can be difficult to master. The same can be said of properly recording drums. While most folks may recommend that you go to a 'real studio' to record drums, that isn't always a possibility. They will also tell you that drums are difficult to record properly, which is at least partly true. But it's also true that there's a lot you *can* do, even with a very limited setup – if you know some very basic techniques.

To introduce you to the world of drum recording at home, I've gathered some of my favorite tips and recording techniques in hopes of encouraging you to try your hand at recording some drums in your personal home studio. I'll cover a few different scenarios from the single microphone approach on up to the many options that become available to you when you have multiple microphones.

Drums in 'da House

There are many ways to approach recording drums besides the 'mic everything that moves' approach, including many time honored 'minimalist' approaches. Sometimes all it takes is a well placed mic or two to capture a perfectly usable drum recording. Luckily, this 'minimal' approach works well in the home studio environment, especially considering the limited resources that are typically available.

It's worth mentioning that there are as many drum 'sounds' as there are musical styles. Certain drum sounds can require certain drums/heads and certain recording gear to accurately reproduce. Other drum sounds are easier to reproduce with limited resources, mainly because that's how they were produced in the first place. Try to keep your expectations within reason regarding the equipment and space you have available!

Issues to be Aware of:

First, let's cover some of the potential issues you may run into when bringing drums

into your home studio:

The first issue is that drums (by design) make noise – LOUD noise. Some folks just don't like noise. This is usually the first hurdle to overcome when considering recording drums at home. The best advice may simply to be considerate of others and be prepared to work around their schedules. There is little you can do (outside of spending loads of cash) to totally isolate the drums from the outside world.

While it is unlikely, you may run into a situation where a noise from outside will intrude on your recording. Like already mentioned, there is little you can do about this other than work around the schedules of others. Most home recordists will likely have already run into these issues before, and have learned to work around them!

The second hurdle is usually not having enough microphones to 'do it right'. There are some time-tested ways to get great drum sounds using fewer mics, or even just one good mic. Rather than looking at this as an obstacle to overcome, I prefer instead to call this the purist approach!

A possible third hurdle is the sound of the room you're recording in. It can be too small (or even too big), too live or too dead, too bright or too dark. Some of these issues can be dealt with by instrument placement or hanging packing blankets, some you try to avoid with close miking! Generally speaking, a smaller/deader/darker room will be easier to deal with than the opposite. The thing to understand here is that the room itself will almost always be a factor, since the farther you move a mic from the source of the sound, the more of the room sound you will pick up.

Finally, you should also be prepared to provide headphones (at least the drummer will want phones, but will often bring their own), and make sure you have all the cables you need and that they are long enough to reach where they have to reach.

Be Prepared

Options are good – multiple cymbal choices, a few different snares to choose from, or alternate drum heads or sticks/mallets, or even different mics are all good options to have on hand (but not absolutely essential).

Ask the drummer to bring a small rug to set the drums on (a common 'accessory'), and be prepared to provide one if they don't have one (assuming you don't already have

carpet). Also consider having a few packing blankets on hand to temporarily tame any 'overly live' walls or other surfaces.

One thing before I forget – a drum kit is only as good as the drummer that is tuning and playing it. A drummer should have decent gear (no 'pitted' heads, unexpected rattles, or malfunctioning hardware please), the basic skills to tune the kit, good time/meter, and be able to hit the drums consistently. Many folks overlook this last quality, but the sound of a drum can change drastically with different stick position and velocity. The more consistent a drummer is (both with timing and with dynamics), the more 'solid' the sound will be in the end (and the better it will make you look as well!).

And finally, the actual drum part is important too – not every drummer will share your musical vision and it's up to you to keep the drum part 'musical' (whatever that means to you) and not too 'drummery' (overly busy and showing off). It may be helpful in some circumstances for you to program the drum part ahead of time (either alone or with the drummer) so that you have a reference point and are all on the same page. Let the drummer listen this track to prepare for the session, and let them know how strictly you'll need them to stick to the programmed part.

To Recap: Issues to address prior to a drum session:

- Noise Issues
- Drum/Cymbal Choice and Tuning
- Drummer's Timing and Dynamics
- Mic Choice/Placement
- Sound of the Room
- The Drum Part/Pattern

Session Day

Space is the Place

If this is the first time you're recording drums in your space, you may hear things you never heard before. This is where the packing blankets can come in handy, especially if there is ringing (Flutter Echos) or if the space is just too bright or 'roomy' sounding. If you hear these things, try to cover any large flat spaces, especially glass or mirrors. As with every other aspect of recording, you will have to experiment a bit to see which locations help with your specific issues. You may be able to locate the obvious problems

ahead of time by simply clapping (and listening) while walking around your studio space.

The physical placement of the kit in your space may be dictated by available space, but if you do have the option, try moving just the kick around and listen in the room to how it sounds. You will probably find that you prefer one location over another – I suggest choosing the position that produces the most low end, as this is the toughest frequency to add if not present in the original source. Also listen to the snare, but keep in mind you'll have to compromise in placement between the sound of all the drums in the room. You're looking for the place where the entire kit sounds its best. Don't forget to move yourself around with each new kick position. If you find a spot that sounds particularly good, put a mic there!

Once you settle on placement for the kit, let the drummer finish setting it up and fine tuning it before you begin to place microphones. You may have to guess at the placement at first, then tweak it by listening. When recording drums in the same room as your speakers, you can better judge the sound by recording the drums first and then listening to playback to make any decisions. Even when drums are in the next room, the "bleed" you hear through the wall, being mostly low end and coming from outside of the speakers, will give you a false sense of 'largeness'. So be prepared: the first 'playback' can often come as a bit of a disappointment! It may help to have a reference recording of drums that you like as a 'sonic comparison' to refer back to from time to time when getting initial drum sounds.

Now let's move on to discussing where to put the mics, once you get the drums all setup, tuned, and ready to rock. Now may be a good time to tell the drummer to get ready to play the same beat over and over for the foreseeable future!

If you only have one mic:

[NOTE: Choosing the Microphone: Any microphone that is a good vocal mic will be a great place to start when miking the drum kit with a single mic.]

There are not many options to consider when you only have one microphone to mic an entire drum kit – however, this can actually be a good thing! First off, you don't have to worry about mic selection as the decision has already been made for you. Second, there is no chance in the world for any phasing issues to be a factor! That leaves mic placement as the only concern, and that's where the fun begins.

