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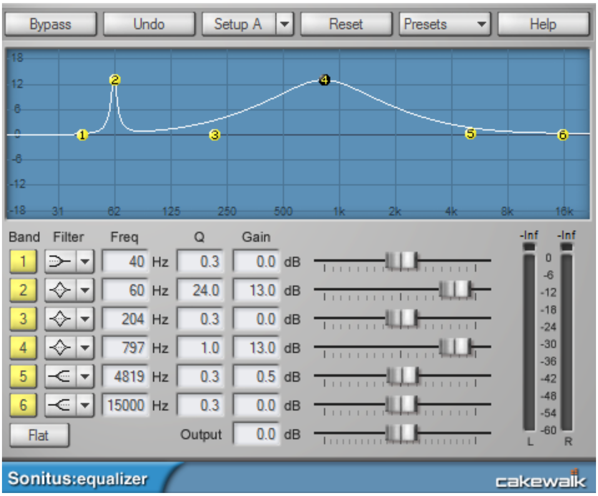
THE KEY PRINCIPLES OF MUSIC TECHNOLOGY

1- Filtering

EQ

EQ or an equaliser is **frequency-based volume control.** This allows you to control the volume of certain frequencies, enabling you to make tracks sound more 'bassy' or 'trebly'.*



This is achieved using 3 main parameters:

- Frequency (Hz/kHz)
- Gain (dB)
- Bandwidth (Q)

The **frequency** parameter spans from 20Hz to 20kHz; considered to be the range of human hearing. By changing this you are selecting the frequency you want to boost or cut.

Gain is simply how much your selected frequency/frequencies are being boosted or cut by, measured in dB.

Bandwidth or **Q** dictates how many frequencies are effected by your gain. E.g: in the picture above, band number '2' is boosting a smaller range of frequencies than band number 4.

<u>**Using and Describing EQ**</u>

*Your use of EQ will fall into two categories: **Correctional** EQ and **Creative** EQ.

<u>Correctional EQ</u>

Correctional EQ is when there are **unwanted frequencies** in your sound source. Sometimes these frequencies can be removed at the recording stage (e.g. low-cut switches on microphones) or they will need to be removed in the mixing stage. These sounds tend to be either very low bass tones muddying your mix, very high ‘hiss’ sounds or unusual resonance from some instruments (snare drums sometimes have ringing notes that you may wish to remove.)

<u>Creative EQ</u>

This is the primary function of EQ. It is where you define how you want your overall mix to sound by **creatively adjusting the tone** of different tracks. E.g. filtering (reducing) the treble frequencies of your acoustic guitar because you want a deeper, mellow sound.

As well as filtering tracks to get the sound you want, it is also vital that you filter your tracks so they are all able to co-exist in your final mix. As you can see in *1.1* all instruments occupy multiple frequencies and even ‘bassy’ instruments such as the cello can dominate the frequency range.

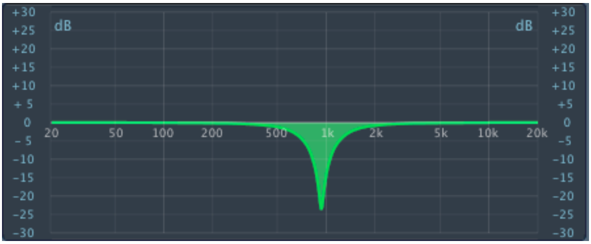
If 2 instruments are both vying for the same frequencies then they tend to get in each other’s way and ‘muddy’ your track. To solve this you need to filter the frequencies that are not important to the sound you want to achieve, and ensure that each instrument has its own space in the frequency range.

More advice on achieving this can be found in the “Mixing and Mastering” chapter.

