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advanced reverberation

Part 1: **Paul White** looks at how real reverberant spaces affect the perception of sounds within them, and shows you how your digital reverberation unit can simulate this process.

Because studio reverberation attempts to emulate something that we all experience every day in nature, it has to be good to avoid being found out! Everything we hear involves reflected sound as well as the direct source of the sound itself, and, through the process of evolution, we've learned to interpret this reflected sound in a way that tells us something about our immediate environment, even in the dark. Take away these reflections and the sound will be perceived as almost alarmingly 'dead', as anyone who's been inside an anechoic chamber will confirm. We need sonic reflections to give a sound a sense of place and reality, yet when an instrument is close miked (or synthesized) in an acoustically absorbent studio, the contribution of these natural reflections is diminished to the point where we need to add artificial reverb to restore a sense of reality.

The reason studio reverbs have so many adjustable parameters is that the nature of reflected sound changes enormously depending on the surroundings. Compare the way clapped hands sound in a cathedral with the way they sound in the middle of a wood, and you'll hear what I mean. In any large enclosed space with hard surfaces, the reflections may take a considerable time to die away, whereas in a domestic room the reflections are so unobtrusive that they fuse with the original sound and are rarely perceived as being present, even though the sound would be radically different without them. Because the reflection pattern gets more complex every time it hits another surface, the individual reflections very quickly merge into a reverberant wash, hence the term reverberation, or reverb for short.

A studio reverberation device simulates the way that sound is reflected and re-reflected from surfaces, and the way this process is modelled fools the ear into 'believing' that the sound exists in some type of real environment. A number of parameters can be changed to alter the type of environment being simulated, and the purpose of the first part of this article is to look at both the common and less common parameters that can be adjusted in a typical studio reverb processor, whether real or virtual.

The Movement Of Sound

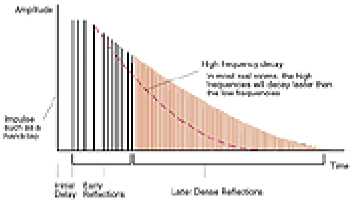


Figure 1. The anatomy of a reverb.

In nature, sound travels out from its source in the form of spherical wavefronts travelling at the speed of sound — very roughly one kilometre every three seconds. These wavefronts keep moving away from the source, diminishing in amplitude according to the inverse square law (the level falls in proportion to the square of the distance travelled), until they encounter a reflective surface. All surfaces absorb some energy, reflecting the remainder back into space, the amount being reflected or absorbed being determined by the physical properties and shape of the surface material. For example, a flat marble surface will reflect most of the sound energy like a mirror whereas a tree trunk covered in textured bark will absorb more of the energy and its irregular shape will scatter the reflections rather than reflecting them in a specific direction. This reflected sound then reflects again from any further obstacle it encounters, and as the complexity of the reflection pattern increases, the intensity of the sound decreases due to distance (the inverse square law again) and absorption by the surface materials. In larger spaces, the high-frequency absorption by the air itself also becomes a relevant factor.

It's not easy to say when a reverberant signal finally disappears, so there's a standard measurement of reverb decay defined as the time taken for the level of the reverberation to decay by 60dB. This time is also known as the RT60.

The spacing between the initial reflections is an important factor in the perception of room size, where the word 'room' in this context describes any type of reverberant space. The larger a room, the further the sound will have to travel before it is reflected, and so the time intervals between the individual early reflections will be significantly longer than in a small room. After a short time, the reflections will become too dense and chaotic to perceive individually, but those first early reflections immediately following the start of the sound provide the ear/brain with strong clues as to what type of space the sound is being heard in.

Leaving aside for a moment reflections from the floor, no early reflections are heard until the sound has reached the nearest wall or obstacle and reflected back to the listener. This, initial delay between the direct sound and the first reflected sound, provides what is perhaps the strongest clue as to the room size, and if the reflection returns as a discrete echo, it suggests that the reflective surface is both solid and flat. A more diffuse echo (one with less pronounced individual reflections) suggests irregular surfaces. Figure 1 shows the evolution of reverberation in a hypothetical room following a single percussive event.

Directivity In Reverberation

The reason why spaces sound spacious is all to do with our surround hearing abilities. The sound source may originate from a single point (mono), but the reflection patterns arriving at each ear will be slightly different because the ears are in slightly different positions within the room and are pointing in different directions. Normally, a reverb unit emulates this by creating a slightly different pattern of early reflections and late reverberation for the left and right channels. The fact that they are slightly different is enough to fool the brain into perceiving space, which is why the majority of reverbs work perfectly well as mono-in, stereo-out. Even in a real room, the reverb pattern becomes chaotic so quickly that it can be argued there's little to gain (at least in a stereo situation) by religiously ray-tracing the early reflections based on the geometry of a real room, although TC's System 6000 processor does something along these lines to enhance positioning in 5.1 surround mixes. The 6000 also recalculates the reflection patterns based on the position of the sound source and the listener within the imaginary room, which again helps reinforce the psychoacoustic impression of position within a surround mix.