Sometimes you have limitations in space that prevent certain mic positions (low ceilings, close walls), sometimes there may be one drum or cymbal in the kit that is louder or softer than the rest and may dictate mic position – you never know what you may run into. But if you can find the 'sweet spot', you'd be amazed at how good one mic can sound!

It's best to have a friend help with this next part, have them move the mic around the drum kit as the drummer plays a simple beat. Listen to how the 'perspective' changes. You can learn a lot about how a drum kit sounds (generally and specifically) by listening to a single microphone moving around a kit. You may have to record this first, and then listen on playback – if so, be sure to 'voice annotate' the movement, describing where the mic is as it's moved.

One mic moving from front to back of drum kit

When you listen to this recording, you can hear the emphasis change from a 'kick heavy' sound in front of the kit, to a more balanced sound in the back of the kit. The microphone, a Lawson L-47 (large diaphragm tube condenser) is about four feet off the ground. You can faintly hear me describe my position as I move the mic.

If I had to pick just one microphone position, I'd say my favorite single mic position is just over the drummer's right shoulder (and slightly to their right), pointing down at the kick beater area. Use the drummer's head to block the hi hat if it's too loud. Raise the mic higher if you have the space and want a more distant sound.

For an even more distant sound, position your single mic out in front of the kit and at waist high (to start). Moving the mic up and down can dramatically change the tone of the kit, helping you to find the spot with the best balance between drums and cymbals.

Further options with a single microphone:

Consider recording each drum separately (kick, then snare, then hi hat), one at a time. The "Every Breath You Take" approach. Or at least take samples of the each drum, and program patterns using these sounds.

In fact, if you take the time to bring drums into your home studio, you should at least record a few hits of each drum – you can cut the samples out later if time is a concern. No time like the present to start building or add to your personal drum sample library.

If you only have a few mics:

Two mics:

- First Choice: Right Shoulder (RS) position, plus Kick (K) or possibly Snare (S)
- Second Choice: Stereo Overheads

Three mics:

- First Choice: RS plus K $amp; S
- Second Choice: Kick, plus Stereo Overheads

Four mics:

- Stereo Overheads plus K & S

With four mics you can have stereo overheads plus close mics (spot mics) on Kick and Snare. Having two mics for overheads doesn't mean they have to be exactly the same exact model microphone (but should be as similar as possible). With two mic for overheads, you have many choices of microphone configurations including A-B (spaced pair), X-Y (coincident), ORTF (near coincident), M-S (using one cardioid and one figure 8 mic), the Glyn Johns or "RecorderMan" approach, or you can even try a Blumlein Pair if you have two mics that can do a 'figure 8' pickup pattern.

Beyond Four Mics

Going beyond 4 or so mics means you will begin to mic toms or even hi hats or ride cymbals. You may also opt to record more distant 'room' mics if you have enough microphones, preamps, and inputs to your recorder. The sky's the limit, but don't be too concerned if you try a mic position that ends up being discarded in the end.

Further options with a single microphone:

Obviously, with only one or two microphones to cover an entire drum kit, you can't place the mics very close to any one drum. But when you have more mics at your disposal you may begin to use what are sometimes called 'spot mics', or more commonly 'close mics'.

[NOTE: For drums, dynamic mics with cardioid or hyper-cardioid pickup patterns are

preferred for close miking, while large and small diaphragm condensers are preferred for overhead and room mics.]

With close mics on a drum kit, you are attempting to isolate each drum from the rest of the kit – this is not a precise science, as you will always have a bit of the other drums 'bleeding' into every other close mic. By positioning the mic close to the desired drum, and also paying attention to the pickup pattern of the mic you can achieve a workable amount of isolation.

When considering the position of a microphone, the most important aspect of close miking is the actual position of the mic's diaphragm in the 3D space. The second more important aspect is the pickup pattern of the mic, and how you are 'aiming' it. Most of the time, when considering close miking a drum kit, you are not only aiming the mic AT the desired source but also AWAY from all 'undesired' ones. Every directional mic has a 'null' point where it is the least sensitive, usually at the back of the mic. By aiming this 'null' point at the potential offenders you can reduce the level of the offending instruments. One common example is aiming the back of the snare mic at the hi hats to minimize the amount of hi hat bleed (a common problem with a close snare mic).

Kick Starters:

If there's a hole in the front head of the kick, placing the mic diaphragm just inside this hole is a great place to start. With the mic further inside the drum, you can sometimes find a 'punchier' position. With the mic outside the front head, you can get a bigger/fuller sound.

Snare Position:

The best place to start when miking a snare up close is a few inches above the drum head and just inside of the rim when viewed from above. I usually aim the mic down at the center of the drum, which also helps to aim the 'null' at the hi hat. But remember, it's the position of the diaphragm in the 3D space that contributes most to the sound of the snare when the mic is this close. Moving the entire mic up and down, or in and out will produce a more dramatic change than simply 'aiming' the mic differently.

Overhead Mic Options:
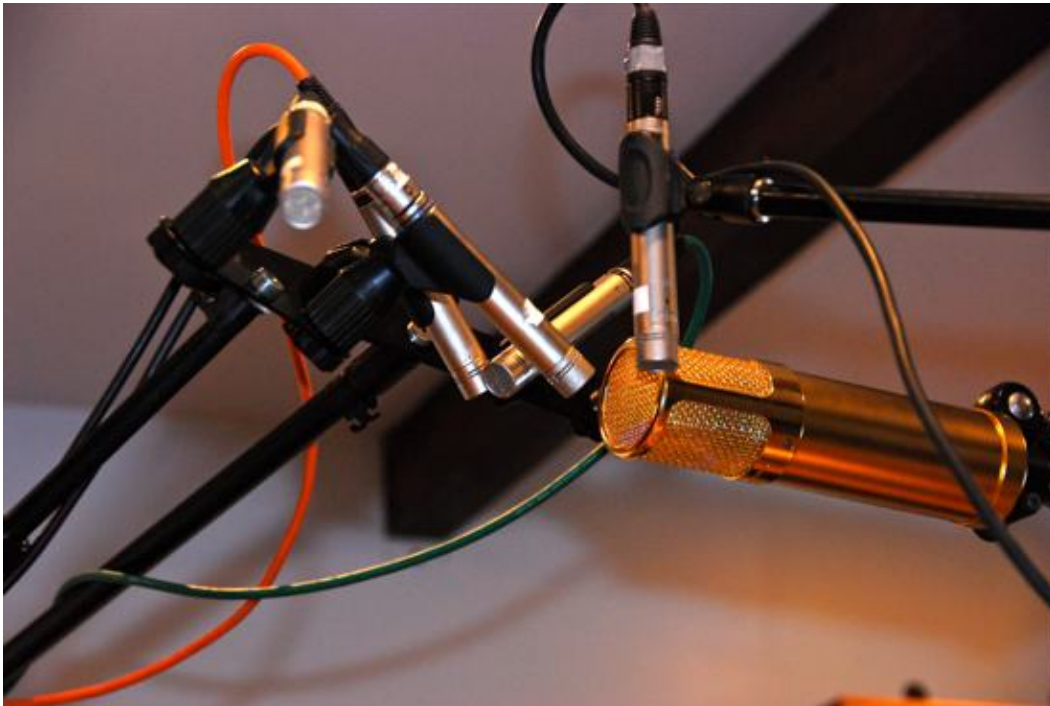
Overhead microphone 'cluster' for comparing different positions/techniques