Describing EQ

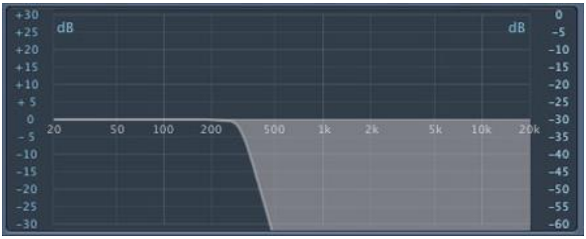
Both in the exam and in your logbook you will need to be able to describe filtering/EQ in the correct way:

A Notch Filter leaves most frequencies unaffected and only cuts or boosts over a small bandwidth.

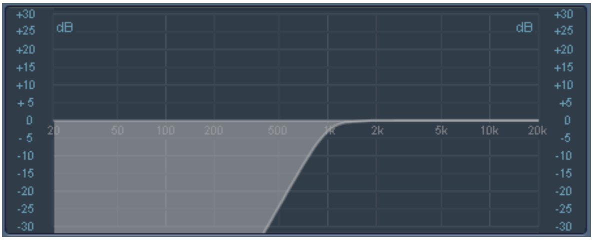


A Shelf Filter cuts or boosts a much wider range of frequencies starting at the **cutoff** and ending at the edge of the frequency range. There are two main types: LPF and HPF.

Low Pass Filter (LPF) is a filter that allows the **low** frequencies to **pass** leading to a bassier sound.



High Pass Filter (HPF) is a filter that allows the **high** frequencies to **pass** leading to a more trebly sound.



Envelope/ADSR *1.2*

If you are asked about the filtering of audio in the exam, you will need to be able to describe the EQ you can hear. If you are asked about the **filtering of a synthesizer** in the exam, you will need to describe the EQ **and the envelope**.

The envelope is the term given to the 4 stages of any sound:

- **Attack**- How quickly the sound reaches full volume after it has been activated (key pushed, string plucked)
- **Decay**- How quickly the sound drops to the **sustain** level after the initial peak.
- **Sustain**- The "constant" volume that the sound takes after **decay** until the **note is released**.
  Note this specifies a volume level rather than a time period.
- **Release**- How quickly the sound fades when a note ends (the key is released).

Physical sounds have their own ‘pre-set’ envelope and can even start the **release** before a note is released. (Think of a piano note ending even with the still key pressed down.)

However when using a **synthesizer** you are able to choose all of these settings to create a ‘unique’ sound. This is regarded as a form of filtering as you are removing aspects of the sound you don’t want; e.g. making a note fade out more quickly.

Describing envelope

When describing the envelope you describe **A**, **D** and **R** in terms of speed (quick/fast or slow), and **S** in terms of volume. (low or high).

For example a note which instantly starts but then takes a long time to finally end (such as a glockenspiel) can be described as having a fast attack and slow release.

*In an exam **A** and **R** tend to be the easiest to identify and describe.*

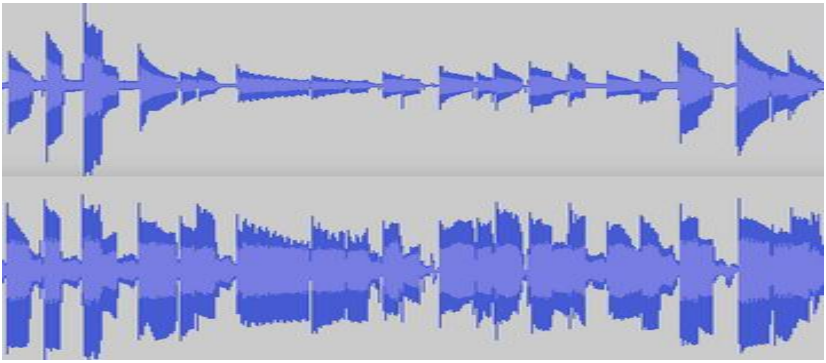
2- Dynamics Processing

Dynamics processing deals with altering the dynamics (quietness and loudness) of your tracks using 3 tools. **Compressor**, **Limiter** and **Gate.**

Compressor *2.1*

[For sidechain compression see 6- "Mixing and Mastering"]

A compressor reduces the **dynamic range** of your tracks. Put simply, it is used to make the loudest parts of your track and the quietest parts of your track sound a similar volume.