Absorption Of Sound Energy

All materials reflect sound more efficiently at some frequencies than at others and, in most cases, high frequencies tend to be absorbed more readily than low frequencies. This produces a longer reverberation time at low frequencies than at high frequencies. In very large spaces, the effect of the air absorbing high frequencies may exaggerate this tendency, giving the reverb tail a rolling or thunder-like quality.

From this, we can deduce that, to create a natural-sounding reverb, the reverberant sound will probably need to be low-pass filtered to remove the top end that doesn't exist in nature. Furthermore, the high-frequency content will tend to reduce during the course of the reverb decay. In practice, this filtering is applied within the recirculating delays that create the reverb tail as well as to the processed signal as a whole. Only special (not based on nature) reverb effects or the simulation of unusual environments such as tiled rooms require reflections with a significant amount of high-frequency energy.

Artificial Reverb

Recording studios originally used live rooms, springs or plate reverbs to add artificial reverb to recordings, but today the digital reverb unit is used almost everywhere. However, digital reverb is actually quite complicated to design effectively and there's almost as much art as science behind the more successful models. Somewhere between 1000 and 3000 separate echoes are needed every second to create a believable sensation of natural reverberation, and the spacing between these reflections has to be as random as possible or the resulting reverberation will ring in a most unnatural fashion.

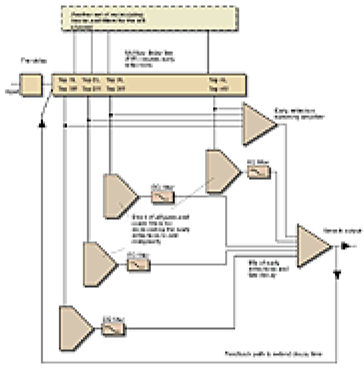


Figure 2. A possible reverb algorithm. Note, however, that each individual designer combines the basic elements in different ways to come up with their unique solution.

Strong early-reflections patterns can be created by using a multitap delay line where the tap levels tend to get quieter the longer the delay time. These reflections provide the initial clues as to room size and character, but in real life they're not just straight echoes — their frequency content is modified by the surfaces they encounter and they will also be diffused to some extent so that each reflection is actually a cluster of reflections rather than a single event. How well this can be achieved digitally depends on the processing power available, which is why the very clear reflections produced by some cheaper units or plug-ins can sound unnatural. In many situations, the effect of diffusion on the early reflections is simply to colour them rather than to add any obvious time-domain artifacts — when multiple similar signals are added, the phase differences cause effects like comb filtering that change the frequency content of the sound. The same is probably true of floor reflections. Lexicon are one company who have looked into this aspect of reverberation with greater depth than most, which is why their recreations of smaller rooms sound particularly convincing.

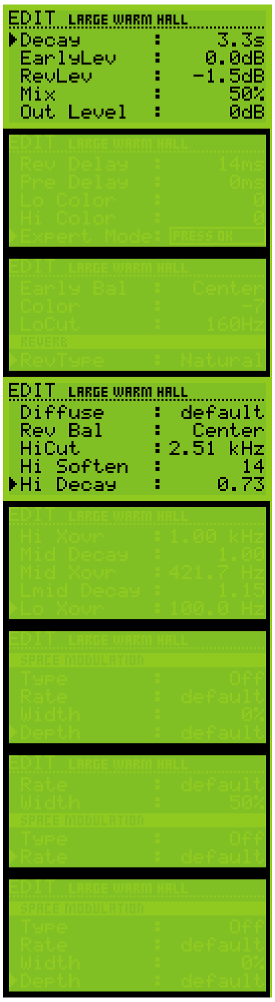
What happens next depends on who designed the reverb unit. Most models take the early reflections and then feed them back into a series of recirculating delays and filters to simulate the complexity of the later stages of the reverb tail as shown in Figure 2. Setting the recirculating delays to produce a reasonably random energy density in the reverb tail, without the reverb taking on a metallic ringing characteristic, is quite difficult and involves a lot of fine tuning.

Some companies, most notably Lexicon and TC Electronic, use a completely separate process for generating the early reflections and the dense reverb that follows them. Though this doesn't mirror the way reverb is generated in real life, it tends to produce more believable results and also allows the user more freedom when adjusting the parameters, as the early reflections and later reverb are not so closely linked.

Key Reverb Parameters

The important reverb parameters generally placed under user control are: early reflection pattern, pre-delay time, overall decay time and high-frequency damping. There are numerous others that may or may not be accessible to the user, but these are the important ones common to most programmable units.