Probably the most common miking of overheads is a spaced pair of cardioid condenser mics facing down, and about 6-8 or more feet above the ground (2-4 feet above the drums and cymbals), and as wide as required for the kit (follow the 3:1 rule for better mono compatibility, see below). Also common are an ORTF or X-Y miking configuration, but we will demonstrate all the above approaches so you can hear the differences for yourself.

There are two different general approaches to overhead drum mics: capturing the entire kit or capturing just the cymbals. With the first approach, you go for the best overall drum sound/balance from the overheads. With the second, you only worry about capturing the cymbals and usually filter out much of the low frequencies. The following techniques can be applied to either approach, with varying degrees of success.

If you have fewer overall mics on a drum kit, you will most likely need to capture the entire kit with the overhead mics. In fact, it's often best to begin with just the overhead mics and get the best possible sound there first. Then you add the kick and snare 'close mics' to bring out the missing aspects (attack, closeness) to fill out the sound coming from the overheads. So with fewer total mics, the overhead mics become VERY important.

Here are the various overhead techniques we will explore, with a short description of the technique. Also listed is the gear used to record the examples of each technique. Where possible we used the type of microphone typically used for that miking technique.
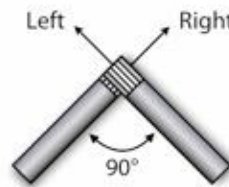
X-Y, or Coincident Pair

Rode NT-5s, Digidesign "Pre" mic pre

With this approach you are placing two mics as close together as possible, but aimed at a 90° angle to each other. The mono compatibility is second to none, but the stereo image isn't that wide. (see illustration below)

Some Common Stereo Miking Configurations

ORTF, or Near Coincident Pair

Rode NT-5s, Digidesign "Pre" mic pre

ORTF allows you to combine the best of a spaced pair and an X-Y pair. You get decent mono compatibility, but a wider stereo image. Like X-Y, one advantage is that you can use a 'stereo bar' to mount both mics to the same stand. This saves space and makes set-up a breeze as you can 'pre-configure' the mics on the stereo bar before you even put them on the stand. (see illustration above)
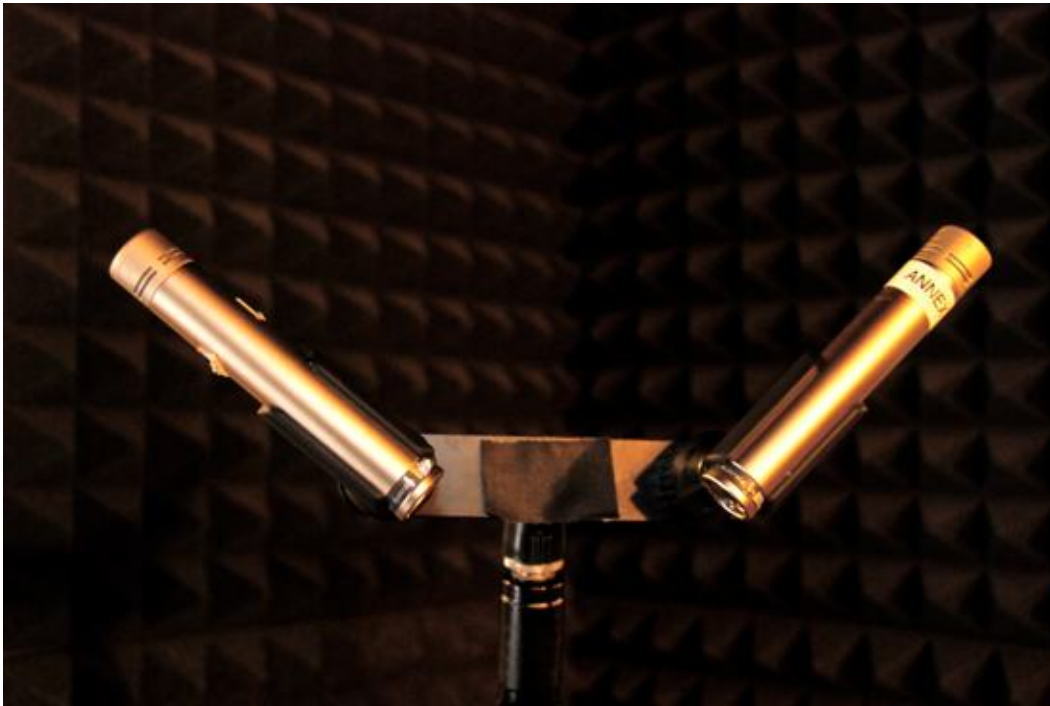
Rode NT-5s mounted on the "Stereo Bar" attachment, set to ORTF

A-B, or Spaced Pair

AKG c3000, Digidesign "Pre" mic pre
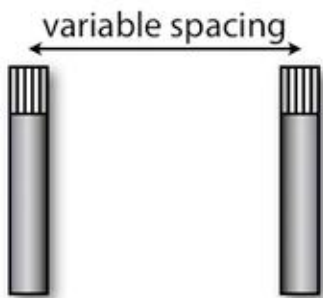
This common miking approach can be use for mainly cymbals or the entire kit. Either way, you may want to be familiar with the 3:1 rule for multiple mics: for every "one" unit of distance from the sound source to the mic, the two mics should be three times this distance from each other. If the mics are one foot above the cymbals, they should be three feet from each other. The main reason for this 'rule' is to help with mono compatibility, so don't sweat it too much if you can't hit these numbers precisely. If you check for mono compatibility (assuming it's important in your work) and you don't hear a problem, you're fine! By the way, in our example the mics are about two feet from the cymbals, three feet from each other, and doesn't seem to be a problem.