Part of an audio track before and then after compression.
Notice the change between the quietest & loudest sections.

This is achieved using 3 main **parameters**.

- Threshold (measured in dB)
- Ratio
- Make-up gain/gain.

And also has additional parameters:

- Attack and Release
- Knee

Threshold

The threshold is measured in dB and is essentially an invisible line applied to your audio track.

It is the point at which you tell your compressor to start working and to reduce the volume of any sounds that go **above** the threshold.

Any sounds **below** the threshold are not compressed.

Ratio

Once a sound has gone above the threshold, it needs to be reduced. The ratio is your way of telling the compressor how much you want the volume reduced by.

The ratio is literally written as a ratio- e.g. 3:1/10:1.

If your ratio is set as 3:1, every 3 decibels that are **over** the threshold get reduced to 1.

See the example below to show how this would change an outputted signal:

Original Signal: 16dB	→	**Compressor** Threshold: 10db Ratio: 3:1	→	**Outputted signal:** 12dB

The original signal of 16dB has gone above the threshold by 6dB.

No sounds below the threshold are compressed which means the first 10dB are untouched.

The 6dB over the threshold are reduced by 3:1 which leaves 2dB over the threshold and a total outputted signal of 12dB.

Make-Up Gain/Gain

Now that your compressor and your ratio are working together, you have essentially created a 'ceiling' of volume that your sounds are unable to get past.

The make-up gain/gain enables you to now turn up the whole track. This will make your quietest sounds louder and your loudest sounds hit the 'ceiling.'

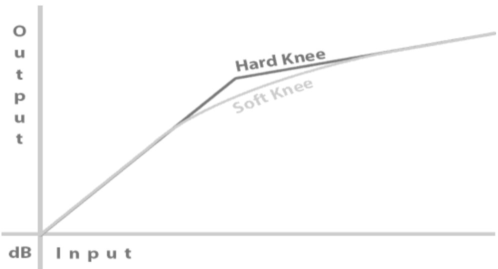
It is this process of having a volume ceiling (threshold & ratio) and a rising floor (make-up gain) that make your quietest sounds and your loudest sounds closer together, and reduces your dynamic range.

Additional Parameters

Attack: Measured in milliseconds. How long the compressor takes to completely reach the desired ratio reduction after the sound has gone **above** the threshold.

Release: Measured in milliseconds. How long the compressor takes to completely stop compressing after the sound has dropped back **below** the threshold.

Knee: (Described as 'soft knee' or 'hard knee') adds a small amount of compression either side of the threshold to 'smoothen' the transition from below to above the threshold.



Attack, release and *knee* are all used to make the compression sound less obvious when a wave passes the threshold.

Limiter

A limiter also reduces the dynamic range of a track using the **same parameters**, and is identical to a compressor in almost every way.

The only difference is that a limiter has the **ratio** and **attack** set to the **highest possible levels**; reducing sounds above the threshold instantly and almost removing them completely.

*N.B. This form of dynamics processing can be very easy to hear even when well executed, so should be used sparingly! Also if you use a limiter, a compressor should always have been used **first**.*

Gate *2.2*

A gate is used to remove or reduce unwanted noise from a track. This is achieved using the following parameters:

- Threshold
- Attack, Release, Hold and Hysteresis
- Depth/Mix/Reduction*

Threshold:

Like the compressor and limiter, the threshold of a **gate** is measured in dB.

However instead of altering sounds above a threshold, a gate acts almost like an upside-down compressor and limiter; reducing or almost removing noise that is **below** the threshold.

This is useful if the **signal** (the sound you wanted to record) is louder than the **noise** (any sounds that you didn't want to record- hiss, talking, etc) as you are able to put your threshold in-between your signal and noise and remove the noise.

Attack, Release, Hold and Hysteresis *2.3/2.4*

A gate can be described as open (allowing the sounds through) or closed (not allowing the sounds through.) When using a gate it is vital to make the transitions between open and closed as smooth and unnoticeable as possible:

Attack: Measured in ms. How long the gate takes to become fully **closed** after a sound has gone **below** the threshold.

