REVERB FROM FIRST PRINCIPLES

Many of today's digital effects processors offer you considerable control over the creation of artificial ambiences for your music, and if you know how reverberation works in real spaces, you'll be better equipped for designing fake ones. HUGH ROBJOHNS boldly goes...

Reverberation is something that few people are consciously aware of, yet it is one of the most fundamental aspects of a room's sound character. If you were to blindfold someone, take them to an unfamiliar building and lead them through a succession of rooms, clapping or shouting in each one, they would almost certainly be able to give a pretty accurate description of the size of each room. If they were being particularly perceptive, they might even be able to suggest where they were standing in each room and probably even give some idea of what there was in terms of wall coverings, curtains, soft furnishings and so on! In other words, it is reverberation that gives your brain most of the information it needs to create an aural picture of your immediate environment.

Every room has its own sound or 'acoustic', and part of the job of a recording engineer is to assess whether a room's characteristic sound is worth using in a recording. If the dimensions, layout and fabric of the room enhance the recorded sound quality, all well and good, but if not, the microphone technique used should minimise any room sound so that an artificial acoustic can be added later from a reverb processor.

In this age of digital technology, artificial reverberation is not only more affordable than ever before, but can also be stunningly realistic and very controllable. With a good understanding of the physics of natural reverberation, and the fundamental operational principles of reverb processors, it is possible to quickly create the illusion of any acoustic environment you can imagine.

Remember, though, that if you choose to use artificial reverb, it is essential that the recording has the absolute minimum of the recording venue's room sound. If the original environment can be heard, adding extra 'fake' reverbs will just result in a cluttered sound and the mix will often become confused and indistinct.

TIMING

To understand what a reverberant sound actually is and what information our hearing system is able to extract from it, you need to think about how sound waves travel and what happens when they encounter various surfaces.

The first thing to consider is how fast sound travels in air. My old school physics books said 760mph and the new ones probably say 340 metres per second, but I find it hard to relate these numbers to anything meaningful (other than the average speed of traffic around the Evesham bypass in the morning....).

A much more useful figure to tuck away in the dark and dusty recesses of your mind is that sound travels roughly a foot each millisecond. Assuming that you're not too young to relate to feet as a valid dimensional measurement, this rule of thumb will allow you to calculate and set one of the most critical parameters of any reverb processor. One quick side note: the speed of sound varies with the condition of the air. Temperature, humidity and pressure (ie. altitude) all have significant effects on the speed of sound, and in certain applications the 1ft/ms guide is not sufficiently accurate. However, as far as dialling up room sounds on a reverb unit is concerned, it's close enough. Imagine that you're standing in the middle of a very large, brick-walled barn with a deep covering of straw on the ground. You have a spontaneous urge to clap your hands: what happens? Well, the very first thing you'll hear is the direct sound of your hand clap, and it's this direct sound that the brain uses to pinpoint the direction of the sound source.

Assuming that the hand-clap radiates sound waves in all directions simultaneously, the next thing you'll hear will be reflected sound from the various room boundaries or nearby objects. As the floor of this imaginary barn is covered with a deep layer of straw, there will be no significant reflection from the ground (although in practice this is often a dominant source), so the first 'room sounds' will be reflections from the side walls and ceiling. If the barn measures 40ft by 60ft and is 20ft high, and assuming we're standing in the middle of it, the very first reflection will come from the roof, after about 30 milliseconds.

This time delay can be estimated by working out the distance the sound wave has to travel -- if you're a man standing up in the barn, your hands and ears are likely to be about four and six feet respectively above the floor, so (20-4) + (20-6) = 30. Sound travels roughly 1 foot a millisecond, so it will take 30ms for the sound of the clap to reach the ceiling and return to your ears. Similar calculations reveal that the side-wall reflections occur 40 and 60ms after the initial direct sound.