Glyn Johns Approach

Lawson L-47, API mic pre

This is a four mic approach, which using a close mic for kick and snare, and two over-heads in a 'non-standard' configuration. The first mic is centered directly over the snare, between three and four feet away. The second mic is aimed across the drums from the floor tom area, and must be exactly the same distance from the snare. Some folks pan the two overhead mics hard left/right, other suggest bringing the 'over snare' mic in half way (or even both mics in half way).

Recorderman Approach

Rode NT-5s, Digidesign "Pre" mic pre

Named after the screen name of the engineer who first suggested this approach, it is similar to the Glyn Johns approach in that you begin with a mic directly over the snare drums. But it diverges from that approach with the second overhead mic, placing it in the "Right Shoulder" position. This can also be considered an extension of the one mic 'over the right shoulder' approach. Fine tuning is achieved by measuring the distance from each mic to both kick and snare, and making each mic equal distance from each drum. This is easily accomplished by using a string, but difficult to describe in writing. For a further explanation of this technique, check out this YouTube video.

Blumlein Pair

Royer 122 ribbon mic (figure 8), Focusrite mic pre

Named after Alan Blumlein, a "Blumlein Pair" is configured using two 'figure 8' microphones at 90° to each other and as close together as possible. This approach sounds great for room mics, by the way.

Mid-Side

Lawson L-47s, API mic pres

M-S (mid-side)
(VIEWED FROM ABOVE)

Cardioid aimed forward
Figure 8 aimed to the sides

The Mid-Side technique is the most intriguing mic configuration in this group. In this approach, you use one cardioid (directional) mic and one 'figure 8' (bi-direction) mic for the recording. But you need to use an M-S 'decoder' to properly reproduce the stereo effect. The 'decoder' would allow you to control the level of the mid and the side microphone, allowing you to 'widen' the stereo image by adding more 'side' mic. This technique (along with X-Y and Blumlein) has great mono compatibility. This is because with M-S, to get mono you just drop the 'side' mic all together and you're left with a perfect single microphone recording in glorious mono.

The Session

I invited a few engineer friends to the Annex Studio for a 'drum day' to record the examples for this article. It's always more fun to do this stuff with some friends! It's a good idea to have someone move the mics while you listen – sometimes the mic doesn't end up in a position that 'looks right' (even though it may sound perfect!). We took the time to get each approach setup as precisely as possible, and recorded all of them in a single pass so they could be compared side by side.

The recording space is a large, irregularly shaped room, about 24 by 30 'ish feet with 9 foot ceilings. There are wood floors throughout (carpet under the drums) and we hung one large stage curtain to tame the room a bit for this recording. The overhead mics, for the most part, were about 6-7 feet above the floor (2-3 feet from the ceiling).

The Reason Song File

I've provided the Song File because it's easier to compare between the different miking positions when you can switch as a track plays. I've set it up so that there are "Blocks" with the title of each section. Just click on a block and hit "P" on the keyboard and that section will begin loop playback. As it is currently setup, you must mute and un-mute tracks in the sequencer – you could also do this in the SSL Mixer by un-muting all the sequencer tracks and using the Channel mutes instead.

Single Mic Sweep, front to back

The first track is a single microphone starting from in front of the kit, and slowly moving around to the back and ending up in the "Right Shoulder" position. Listen closely and you'll hear me describing my position as I move.

Compare Overhead Mic Positions

Next you will find a few bars of drums with close mics on Kick and Snare, and the following overhead tracks: X-Y, ORTF, A-B, RecorderMan, Glyn Johns, Blumlein. Playing this clip allows you to explore the different miking techniques, and allow blending of the close mics at will. All the "stereo" overhead tracks are designed to be heard one at a time, although the mics are all in phase so they certainly could be used in combination with each other if you're feeling creative. But the main purpose of this clip is to allow you to hear the difference between the various miking techniques presented.

Moved the Royers to a Room Mic Position

The third clip is a similar drum pattern, with the Royer ribbon microphones (Blumlein Pair) moved to 15 feet in front of the drums. This is our typical 'room mic' position and mic choice, and is the only difference between the previous clip and this clip. In my opinion, the sound of this miking technique combined with the 'color' of a ribbon mic makes the perfect 'room' sound. For a room mic to work, the room must sound great, of course. But also it has to be more diffused and a bit 'out of focus' compared to the close mics, which produces a similar effect as the 'blurry' background of a photo. As in the photo example, having a blurry background can help to put more focus on the foreground (close mics).

Fun with Mid-Side – Adjust M-S in Rack

Finally we have a Mid-Side recording (plus the Kick and Snare close mics) to play with. We didn't have enough mics to include it in the first round, but wanted to present it as an additional track. In addition to drum overheads, the Mid-Side approach also works well with room mics, because you can increase or reduce 'width' after the recording. I've inserted an M-S decoder on the Insert for this channel in the mixer, and by going to 'rack view' you can use the M-S combi to adjust the balance between the Mid and the Sides.

The Microphones

Kick: Sennheiser 421, API mic pre
Snare: Shure SM57, API mic pre
X-Y, ORTF, RecorderMan: Rode NT5s, Digidesign "Pre" mic pre
A-B: AKG c3000, Digidesign "Pre" mic pre
Blumlein: Royer 122 ribbon mics, Focusrite mic pre
Glyn Johns, Mid-Side: Lawson L-47s, API mic pres

The Drums

1967 Gretsch kit
22×14 Kick
16×16 Floor Tom
13×9 Rack Tom

14″ Pearl Snare
Zildjian and Paiste Cymbals

Additional Thoughts

There are always other ways to record drums. Here are a few slightly out-of-the-box approaches for your consideration.

The "Every Breath You Take" Approach:

You don't necessarily need to record the entire kit at once – this can help if you only have one mic. Things to plan for: the drummer must know about this in advance. It's not as easy as you would think to only play one instrument at a time! This approach can work especially well if you're building up a rhythm track, much like you'd program a track with a drum machine. Start with the kick, then add snare, then hi hat. Move on to the next beat. Then for fun you can us one of the 'One Mic' approaches.