Release: Measured in ms. How long the gate takes to become fully **open** after a sound has gone back **above** the threshold.

Hold: Measured in ms. After the gate has closed, how long the gate **stays closed**- even if the sound goes back over the threshold.

Hysteresis: Measured in dB, hysteresis creates a **second threshold** below the main threshold. One to open the gate and another to close it. This means that once a signal has dropped below the close threshold, it has to rise to the open threshold for the gate to open again.

Depth/Mix/Reduction
(Called different things depending on your software)

Usually measured in negative dB (or sometimes a %) this parameter is where you can control how much you are reducing the noise below the threshold by.

E.g. -10dB means that any noise below the threshold will be reduced by 10db.

Alternatively a setting of 50% means that any noise below the threshold will be reduced by half.

3- Effects Processing

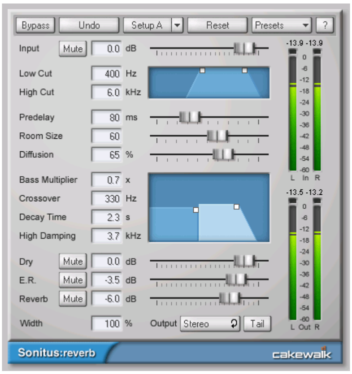
Effects, filtering & dynamics processing cannot all be added to a track at the same time; each has to be applied one after the other. The order this happens is called the **signal process.** Generally the signal process starts with filtering, then dynamics processing and finally you add various effects.

*One famous exception to this is **gated reverb**; a very popular effect applied to 80s snare drums. This effect applied the reverb first, and then gated out the **reverb tail** leading to a very distinctive sound.*

Reverb

Reverb is the persistence of a sound after that sound has been produced. This ‘persistence’ of the sound stems from waves reflecting off of various surfaces; essentially creating **multiple echoes** that are too close together to distinguish.

Most recordings take place in a small live room, sometimes with acoustic treatment to purposefully to reduce the natural reverb of a small room. Therefore to achieve a rich and effective sound you must add reverb artificially.



As you can see above, as with all reverb units and plugins there seems to be a sea of different settings and parameters. In this chapter we will be focusing on the most important.

Wet and Dry

When sounds have reverb, the original sound is known as the **dry signal** and any reverberations are known as the **wet signal**. When listening to and describing reverb you can use these terms as adjectives, to comment on the amount of reverb that has been applied; e.g. the guitar track is very dry.

When adding reverb to your tracks you can control how much **dry signal** and how much **wet signal** can be heard. Often this is done with a single control which mixes the two together and allows you to control whether you would like more wet signal or more dry signal.

In Sonar (SonitusFX) there are two individual controls: **Dry** and **Reverb** both measured in dB that control the respective gain of your dry signal and your wet signal.

Pre-Delay

A massively important parameter on a reverb. The pre-delay is measured in ms and is the amount of time between your dry signal and the start of your reverb (or your **early reflections**.) The pre-delay time is directly proportional to a room's size.

Room Size

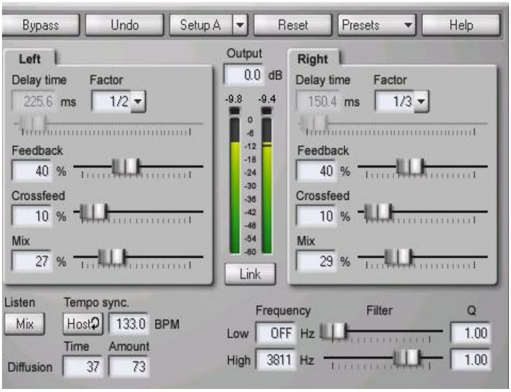
Multiple parameters can make the room sound larger (especially pre-delay and decay time) so **room size** is not the only parameter you should change if you want to make a room sound large. However what room size will change is the pattern and spacing of your different reflections/echoes.

