a little cheat sheet on Signal Processing

by Chris Kubick for my friends in sound art 173

Gain:

Gain is often applied to make a sound louder or, more accurately, to raise the level of the signal. If a signal is too soft after you turn the volume on the track all the way up, it might be a good time to use a Gain plug-in. Normalizing plugins are another version of this concept: they work by figuring out the difference between the loudest part of a selected audio clip and a desired loudness (eg -3 db), and raising the signal to the desired loudness. Settings are very basic on these devices, which are basically a lot like volume knobs.

Compression:

Compression is used to tighten the dynamic range of a signal. If you have an audio cllip which peaks at -6 db but has very quiet places, too, which are say -90 db, then it has a wide dynamic range. You might use a compressor to make the softer parts louder and the louder parts softer. Compression is often used to fatten a drum sound, or to make a speaking voice more consistent in level. Because compression used with normalization has the effect of making something seem louder , compression is often over-used; on tv and radio many people think louder is better. Because of this, many discerning ears believe that the indescriminate use of compression is destroying modern recording. Here, even more than with many other effects, a little goes a long ways....

Settings:

Usual settings on a compressor include:

Threshold: this is level at which the compressor kicks in. eg at a setting of -12db, any signal above this threshold will be compressed.

Ratio: ratio is basically the ammount of compression which is applied, or, how strongly the signal is squished.

Attack Time: Attack time is the

A Noise Gate is a kind of reverse compressor, which eliminates all audio under a defined threshold .

EQ:

EQ or equalization is used to boost or lower specific frequencies. EQ's may be used for noise r eduction, either by rolling off the high (hiss) or low (rumble) end of a signal or by choosing a s pecifically noisy frequency (eg 60 Hz electrical hum) and turning it down. EQ's may also be use d to tailor a sound, as you do when you adjust the treble and bass on a stereo. The ears know. A hi pass filter means only sounds above a selected point will pass ; the opposite with a lo pass filter. A hi cut filter will remove

frequencies over a selected frequency, and the opposite is true of a lo cut filter. In actual practice, most cut and pass filters have a rounded response, meaning that frequencies are gra dually attenuated on the other side of the cut off point. An exception to this is the brick wall fil ter, which, theoretically, places an exact limit on a cut filter (in practice this is impossible, however, and there is just a less gradual roll off). Extreme effects can be made by using a notch filter, which is a filter which affects a very narrow frequency. Extremely narrow notch filters will create a somewhat unpleasant ringing tone at the chosen frequency.

Settings:

Q: this is the width of affected frequencies.
Frequency: refers to the center of the filter, from which the Q radiates
Gain: oh, that again.

Note: in audacity, as with many EQ's, you can draw the shape of the Q that you desire. this can be pretty fun.

Pitch Shifting:

Pitch Shifting is an effect which, naturally enough, changes pitch.
Settings may be given in terms of musical tones and/or percentage shift.

Time Shifting:

Yep. Used to lengthen or shorten an audio clip.

Note: Pitch Shifting and Time Shifting were originally accomplished naturally by taking a tape and playing it at different speeds. Nowadays, with digital effects, it is possible to create these effects independent of each other, i.e. it is possible to shift a sound down by half and keep it the same length; but, listen carefully! digital artifacts may result. To explore this, compare speed shifting and tempo shifting in the audacity effects menu.

Delay:

Delay basically repeats a signal. You can do this by hand by duplicating an audio. Delay can be used to create phasing effects, when small ammounts of delay are used -- in this case sounds are being repeated, out of phase with each other; many many delays working together, properly tuned can be used to make reverb.....
Settings on delays usually include a delay time , a rate of decay , which affects how the signal will fall off, and often feedback , which will keep repeated sounds repeating at a chosen level/length of time.

Reverb:

Reverb is actually a collection (architecture) of delays which are chained together to provide a sense of space. Churches, concert halls, staircases, showers. Carnegie hall is carnegie hall because it gives such a good reverb sound to music that is played in it. A decent sounding reverb has long been a goal of effects designers; imagine if you could make recordings in Carnegie Hall without actually going there! Of course, the definition of decent is constantly shifting, so that a decent digital reverb from the 80s now sounds absolutely, ridiculously artificial, (and is sometimes just the thing!).....in the same vein, spring reverbs from the 60's (basically a box with a spring or series of springs in it) are highly sought after for their signature sound. For a number of years, digital reverb has been made by combining a number of delays with a number of EQ filters, in an effort to model acoustic spaces. Remember: natural reverb is created because sounds bounce

off of many surfaces, creating a natural and naturally chaotic series of delays.
Recently, effects designers have turned to convolution as the latest holy grail of reverb design. A convolution or Impulse Response reverb works by combining two sets of digital audio data: one set of numbers represents the audio signal (ie a voice); the other is a map of a particular space, which is made by ffiring a cap gun in a room, recording the result, and then digitizing it.

Settings include:

Room Size: kind of self-explanatory
Reverb Time:
Damping: this controls the ammount at which hi frequencies get attenuated or lowered
Wet/Dry mix: as with all effects, this controls the relationship of the original sound to the effected sound
Early Reflection (level and/or number and/or speed): these controls affect the acoustic modeling of the space; basically, by changing this, in combination with EQ settings such as Damping , Bandwidth , Hi Freq Rolloff and the like, you are determining what the virtual room you are constructing is made of.
Tail: this is

Sometimes you can draw the room you want to create; can choose materials hat the room is made of, etc etc.....but the basics remain the same.

Aost signal processing consists of the processes described above, or some combination of them. For in tance, A De-esser is a compressor which uses EQ to focus on a particular range of frequencies, so that the frequency where esses are problematic can be controlled (this frequency is different for different people but is generally centered somewhere in the vicinity of 3500 Hz); a Multi-band Compressor is a compressor combined with Multiple EQ's, and is often used in Mastering to fatten specific frequencies in a mix. Stranger sounding effects might mix a delay with an EQ. You can do that yourself of course, b doing them in succession. By changing the order in which you apply effects you will get very differen results.

Aore advanced signal processing often adds an LFO or Low Frequency Oscillator into the mix. An Osc llator is, basically, a pulse. (All sounds are oscillations, or pulses). A LFO is a pulse which moves at a sl w rate. An LFO may be used to vary a parameter over time. An example of an effect which uses a LFO is Tremolo. A Tremolo effect uses a LFO to vary gain over time. You can experiment with LFO's in aud city using the wahwah effect and the phaser effect.

There are even more interesting effects which involve analyzing a signal and then transforming it base l on an analysis. Examples of this are noise removal, eq matching, FFT transforms.

Have fun, and remember, in most cases, less is more with effects. There's that fine line between using a effect and overusing an effect. Too much of a particular over-used effect (reverb, anyone?) can get cli hed pretty fast.....but then again, that can be a great thing, especially if you really go all out.....

heers,

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