The Quiet Approach…shhhhh:

Sometimes in the studio, less actually IS more! Case in point, recording drums that are lightly tapped can sometimes produce huge sounds when played back at loud levels. This approach will work best if you can record one drum at a time, and will certainly help with neighbor issues as well! You can also apply this technique to sampling as well. Consistency is the key when playing softly – sampling can help if you can't play softly at a consistent level.

Sampling, Why Not!?:

Sometimes you don't have all the ingredients for a full drum session. Don't overlook sampling as a way to get around some of these issues – and why not do it anyway! Don't forget to record multiple hits at multiple levels, even if all you need at first is one good single sample – these additional samples may come in handy later, and you never know when you'll have the drums all tuned and setup again (and it only takes a few minutes)!

Percussion

The 'shaker' family of percussion can be recorded with any mic, depending on the sound you're going for. As a starting point, any mic that's good on vocals or acoustic guitar will work fine for the 'lighter' percussion like shakers and bells etc. For hand drums like Djembes and Dumbeks, or Congas and Bongos, you can approach them like kicks/snares/toms. A good dynamic mic on the top head, and sometimes (for Djembes in particular) a good kick drum mic on the bottom. Watch for clipping – these drums can be VERY dynamic!

Special Thanks:

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Post tagged as: **Drums**, **Reason**, **Record U**, **Recording**

Vocal Production and Perfection

By Giles Reaves

So you finally finished recording all your vocal tracks, but unfortunately you didn't get one take that was perfect all the way through. You're also wondering what to do about some excessive sibilance, a few popped "P"s, more than a few pitchy lines and some words that are all but too soft to even be heard – don't worry, there's hope! And hey, welcome to the world of vocal editing.

A Little History…

Since the beginning of musical performance, singers (and instrumentalists) have craved the possibility of re-singing that one "if only" note or line. You know the one: "*if only* I had hit that pitch, *if only* I had held that note out long enough, *if only* my voice hadn't cracked", etc. With the advent of early recording technologies, these 'if only' moments were now being captured, and performers were forced to face reliving those 'if only' moments forever! One 'if only' moment could ruin an entire take.

With the popularity of analog tape recording in the mid 20th century also comes the popularity of splice editing. Now you can record the same song two different times, and choose the first half of one take and the second half of another. Next comes multi track recording, where you don't even have to sing the vocal with the band!

Multi track recording introduced punching in and out, which allowed re-recording of just the "if only" moments on an individual track. But more importantly as it relates to the subject at hand, multi-track recording also introduced the idea of recording more than one pass or 'take' of a lead vocal, leading to what is now known as "vocal comping". More on that in just a bit.

But before we get into the nitty-gritty, here's a brief outline of the typical vocal editing process for lead and background vocals. Of course, much of this is subject to change according to production direction, or the vocalist's skills and past experience.

1. **Recording:** This ranges from getting the first take, to punching in on a take, to recording multiple takes for comping.
2. **Comping:** Combining various takes into one final track, tweaking edits to fit, crossfading if needed.

3. **Basic Cleaning:** Listen in solo one time through. Typical tasks include removing the obvious things like talking, coughing, mouth 'noises' etc., checking all edits/crossfades, fading in/out where necessary.
4. **Performance Correction:** Timing and pitch correction takes place after you have a solid final comp track to work with.
5. **Final Prep:** this includes everything from basic compression/EQ, to de-essing, reducing breaths, filtering out Low Frequencies, etc.
6. **Leveling:** During the final mix, automating the vocal level (if needed) to sit correctly in the mix throughout the song.

Note that at many stages along the way you will be generating a new 'master vocal' file (while still holding on to the the original files, just in case!). For example, let's say you record 4 vocal 'takes' which become the current 'masters'. The you comp those takes together to create a new "Comp Master" vocal track, and then you tune/time the Comp Master and sometimes create a "Tuned Vocal Master" track (which is then EQ'd and compressed to within an inch of its life while simultaneously being drowned in thick, gooey FX, all before being unceremoniously dumped into what we like to call the mix).

Recording Vocals for Comping

In order to comp a vocal, you must first have multiple vocal tracks to choose from. Recording comp tracks can be slightly different from recording a 'single take' vocal. For one thing, you don't have to stop when you make a mistake — in fact, many times a performer gets some great lines shortly after making a mistake!

I tend to ask for multiple 'full top to bottom' takes from the vocalist, to preserve the performance aspects and to help keep things from getting over-analytical. Then I use comping to work around any mistakes and 'lesser' takes, choosing the best take for each line. Often the vocalist will be involved with the comping choices, so be prepared to be a good diplomat (and don't be too hard on yourself if you're comping your own vocals)!

How many tracks?

This will be different for every singer, but for comping I generally suggest recording around three to five tracks. Any less and I don't feel that I have enough choices when auditioning takes — any more and it becomes difficult to remember how the first one sounded by the time you've heard the last take.

When recording tracks that I know will be comped, I usually let the singer warm up for a few takes (while setting levels and getting a good headphone mix) until we get a 'keeper' take that is good enough to be called 'take one'. From there, simply continue recording new takes until you feel you have enough material to work with. If you find yourself on take seven or eight and you're still not even getting close, it may be time to take a break!

In Reason, when tracking vocals for future comping, you simply record each 'take' on the same track. With 'tape' recording this would erase the previous take, but with 'non-destructive' recording you are always keeping everything (with the newest take laying on 'top' of the previous take). When you enter Comp Mode, you will see each take just below the Clip Overview area (with the newest take above the older takes). The 'takes' order can easily be rearranged by dragging them up or down. Double-click on any 'take' to make it the 'active take' (it will appear in color and in the Clip Overview, and this is the take you will hear if you hit play). Now comes the fun part.