Decay Time

Also sometimes known as reverb time, this parameter dictates how long it takes for the reverberated sound to finally stop by changing the reverb tail.
The larger the room, the longer the reverb lasts for.

Delay/Echo

Delay is any processing where a signal is duplicated and then the duplicate signal plays after the original.



Delay Time (Normally measured in ms)

How long from when the **original signal** plays to when the **duplicated signal** plays (and every delayed signal after that). In Sonar (SonitusFX) this can also be measured as a **factor** of a crotchet- so your delays can be exact musical notes apart.

Feedback Level

How many times the original signal is duplicated before it ends. The higher the level, the more repeats. Setting feedback level to minimum still means you get **one** duplicated signal.

Slapback echo was a very popular effect in 50s Rock n Roll. It uses a delay time of 75-250ms and little to no feedback.

Mix

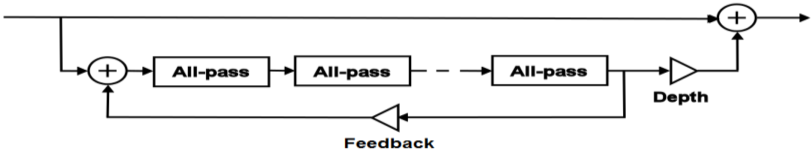
How loud the delayed signal is compared to the original signal.

Also notice how you are able to make changes to both the left and right. Delay/Echo being panned is another important feature.

Flanger and Phaser *3.1*

Flanger and Phaser are two effects that have so many similarities that they are often interchanged. Both create a duplicate of a signal and then play it **alongside** the original signal with a slight delay. The difference between them is in **how** this delay is achieved.

Phaser splits a sound into two paths and one of the paths is sent through multiple **all-pass filters**. Even though an all-pass filter allows all sounds to pass through it, the processing time means that there are fractions of a second delay between the two signals when they play together.



Flanger mixes two identical audio signals together, with one signal delayed by a small, controllable and gradually changing period. Usually smaller than 20ms.

In both the **flanger** and a **phaser**, when the two signals mix the frequencies that are out of 'phase' because of the delay cancel each other out. This causes a distinctive sound similar to a "jet plane woosh" where certain frequencies appear exaggerated.

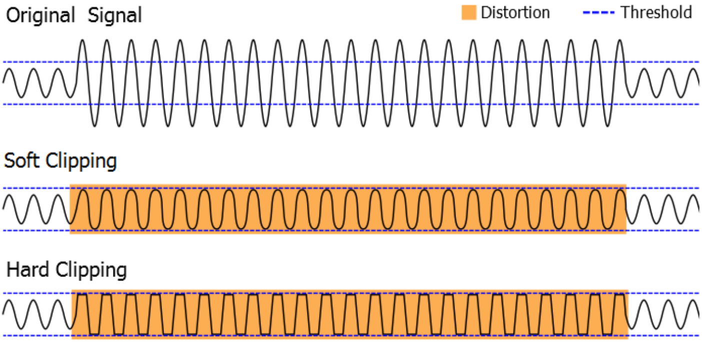
*Because of the increased control you can have on the delay time of a **flanger** compared to a **phaser**, flangers tend to sound more pronounced and natural, like the "jet plane whoosh" effect, whereas phasers tend to sound more subtle and ethereal.*

*NB: **Chorus** is a very similar effect as it also has two matching timbres playing together with a slight delay. However crucially the **pitch** is slightly different. This can be done naturally, like with a 12 string guitar, or artificially using an effects unit that duplicates the audio signal and then detunes it.*

Guitar Effects

Overdrive and Distortion

Overdrive and Distortion are both effects that create harmonic overtones by forcing a signal to ‘clip’. **Clipping** naturally occurs when a signal is too loud for an amplifier and so the wave is flattened. This is either done gently (soft clipping) or very abruptly (hard clipping). Hard clipping sounds ‘harsher’.



Distortion distorts a signal **before** it reaches the amplifier by artificially clipping the sound to emulate natural distortion.

Overdrive forces a sound wave to go above the maximum output of an amplifier. This crates natural clipping of the wave.