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The time taken for these first reflections to be heard goes a long way towards defining the perceived acoustic size of the room. Short delays imply small rooms, and long delays large rooms. To mimic this natural characteristic, artificial reverberation units normally allow the user to set the time delay between the direct sound and the very first reflection with a parameter called Pre-Delay. This is generally adjustable in millisecond increments (or finer) over an astonishingly wide range.

So the first reflection comes from the closest surface, and is followed by others from the adjacent room boundaries, the whole ensemble being known as Early Reflections. The timing, relative amplitudes and timbre of these individual reflections are determined by three things: the placing, angles and nature of reflective surfaces; the location of the sound source; and the position of the listener. Moving any of these will change the relative timing of the early reflections quite significantly, but our hearing system is remarkably good at extracting even the most subtle information. Consequently, these initial reflections and their relative timing are very important in defining an imaginary acoustic space.

The better reverb units allow the user to alter not only the value of the pre-delay, but also the number, grouping, timing, amplitudes and tonal qualities of the first reflections. In some cases, these parameters are preset by the manufacturer and are simply selected from a list of programmed options, such as Hall, Chamber, or Plate, although occasionally they are derived from measurements taken in genuine acoustic environments. The most sophisticated machines allow the user to specify the cubic volume of the imaginary room, or even its precise dimensions, together with the source and the listening positions!

TAILS

So first we hear the direct sound; then, a short time later, a number of discrete reflections return from the various surfaces in the room. However, these reflections don't just stop when they reach the listener -- they continue until they reach other surfaces, where they instigate more reflections. These reflections start even more reflections and the sound density becomes too great to allow us to distinguish the reflections as separate events.

At each reflective surface, some of the sound energy is absorbed, and more is lost as the sound travels through the air -- which is why reverberation gradually dies away. This reverberation 'tail' may last for anything from, say, 0.3 seconds (for a dead-sounding room), through to several seconds in a church or big concert hall.

The length of the reverberation tail is usually specified in terms of its 'RT60'. This is defined as the time taken for the reverberation to fall by 60dB in level below the original direct sound. Every reverb unit allows this time to be adjusted, normally through a parameter called Decay Time.

It's important to note that the reverberation tail lasts for different durations at different frequencies. High-frequency sound waves have a lot of trouble persuading air molecules to vibrate quickly enough to pass the sound energy onwards. Consequently, high-frequency sounds tend to die away, as they travel, much faster than mid-frequency sounds. On top of that, high-frequency sounds are absorbed by soft furnishings (which includes people and even wallpaper!). On the other hand, high frequencies reflect strongly from a wide range of surfaces, such as windows, sound desks, equipment racks, and so on. At the other frequency extreme, low frequencies are only reflected by large and very solid objects, so there may be little LF in the reverberation at all in some circumstances, but a definite bass 'bloom' in cave-like rooms!

To help provide this level of realism, most reverb units allow you to adjust the reverberation time for high (and sometimes low) frequencies relative to middle frequencies, and introduce some kind of overall equalisation to the reverberation tail.

KNOBS

Let's recap on the parameters that today's digital reverb processors are likely to offer for the simulation of real acoustic spaces. Firstly, although I haven't previously mentioned it, there is usually a means of balancing the direct sound against the reverb. The direct sound is often referred to as 'dry' and the reverberation as 'wet', so a wet/dry control will probably be in there somewhere. Some of you will know how unpleasant it sounds when an analogue-to-digital (A/D) converter is overloaded, so a critical control on digital reverbs is the input level control, and its associated headroom meter.

The first control which defines the reverb character is pre-delay, which effectively defines the distance of the first reflective surface. This will be followed by one or more parameters for controlling the number, timing, amplitude, and timbre of the other early reflections. Some machines provide controls called 'Pattern', 'Level' and 'Room Size'; others might simply offer preset venue simulations ('Hall', 'Chamber', 'Jazz Club', and so on).

After the early reflections, the reverberation tail is set by a control for overall decay time. This is normally accompanied by a parameter that adjusts the relative decay time at high (and perhaps) low frequencies. There's usually also some means of setting overall tonal characteristics, although this may be little more than a simple bass and treble equaliser.