Vocal Takes in Comp Mode
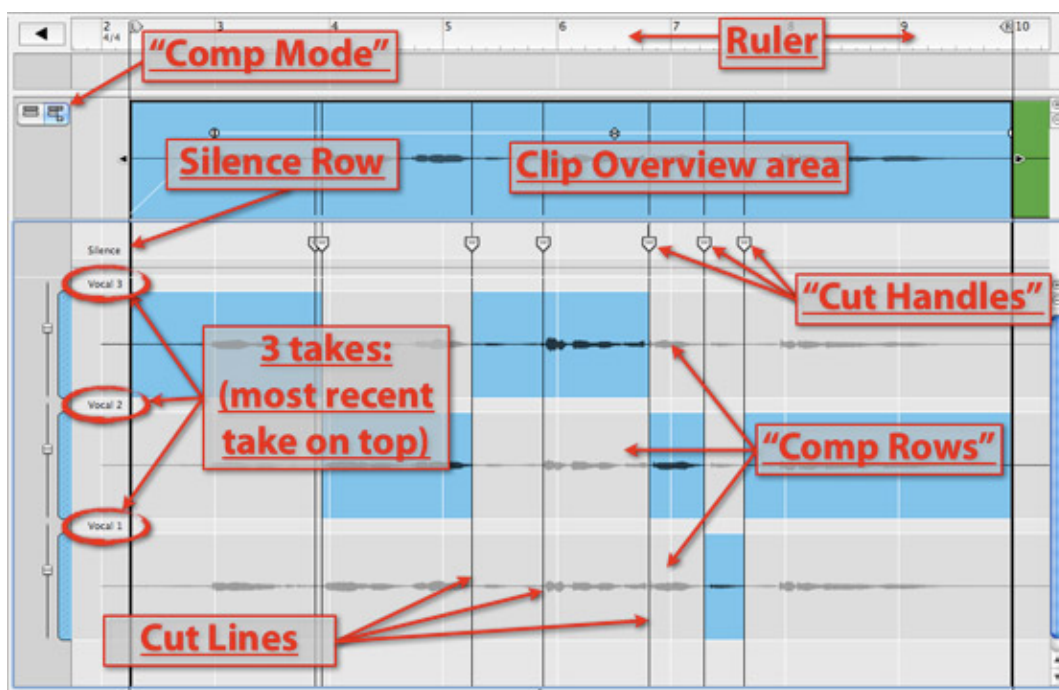
Vocal takes in comp mode.

To combine or 'comp' different parts of different takes together, use the Razor tool as a 'selector' for the best lines/words. After creating cut lines with the Razor tool, you can

easily move them earlier or later by dragging the 'Cut Handles' left or right. You can delete any edit by deleting the Cut Handle (click on it and hit the 'delete' key). Create a crossfade by clicking/dragging just above the Cut Handle. Silence can be inserted by using the Razor to make a selection in the "Silence" row, located below the Clip Overview and above the Comp Rows.

Comping Vocals

Comping (short for compositing): picking and choosing the best bits from among multiple takes, and assembling them into one continuos 'super take'.

Now that you have your vocal tracks recorded, how do you know which parts to use? I've approached this process differently through the years. Previously, I'd listen to each take in its entirety, making arcane notes on a lyric sheet along the way — this was how others were doing it at the time that I was learning the ropes. More recently I've taken another approach that makes more sense to me and seems to produce quicker, smoother, and better comps.

Currently, my auditioning/selection process consists of listening to one line at a time, quickly switching between the different takes and not stopping for discussion or comments. This is the basic technique you will see me demonstrate in our first video (see below).

Now it's time for a little thing I like to call a Video Detour. Enjoy De-tour (a-hem). Follow along in this 'made for internet' production as I comp the first verse of our demo song "It's Tool Late" (by singer/songwriter Trevor Price).

Note: watch your playback volume – the music at the top comes in soft, but it gets louder when the vocals are being auditioned.

Comping a Vocal using Reason's "Comp Mode"

Correcting/Editing Vocals

The three most common issues with vocals are pitch, timing, and level/volume. All three are easy to correct with today's digital tools and just a little bit of knowledge on your part.

Vocal timing

After comping, I usually move on to correcting any timing issues. You may also jump straight into dealing with any tuning issues if you prefer. Often times there isn't a lot of timing work that needs to be done on a lead vocal. But when you start stacking background vocals (BGVs) things can get 'messy' very quickly. User discretion is advised.

In our next video example (it's coming, I promise), I will show you how to line up a harmony vocal track with the lead vocal. I will use the lead vocal as the timing reference, moving the harmony track to match the lead. Since you can only see one track at at time when editing, I use the playback curser (Song Position Pointer in Reason) to 'mark' the lead vocal's timing, then when I edit the harmony track using this reference point to line it up with the lead vocal.

I will also use the following editing techniques:

Trim Edit, where you simply trim either end of a selected clip to be shorter or longer as desired, which will expose or hide more or less of the original recording that is inside the clip.

Time Stretch (called Tempo Scaling in Reason), where you use a modifier key [Ctrl] (Win) or [Opt](Mac) when trimming an audio clip, allowing you to stretch or shrink any clip (audio, automation, or MIDI) which changes the actual length of the audio within the clip.

Clip Sliding (my term), where (in Comp Edit mode) you use the Razor to isolate a word or phrase, and you slide just that clip right or left to align it – using this technique allows you to slide audio forward or backwards in time without leaving any gaps between the clips!
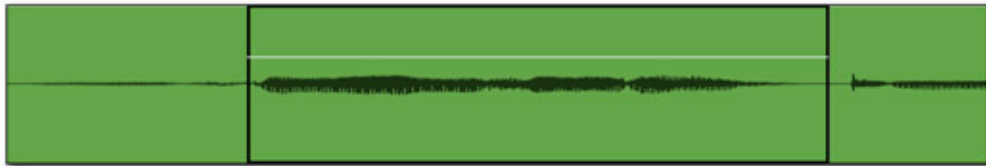

Three Easy Steps to perform a 'Clip Slide' Edit

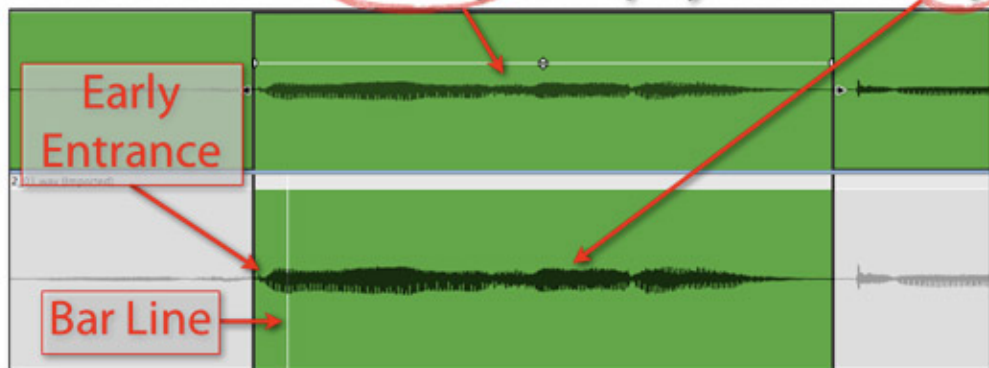Here is the original Audio (our Vocal Track)

Step 1

Use the Razor Tool to select the area to 'slide'...