Wah-Wah

Wah-wah is an effect originally designed to mimic the human voice. It creates a **band pass filter** which is then ‘swept’ across the frequency range by the user.

In simple terms this means that it boosts a small range of frequencies and then (typically via a pedal) a user varies what **frequencies** are being boosted. As well as a pedal this frequency “sweep” can also be controlled automatically using an LFO (Low Frequency Oscillator).

4- Synthesisers, Samplers and MIDI

Subtractive Synthesis *4.1*

Synthesis in Music Technology refers to the creation of a sound. Additive synthesis creates complex sounds (audio waves) by combining simple audio waves together.

A good real-world example of this is a pipe organ, where each 'pipe' creates a single sine wave and multiple pipes can be engaged to get the exact sound you want.

Most synthesisers you will be using deal with **subtractive synthesis** where you start with a more complex wave and then start taking away or **filtering** the elements of the sound you don't want.

Stage 1: Oscillator

The first step when creating a sound is to generate waves using an oscillator. An oscillator generates a single, constant wave that can then also be combined with other oscillators.

Oscillators have different waveforms to choose from (**Sine**, **sawtooth**, **triangle** and **square** are the most common *4.2*) that all have a characteristic sound and are used for different effects.

*E.g. **Sine waves** can emulate sounds such as flutes, whereas the nature of a **sawtooth wave** gives it a timbre similar to brass.*

Other settings include the ability to detune the synth to a different note, and also adjust the **PWM** (Pulse Width Modulation). **PWM** sends a pulse that has a similar effect on the sound to that of a chorus effect or two slightly detuned oscillators playing together.

Stage 2: Filtering **[See Chapter 1- Filtering]**

Once you have a single or several waves making your sound, you can use a **filter** to change the tone. Different synthesizers will have different levels of possible detail with an EQ.

Dreamstation for example only allows you to add LP (low pass) and HP (high pass) filters and then adjust the cut-off frequency. So for greater control you would need to add EQ to a track separately.

The filtering stage also allows you to use the **envelope/ADSR** and set how quickly or slowly your sound starts and ends.

Stage 3: Modulation and LFOs

The final stage before adding any additional dynamics and effects processing is **modulation**. Modulation involves making slight changes to the timbre (tone) of your sound.

2 important types:
Vibrato: Modulates the sound so there are fluctuating alterations in **pitch**. Similar to a violinist vibrating a string.

Tremolo: Modulates the sound so there are fluctuating alterations in the **volume**.

Modulation Controls:
Modulation Wheel: A **modulation wheel** is a physical control on a MIDI controller. Usually set to control the **frequency** of a **tremolo**, but can be mapped to control anything on the synth.

LFO (Low-Frequency Oscillator): Another method of controlling modulation. An LFO is a wave that oscillates at under 20 Hz (below human hearing) and can control various parts of a synth; such as vibrato, tremolo and even a LPF.

Samplers

A sampler is an electronic musical instrument similar in some respects to a synthesizer but, instead of generating sounds, it uses recordings (or 'samples') of sounds that are preloaded or recorded into it by the user.

These sounds can then be pitch shifted into various notes and played back using a keyboard or similar **input device**.

Samplers offer similar abilities to change the sound as a synthesizer including **filters** and **modulation**.

Musical Instrument Digital Interface (MIDI)

MIDI is the common language between musical instruments and computers. It translates any aspect of a musical note (length, velocity, panning etc.) into **digital information**.

Also MIDI editing is **non-destructive** which means that even after it has been recorded **any** aspect of a MIDI track can be changed. This means that if you play a wrong note, you can go back and change it without re-recording.