Finally, having skilfully set all of these parameters to create a wonderfully believable artificial acoustic(!), you can usually store your fake room in one of a number of user memories.

CREATING SPACES

You cannot create an artificial acoustic space if you don't know what a real one sounds like to start with, and the only way you can find out is to actively listen to sounds in as wide a range of environments and circumstances as possible. Everyone has a very detailed subconscious knowledge of how different rooms sound, and although few are able to analyse the reverberation structure, most spot incongruities in artificial reverberation very easily.

It's very revealing and informative to consciously listen to the sound of different rooms as you move around in a building -- even in places with which you thought you were familiar. Try to analyse in your own mind what sort of pre-delay, decay time, early reflections and high-/low-frequency decays naturally occur to create the 'sound' of that room. Don't just listen to indoor reverberations, either -- try to assess the reverberant features of the local high street, the great outdoors, a wood or forest, or wherever you happen to be. You will find reverberation in places you didn't expect it, and may be surprised to discover that places you assumed to be reverberant actually are not!

MONO AND STEREO

The very nature of genuine reverberation is that it tends to come at you from all over the place, but particularly from the sides of the room. This has significant effects on the compatibility between stereo and mono versions of your mix, since the mono listener is effectively denied any information from the sides of the stereo image.

To see how this happens, consider a simple M&S (Middle & Side) stereo microphone technique being used to record something in a reverberant room: the stereo listener hears the full acoustic in all its glory, but the mono listener hears only the forward-facing 'M' microphone, not the sideways-facing 'S' microphone -- and guess which one picks up the bulk of the room sound? This absence of reverberation in mono afflicts artificial reverb processors as well as natural acoustics. In practice, the amount of reverberation heard in mono may be substantially less than that in the stereo balance, and if mono listeners are likely to be an important part of your music audience, always check for mono compatibility. In general, you almost always have to compromise the balance in some way because either the mono will be too dry, or the stereo will be too wet!

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Tricks worth trying include reducing the stereo width of the reverb (turn the pan-pots in a bit towards the centre instead of having the reverb returns running out to full left and right), or mix in a small amount of reverb from another reverb processor, panned centrally. The extra reverb should be set up with the same parameter values as the stereo reverb, although a slightly shorter pre-delay and longer HF decay time often work well. The balance between the dry sound, the mono reverb, and the stereo reverb needs to be adjusted carefully, while you continuously switch between mono and stereo listening to find the most uniform results in the two modes.

In matrix surround systems (such as Dolby Surround), real or artificial stereo reverb tends to spread across the rear channel quite naturally as a result of the way in which the rear-channel information is encoded and decoded. Altering its stereo width controls the front-back balance, narrowing the reverb pulls it to the front, and increasing the width pushes more to the sides and rear.

Many stereo digital reverb units have a single input and a stereo output, and this often causes people to wonder how the reverberation can be 'true stereo' with only a mono input. The answer is simple if you consider the real situation of a sound source within a reverberant space.

If someone claps, there's only one sound source, yet the reverberation will come from all directions and could be captured by a simple stereo microphone array -- a mono input to the room and a stereo output from it. Of course, in a more complex situation with, say, a string quartet in the room, there are multiple sound sources and each will have slightly different pre-delays and early reflection patterns, but this is usually a very subtle distinction, and in practice the mono-in, stereo-out system of most digital reverb units works perfectly adequately.
Something few people ever check is the line-up of a stereo reverb unit. However, it is a stereo source and should be treated in just the same way as any other stereo signal, which means making sure that the left and right reverb outputs have the same gain and equalisation through your mixer. I find that a quick, easy and reliable method of doing this is to simply dial up a 3- or 4-second decay time and send a brief burst of signal into the machine. Listen carefully to the dying reverb tail: it should decay centrally, possibly even becoming narrower in width as it goes (although this depends on the particular algorithm). If the reverb tail appears to collapse towards one side or the other, your return channels have different gains and should be adjusted.