Step 2

Double Click on this Clip to display the lower clip

Early Entrance

2_01.wav (Imported)

Bar Line

Step 3

Simply click and drag the lower clip to the right

no gaps after moving

2_01.wav (Imported)

Bar Line
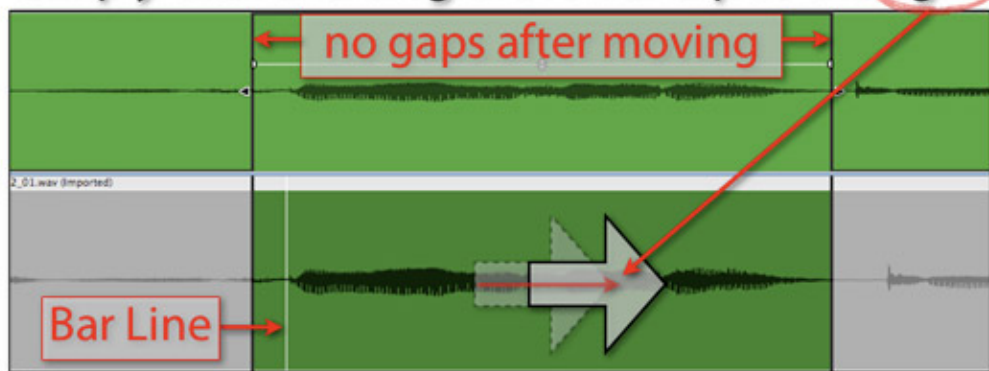
OK, thanks for waiting – here's the video:

Vocal Pitch/Tuning

Possibly an entire subject in itself, as everyone has their own take on vocal tuning. Of

course, it's always best to 'get it right the first time' if you can. But sometimes you are forced to choose between an initial performance that is emotionally awesome (but may have a few timing or pitch flaws), and one that was worked to death (but is perfect in regards to pitch and timing). If only you could use the first take with all its emotion and energy. Well now you can!

Neptune Pitch Adjuster on the lead vocal

In Reason, using Neptune to naturally correct minor pitch issues is about as simple as it gets. The following video demonstrates using Neptune for simple pitch correction, as well as using it in a few more advanced situations.

Vocal Leveling

Vocal "Rides" (as they are called for 'riding the fader/gain'), have been common from almost the beginning of recording itself. In rare cases, you may have to actually ride the vocal while recording the vocal(!) – this is the way it was done back with 'direct to disk' and 'direct to two-track' recordings. But luckily you can now do these 'rides' after the vocal is recorded, or you can even draw in these moves with a mouse (with painstaking detail, if you are so inclined). Most of the time I use a combination of both techniques.

The basic idea with vocal rides is to smooth out the overall vocal level by turning up the soft parts and turning down the loud parts (in relation to the overall mix). The end game is to get the vocal to sit 'evenly' at every point in the song, in a way that is meaningful to you. Or as I like to say, to get the vocal to ride ON the musical wave, occasionally getting some air but never diving too far under the musical water.

Great engineers learn the song line by line and 'perform' precision fader moves with the

sensitivity and emotion of a concert violinist. It really can be a thing of beauty to watch, in an audio-geeky sort of way. For the rest of us, just use your ears, take your time, and do your best (you'll get better!).

There's no right or wrong way to edit vocal levels, only a few simple rules to follow: Obviously, you don't want to ever make an abrupt level change during a vocal (but you can have somewhat abrupt automation changes between words/lines), and you don't want to be able to actually hear any changes that are being made. All level rides should ideally sound natural in the end.

As for techniques, there are three approaches you can take in Reason. The most familiar is probably Fader Automation, which can be recorded in real-time as you 'ride' the fader. You can also draw in these moves by hand if you prefer. Additionally, you can do what I call "Clip Automation", which involves using the Razor to create new clips on any word, breath or even an "S" that is too loud or too soft. Since each separate clip has it's own level, you simply use the Clip Level control to make your vocal 'ride'. Alternatively, you can use the clip inspector to enter a precise numeric value, increase/decrease level gradually in a 'fine tune' way, or simultaneously control a selection of clips (even forcing them all to the same level if desired).

The 'pros' to Clip Automation are that it is fast, you can see the waveform change with level changes, you can see the change in decibels, and you can adjust multiple clips at once. The main con is that you can't draw a curve of any sort, so each clip will be at a static level. All I know is it's good to have options, and there's a time and place for each technique!

Using "Clip Automation" to reduce multiple "S"s on a Vocal Track
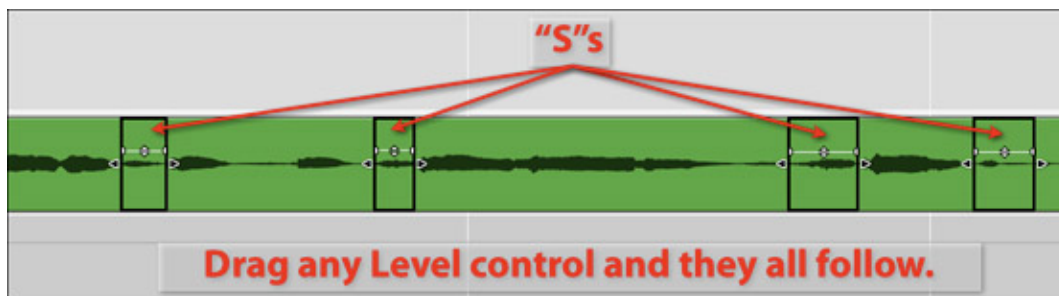
As a 'fader jockey' myself, I prefer to begin vocal rides with a fader (real or on-screen). From there I'll go into the automation track and make some tweaks, or to perform more 'surgical' nips and tucks (if needed) on the vocal track. It's these smaller/shorter dura-

tion level changes that are more often ideally created with a mouse rather than a fader. Reducing the level of a breath or an "S" sound come to mind as good examples of 'precision' level changes that benefit from being drawn by hand.