All MIDI information is either binary (on/off) or has **127 degrees of sensitivity**. *So a note not pressed at all would have a velocity of 0 and a note played as hard as possible would have a velocity of 127.*

Some key MIDI data types:

Note start	Binary
Note end	Binary
Velocity	0-127
Modulation Wheel	0-127
Pitch Bend	0-127 (Fully up by a note is +63, fully down by a note is -63 and the original note is 0.)
Pan	0-127 (Hard right is +63, hard left is -63 and central is 0.)
Volume	0-127

5- Microphones and the Recording Process

Microphone Types

There are 2 main types of microphone that you will be using; **condenser** and **dynamic**. You will need to understand their features and the appropriate one to use for certain instruments. It is also important (particularly for A2) to understand specifically how they work.

A table outlining the key features and differences between the two as well as when they should be used:

Condenser	Dynamic
	
Fragile. Designed as a studio mic and so handled carefully and always stored safely.	Robust. Designed as a 'gig' mic able to be handled more roughly.
Wider frequency range and more responsive to different dynamics.	Smaller frequency range and less adept at capturing the nuances of performance.
NOT suitable for high SPL sounds. (Sound Pressure Level)	Good for recording high SPL sounds.
Best choice for most applications; vocals, acoustic guitars, pianos, violins etc. But **not** very loud noises e.g. drums.	Best choice for drum kits, micing amps of electric guitars and bass as well as loud 'screaming' vocals.
Needs phantom power (48v).	Doesn't need phantom power.
Comparatively expensive.	Comparatively cheap.

N.B. SPL (Sound Pressure Level) describes how much pressure is caused by the sound wave. Factors such as volume, proximity to the microphone and environment all effect it. ***High SPL sounds can damage more fragile microphones.***

Condenser/Capacitor Additional Information

A condenser mic comes in 2 forms. A large diaphragm condenser (LDC) and a small diaphragm condenser (SDC). The big difference between them is their **frequency response** as the LDC tends to pick-up the deeper tones better, whereas the SDC gives a slightly ‘breathier’ response with more treble.

Otherwise they both have the condenser features listed in the previous page and both work in the same way.

Condenser microphones require power from a battery or an external source known as **phantom power** which is 48v.

Condensers also tend to have 2-3 switches so you are able to make adjustments to what sounds the mic is picking up.

One control (that isn’t on any of our mics) is a switch where you are able to change the **polar pattern** of the mic.

There is also an **attenuation switch** and the **low-cut switch.**

Attenuation switch Low-cut switch

The **attenuation switch** reduces any sound you are recording by 10dB. This is useful if you are recording particularly loud sounds and are struggling to stop the signal **peaking**.

The **low-cut switch** reduces any very ‘bassy’ sounds at the point of recording. Useful for removing any unwanted thumps that may be picked up.

How Microphones Work (A2)

Microphones are **transducers** that convert sound energy into electrical energy that can then be ‘read’ by other equipment.

Condenser

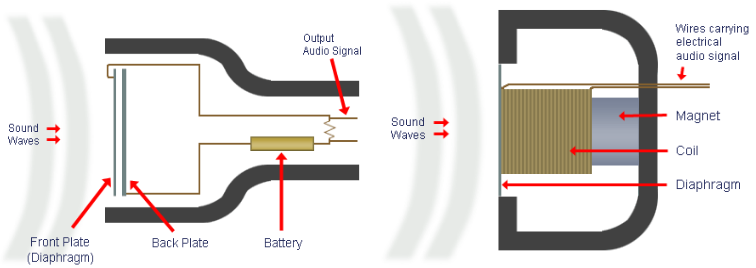
Inside the mic is a capacitor. The capacitor has two plates with a voltage between them. One of these plates is made of very light material and acts as the **diaphragm**.

The diaphragm vibrates when struck by sound waves, changing the distance between the two plates and therefore changing the **capacitance**. This variation in capacitance is what gives us our electrical representation of the sound wave.

Dynamic

A dynamic microphone uses a magnetic field to convert the sound waves into electrical energy. As you may remember from GCSE science, when a magnet moves near a coil of wire an electrical current is generated in the wire. A dynamic microphone uses this principle.

The diaphragm is attached to the coil. When the diaphragm vibrates in response to incoming sound waves, the coil moves backwards and forwards past the magnet. This creates a current in the coil which is channeled from the microphone along wires creating the output.