CHOOSING AND USING REVERB

In general, two reverb units will meet the needs of pretty much every recording situation. One machine would normally be set for a short, bright sound (perhaps a plate setting) for percussive sounds, whilst the other would be set to a longer, warmer patch, providing a 'lush' quality for vocals and solo instruments. You could also try passing some instruments through both reverbs (percussive one first) for a third alternative.

Some engineers like to use several reverbs to create a layering effect, but I generally find that this approach causes a loss of definition and adds confusion to the overall sound. Going back to the idea that the artificial reverb is merely replacing the poor acoustics of a less-than-ideal recording venue, it could be argued that there should only be one reverberation sound for everything, as would be the case if the musicians all played live in the same reverberant room!

The next issue to address is how much reverb to use. The classic mistakes of the novice are using too much reverb return on everything and allowing reverb tails to be too long. Reverb generally needs to be subtle, and ideally only the loudest musical peaks should cause obvious tails. Even the biggest halls rarely have a reverb time in excess of four seconds, and often a two-second decay time is easily long enough.

The choice of reverb parameters is dependent on both personal taste and the nature of the programme material, so it's impossible to give specific recommendations, but try to create life-like environments wherever possible. Most reverb units offer a number of special effects, such as gated or reverse reverbs, and these are best used sparingly, so that they keep their impact. While we're on the subject of special effects, it's worth trying out the pseudo-reverberation programs too. Algorithms such as 'Ambience' or 'Alive' can often add extra definition and life to dull vocals, or spice up closely-mic'd solo instruments without your having to resort to using those horrible exciters (just a personal opinion, of course...).

Normally, reverb sends are taken post-fader, so that direct signal level adjustments are reflected in their reverb returns. However, it can often be useful to send pre-fader, and not allow any direct signal into the final mix at all. This is particularly effective with sustained keyboard string sounds and the like, where it helps to make less-than-ideal synth sounds blend a lot more smoothly.

Another useful trick is to set up a reverb specifically for the keyboard sounds, and route the reverb returns through a chorus unit. This provides a completely different kind of sound to chorusing the keyboards directly and adds an interesting 'swirling' quality which can be very effective if used discreetly.

Although reverb processors are most important during mixdown, they're also vital during recording, especially when recording vocals. Many singers have enormous trouble pitching properly without reverberation and it's essential to have the ability to route reverb returns to the headphone monitor mix. The reverb setting for the cue monitor is not particularly critical to the performance (provided it is broadly appropriate) and need not be recorded, although some engineers do like to record voice and reverb together (occasionally as a complete mix but more usually on adjacent channels on the multitrack machine). This is particularly useful if the reverb plays a part in the performance (through timing or percussive vocal effects, for example).

As artificial reverb becomes more and more elaborate, there's a trap which many engineers find themselves falling into. It's possible to become so engrossed in adjusting each parameter minutely, trying every possible combination along the way, that you lose sight of the original idea. The best way of getting the sound you want, quickly, is to understand the nature of real reverberation and apply that knowledge to creating the acoustic space you've imagined. It's far better to think for a minute or two, and then dial the right numbers in, than to sequentially try every preset on the machine, hoping to stumble across something that sounds OK.

ONE ALGORITHM OR TWO?

A word of warning -- not every reverb processor is as flexible as it might seem. Particularly with multi-effects units, it is quite common to find that there is actually only one reverberation algorithm. The wide range of supplied preset environments (Hall, Room, Plate, and so on) is actually composed of variations in the delay, decay and EQ settings of a single algorithm. In these cases, you'll find that no matter how you adjust the reverb parameters, all settings sound very similar: the overall character of the room does not seem to change, and this is because the pattern of the early reflections remains fixed. The better machines have a number of different algorithms and a variety of early-reflection patterns, which allow a larger range of different room types to be created, each with distinct and individual sonic characters. Fortunately, there is an easy way to find out which category a particular machine falls into. Select two, theoretically diverse, programs -- perhaps a Hall and Plate. Set the delay, decay, EQ and any other parameters to identical values and store the new settings in a couple of user memories so that they can be recalled easily. Next, listen critically to the quality of the reverberation while switching between the two presets. There should be an obvious difference in the character of the room acoustic if the machine uses different algorithms, with different early reflection patterns. (Try closing your eyes and imagining the dimensions and furnishings of the fake room.) If you cannot spot any differences, the chances are that the machine uses the same algorithm for all its reverb programs.