Vocal Track with Level Automation (with the first clip ready for editing…

Leveling the vocal must ultimately be done in context, which means while listening to the final mix that the vocal supposed to be is 'sitting' in (or 'bed' it is supposed to 'lay' on, or choose your own analogy!). This is because you are ultimately trying to adjust the vocal level so that it 'rides' smoothly 'on' the music track at all times (ok, so I'm apparently going with a railroad analogy for now), which doesn't necessarily imply that it should sit at a static level throughout the song.

You would think that a compressor would be great at totally leveling a vocal, but it can only go so far. A compressor can and will control the level of a vocal above a certain threshold, but this doesn't necessarily translate into a vocal that will sit evenly throughout a dynamic mix. Speaking of compression, this is probably a good time to mention that all processing (especially dynamics) should be in place before beginning the vocal riding process, as changing any of these can change the overall vocal level (as well as the level of some lines in relation to others). Bottom line – do your final vocal rides (IF needed) last in the mixing process.

Let's begin – set your monitors to a moderate level and prepare to focus on the vocal in the mix. Oftentimes I prefer smaller monitors or even mono monitoring for performing vocal rides – you gotta get into the vocal 'vibe' however you can.

Things to look for:

Before you get into any actual detail work, listen to the overall vocal level in the mix throughout the entire song. Sometimes you will have a first verse where the vocal may actually be too loud, or a final chorus that totally swallows up the vocal. Fix these 'big picture' issues first before moving on to riding individual lines and words.

When actually recording the fader moves (as in the video), I'll push the fader up or down for a certain word and then I will want it to quickly jump back to the original level. In the "Levels" video, you will see me hit 'Stop' to get the fader level to jump back to where it was before punching in. The reason why I'm doing it this way is that if you simply punch out (without stopping) the fader won't return to it's original level (even

though it's recording it correctly). Long story short, it's the quickest way I found to create my desired workflow, and it works for me (although it may look a bit weird at first)!

Some trends:

Often times you will find that it is the last word or two in a line that will need to be ridden up in level (sometimes the singer has run low on air by the end of a line). Also watch for the lowest notes in a vocal melody – low notes require more 'air' to make them as loud as the higher notes, so they can tend to be the quieter notes in a vocal track. Another thing to listen for are any louder instruments that may 'mask' the vocal at any time – sometimes the fix is to raise the vocal, other times you can get better results by lowering the conflicting instrument's level momentarily. In extreme cases, a combination of both may be required!

Other problems that rear their heads from time to time are sibilance, plosives, and other 'mouth noises'. These can all be addressed by using creative level automation, or by using a device more specifically for each issue – a 'de-esser' for sibilance, a High Pass Filter for plosives, for example.

Now, enjoy a short video interlude demonstrating the various techniques for vocal level correction, including the fader technique as well as automation techniques including break-point editing, individual clip level adjustments, and some basic dynamic level control concepts including de-essing and multi-band compression.

Controlling Vocal Levels in Reason.
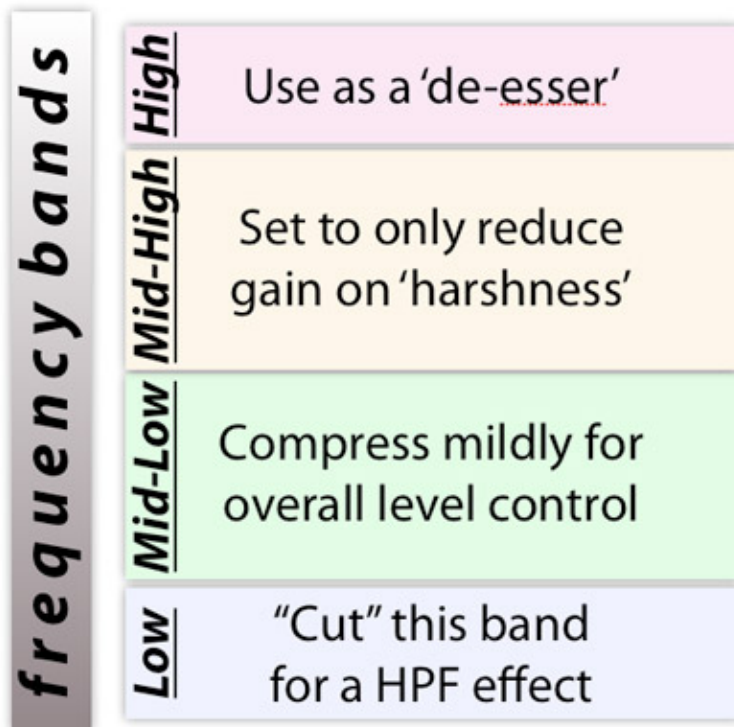
Multi-bands for Multi Processes

I will leave you with one final tip; you can use a multi-band compressor on a vocal track to deal with multiple issues at once. The high band is good for a bit of 'de-essing', the mid band can be set as a 'smoother' to only 'reduce' gain when the singer gets overly harsh sounding or 'edgy', and the lower band can be used to simply smooth the overall level of the 'body' of the vocal. If there are four bands available, you can turn the level of the bottom-most band totally off, thus replicating a high pass filter for 'de-popping' etc. Additionally, adjusting the level of each band will act like a broad EQ!

Setting the crossover frequencies with this setup becomes more important than ever, so take care and take your time. Remember you are actually doing (at least) four different

processes within a single device, so pay attention not only to each process on it's own but to the overall process as a whole. When it works, this can be the only processor you may need on the vocal track.

Multi-band Compressor as 'Multi Processor'

Multi-band Compressor on Vocals

frequency bands

High	Use as a 'de-esser'
Mid-High	Set to only reduce gain on 'harshness'
Mid-Low	Compress mildly for overall level control
Low	"Cut" this band for a HPF effect

…all of the techniques in this article, however helpful they *can* be, are not always required – do I even need to remind you all to 'use your ears' at all times? Using vocal rides as an example, I've mixed two songs in a row (by the same artist), one where the vocal automation looked like a city skyline and the very next mix where the vocal needed no automation whatsoever!

As always; **"listen twice, automate once"**!

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