Early reflection patterns are created by the designers to emulate plates, halls, chambers, tiled rooms and so on, and though their overall duration may be adjustable via some kind of room size parameter, the pattern itself is fixed. The greater the spacing of the early reflections, the bigger the virtual room sounds. The rate at which early reflections build up differs depending on the position of the listener within the hall, so some manufacturers provide a number of positioning options. As a rule, a listener at the back of a concert hall will hear the early reflections building up more slowly than a listener near the front.



These screenshots show the range of effect editing parameters within TC Electronic's high-end M3000 reverb unit, and each of these actually controls numerous internal algorithm parameters. As you can see, generating realistic reverb is a very complex task!

Pre-delay also affects the subjective room size and is simply a delay between the original sound and the onset of the early reflections. This is a simple way of creating an illusion of room size and also helps to separate the dry sound from the reverb, though remember that in a real room the sound will also be coloured by reflections from the floor which may arrive back at the listening position a long time before the reflections from the side walls. However, our ears seem more highly tuned to lateral reflections, which is perhaps one reason we so readily accept these simplified aural illusions. Longer pre-delays can also be useful in conjunction with vocal reverb treatments to prevent the reverb detracting from the clarity and impact of the vocals.

A long overall reverb decay time can be suggestive of large environments, but much depends on the early reflections that preceded them. For example, a very reflective, small tiled room may have almost as long a decay time as a large hall, but the nature of the early reflections and the brightness of the following reverb tail are what give us the clues as to the room's actual size. In my view, the effectiveness with which a reverb unit can emulate a small room that has no obvious reverb time component is what separates the passable units from the really good ones.

High-frequency damping allows the high frequency decay time of the reverb tail to be made shorter than the overall decay time. This emulates the way the materials in real rooms absorb sound, though some units also have independent control over low-frequency damping, either for the creation of special effects or to simulate environments that reflect mainly high-frequency sounds. For example, a wood-panelled hall may reflect mid-range and high frequencies quite effectively while bass-trapping lower frequencies, so to get that wood-panelled sound, you'd need to dial in a little low-frequency damping as well as enough high-frequency damping to replicate absorption by curtains, soft furnishings and the air itself.

By selecting the appropriate early reflections pattern for the environment you wish to simulate, then adjusting the other parameters, the available reverb range is astonishing.

Indeed, some of the small booth settings I've tried on better reverb units actually sound deader than the signal you feed into them in the first place! This is almost entirely due to the way such small rooms colour (filter) the sound. And we all know what a large hall or cathedral reverb patch sounds like, even if there are few musical applications where such extravagant effects makes sense. Most of the useful patches lie somewhere in between these two extremes.

Most advanced reverb algorithms incorporate a room size adjustment that actually changes quite a large number of hidden parameters simultaneously. In fact there are so many parameters attached to a typical reverb algorithm that it would be quite unproductive to give the user access to all of them and, indeed, it's probably only the designer who understands what they all do.

In-depth Reverb Tweaks

Over and above the basic parameters already mentioned, some machines may include a further layer of user adjustment. For example, we talk about high- and low-frequency damping, but who decides where the high or low frequencies start? The answer is that the algorithm incorporates a frequency crossover point between high and low that may or may not be accessible to the user. Where it is, the crossover frequency can be adjusted, which will in turn alter the 'colour' of the reverb tail.

Reverb density is another frequently available parameter, and though some manufactures use slightly different descriptions, density generally relates to the density of the reflections making up the reverb component of the sound. The more tightly packed the individual reflections, the higher the reverb density. Lower densities can produce coarse-sounding reverbs on percussive sounds but are often flattering to vocals and other non-percussive sounds. High-density reverbs tend to sound more natural on drums and percussion.

Related to density is diffusion, which determines the rate at which the reflections increase in density after the original sound. A large, square room with flat surfaces might exhibit a relatively low diffusion rate compared with a similarly sized room covered in irregularly shaped surfaces, which is one reason for concert halls being built with pillars and ornate alcoves. It may also be possible to control the shape of the reverb decay curve as, in a number of real spaces, the decay isn't necessarily a simple exponential curve. Most acousticians believe that decay shapes are very important and that many good-sounding spaces exhibit a double decay characteristic separated by a short plateaux as shown in Figure 3. On those reverb units that generate the early and late reflections by different methods, it may also be possible to delay the late reverberation with respect to the early reflections, which goes some way towards changing the decay shape. It may also be possible to adjust the rate at which the late reverberation tail builds up.