THE MUSEUM OF ARTIFICIAL REVERBERATION

Artificial reverb has been around for more or less as long as people have been performing in non-ideal acoustic environments. In more recent times, however, various electronic and electro-mechanical methods have been developed, although none was as effective as the current generation of digital designs.

One of the simplest and most obvious systems was the echo room -- literally a room, often tiled and full of ceramic sewer pipes to provide a wealth of reflective surfaces. A loudspeaker generated the direct sound, and one or more microphones collected the resulting reverberation. The echo room has the advantage that the reverb is naturally very complex, but it is also difficult to adjust, and requires a large and quiet room!.

One of the first electro-mechanical systems (and one which remains popular to this day) is the plate. This employs a large sheet of metal (typically 6ft by 4ft) suspended on springs within a sound-deadening case as a reverberant space. A vibrating transducer feeds the direct sound into the metal plate and a pair of pick-ups extract the reverberation as the vibrations bounce off the plate's edges. A motorised damping plate parallel to the main one can be remotely positioned at varying distances to control the duration of the reverb. The plate has a characteristic metallic, bright, sound quality which has become intimately associated with pop music. Virtually every digital reverb I have used includes a simulation of the humble plate -- which is a very good indicator of just how popular this mechanical system remains!

Another enormously popular electro-mechanical system is the spring-line reverb. This technique has been around for a very long time, and I'm sure everyone has come across guitar amps with spring reverbs installed. The operating principle is similar to the plate, in that a transducer sets up vibrations in a spring, which rattle back and forth, to be extracted by a pick-up at the other end. The character of a particular spring reverb unit is fixed (other than the wet/dry balance), but can be optimised for the sound source at the design stage by careful choice of the number, length, diameter and compliance of the spring(s).

All manner of record-replay systems have been developed to provide a reverb effect, but none have survived the digital revolution. The earliest ideas simply used a three-head tape machine, where the direct signal was recorded onto tape and the replay signal provided the reverberation. The tape speed and head spacing determined the pre-delay and if some of the replay signal was mixed with the direct signal, a pseudo-reverb could be created. The results are hardly realistic, but the system was popular at a time when the alternatives were too expensive or impractical.

The record-replay theme was further developed into machines like the WEM Copycat and the Roland Space Echo, which used tape loops and multiple replay heads, with the ability to adjust the contribution and feedback of each head -- but then solid-state technology arrived...

Bucket Brigade systems became popular for a brief time (fortunately) just before the first true digital reverb machines hit the market. Bucket Brigade delays were a halfway house between analogue and digital systems, but were no more realistic or flexible than the earlier tape-loop products -- and were often a lot noisier!

The advent of digital technology really revolutionised artificial reverb, basically because the time-domain signal processing of digital audio lends itself very well to the kind of sound manipulation needed to create realistic reverb. Creating a pre-delay is simply a case of storing sound in a memory until the required time has passed. The early reflections are created by replaying the direct sound repeatedly at suitable moments, with level and equalisation changes as necessary. The main body of the decay is created by cycling the direct sound through a complex set of short feed-back and feed-forward delays, configured to introduce the desired equalisation characteristics.

Digital reverbs are available to suit all pockets from a large number of manufacturers, including Yamaha, Sony, Digitech, Klark Teknik, ART, Ensoniq and Alesis, to name but a few. However, if you ask any professional sound engineer to name their favourite reverb machine, chances are you'll hear one name above all others. Lexicon are probably the most popular manufacturer of digital reverbs, and their product line extends from push-button preset units to the most complex state-of-the art systems.