Sampling Reverbs

The most real-sounding space is a real space, which is why companies like Sony and Yamaha have developed so-called sampling reverbs. Put very simply, they work by playing a test signal in a real room and then analysing the result picked up by microphones. The information gathered is used to provide an accurate model of the reverb in that room. The underlying theory is that if you were to take a single sample of audio (effectively a click one sample in duration) and somehow play that into a real room, then look at the level of all the samples in the reverb following it, you'd be able to build up an algorithm to generate that information for each successive sample of a real signal. It would be scaled to the amplitude of each sample of the input signal and the theory is that the result would perfectly reconstruct the reverberation character of the room for that one source and listener position.

In reality, single samples are too short to use, so special test signals have to be developed, but listening tests confirm that this approach to reverberation works. The main limitation of the system is that there's little scope to adjust the reverb sound once the algorithm has been created, because the algorithm defines a particular room — it does not have a set of adjustable parameters like a conventional synthesized reverb.

Sounds Different

Reverb algorithms from different manufacturers sound different for a number of reasons, most of which can be traced back to the available processing power and the designer's approach to algorithm design. Most reverb algorithms incorporate a complex network of delay-based resonators to increase the complexity of the reverb tail, but unless these are set up very carefully, different frequencies decay at radically different rates, giving rise to a metallic ringing effect. This is fine if you're after the coloration of a plate reverb, but less welcome in a natural room simulation. More processing power allows designers to use more building blocks, but the fine tuning 'art' side of the process is still critically important.

One easy test anyone can do to help establish the quality of a reverb unit is to increase the mix of processed to direct sound and then to listen to the result. With a well-designed unit, the character and presence of the original sound is still maintained, even when you're listening to 100 percent processed signal, but with some less sophisticated reverb units the 'wet-only' signal sounds somewhat disembodied. Also play a few solo snare hits through the unit and listen for unnatural metallic ringing on the room and hall presets. Tiled rooms and plates, on the other hand, should include some element of ringing to emulate what happens with the real thing. I'm also wary of long reverbs where the reverb tail exhibits a cyclic fluttering effect lasting between 0.5 and 2 seconds. The decay of a large room isn't perfectly even, but close your eyes and check that the simulated version is at least believable.

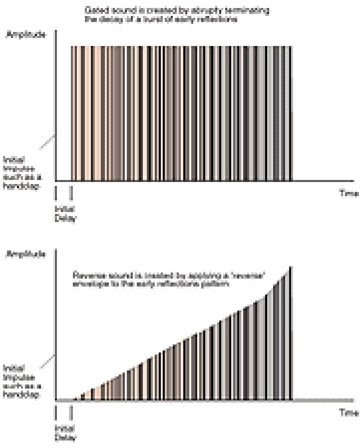


Figure 3. The creation of gated reverb and reverse reverb effects using bursts of early reflections.

One Lexicon development that's now used in different guises by different manufacturers is the inclusion of Spin and Wander parameters. In effect, these modulate the delay times of the recirculating delays within the resonator network so as to prevent static patterns building up that might cause ringing. In physical terms, this is akin to continually moving the walls and objects in a room over a short distance. Providing the effect is kept subtle, it works very well in producing a richer, smoother reverb. Of course, any delay modulation involves pitch modulation too, so Spin and Wander are best avoided when processing instruments such as classical piano where the listener doesn't expect to hear pitch-shift. The effect isn't nearly so pronounced as chorus, because all these little modulations are taking place inside the reverb network rather than all before it or all after it, but some shimmer can be detected on fixed pitch sounds. Sustained sounds can also show up these modulations — if a long sustained tone is played into a reverberant space, the resultant reflections will eventually reach a steady state such that the original sound plus the reflections produces a single, unchanging waveform. If modulation is being applied somewhere, this will never happen — though it may take a trained ear to notice it.

Gated & Reverse Reverb

Originally, gated reverb relied on miking the ambience of a live room, often heavily compressed, then using a noise gate to produce a sharp cutoff rather than a normal reverb decay. The outcome is a burst of reverb following the initial sound, that ends quite abruptly.

Most digital reverbs now offer an emulation of this classic effect as a separate algorithm, and though it was somewhat over-used in the '80s it is still a useful treatment for making drums sound more powerful or for giving electric guitars a raw edge. The main parameter is gate time, though some algorithms also include a trigger threshold below which no effect is produced. Threshold-based effects tend to process conventional reverb via a gate triggered by the input signal, whereas non-threshold dependent treatments tend to apply a group of early reflections that are nominally at a fixed level rather than allowing them to decay over time as they would in nature.

Reverse reverb is a modification of gated reverb where a reverse-type envelope (slow attack, fast decay) is applied to the early reflections cluster. Like gated reverb, the main parameter is the time taken for the reverb to build up and cut off. In use, the effect gives the sound being processed a certain backwards quality, even though nothing is actually being reversed.

Next month I'll be looking at real-life applications of reverb when mixing.

Glossary

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