Cross sections of a condenser (left) and a typical dynamic (right).

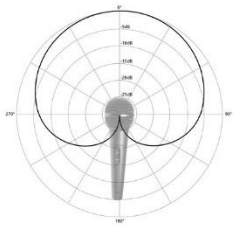
Polar Patterns and Frequency Response

Polar Patterns

Most microphones are not able to ‘hear’ sounds in all directions to the same level of sensitivity. In fact microphones purposefully cancel out audio from certain directions to avoid **spill** (sound you don’t want to capture).

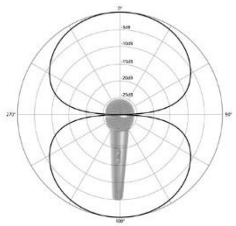
A microphone's **polar pattern** indicates how sensitive it is to sounds arriving at different angles from its centre.

Cardioid



- The most common polar pattern.
- Able to pick-up sounds in front of and slightly to the sides.
- Unable to pick-up behind it.
- Used for close-mic and live gigs as it can reject sounds such as fold-backs and audience noise.

Bi-directional/Figure 8



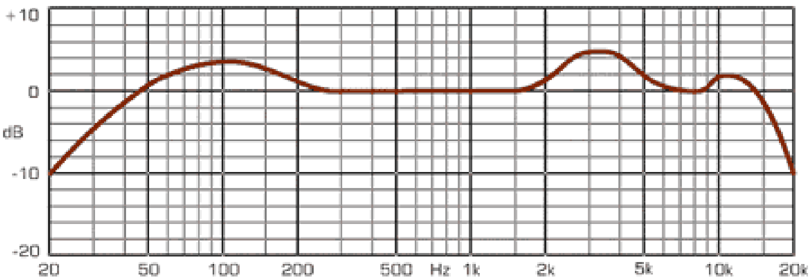
- Able to pick-up in front and behind.
- ‘Blind’ spots on either side.

Omnidirectional

- Can pickup in all directions.
- No ‘blind’ spots.

Frequency Response

As well as polar patterns, another thing you may wish to consider when choosing a mic is the **frequency response**. Usually on the box in the instructions of a new mic you will have a **frequency response graph/curve**. This is a diagram that tells you how effective this mic is at picking up certain frequencies.



Frequency response for the AKG D112

The example above is for an AKG D112 which is designed to be a mic for a kick drum. The peaks denote the frequencies that have been emphasized or attenuated by the mic.

So in this example the bass frequencies between 50-200hz and treble frequencies from 2khz-5khz and then 10khz-13khz will be particularly pronounced, and the frequencies below 50hz and above 15khz will be reduced.

Generally the best condenser microphones have what is called a **flat frequency response**. This means that the microphone will reproduce the desired sounds with no emphasis or attenuation of any frequencies. Basically what you hear is what you will get.

N.B. It is also worth noting that speakers, monitors and headphones have different frequency response curves- so your recordings will not always sound the same on every piece of equipment.

6- Mixing and Mastering

Levels and Stereo Field

After the **recording process** is the equally vital **mixing process**. Mixing is the stage of getting **one stereo track** from the various tracks you've recorded.

The mixing process encompasses filtering, dynamics processing, effects processing and also the **balance** of tracks.

Levels

Put simply, levels are how loud your tracks are in relation to each other. If your guitar track is too loud then you can turn it down using the sliders on Sonar's built-in mixing desk; the **console view *6.1***.

Stereo Field

Stereo field relates to **panning**. When mixing in stereo (which you will be) you have 2 channels; the left ear/speaker and the right. With panning you can determine how much of each track/instrument can be heard in each ear.

Track Automation

It is possible to automate these settings and so have tracks get increasingly louder/quieter automatically or even have a track move from the left ear, to the right and back again.

N.B. Panning is an excellent way of separating tracks so they can be distinguished from each other.

However 'extreme' panning is not considered good practice- i.e. do not have lots of examples of tracks only being in one ear.

Also make sure that, after you've panned your tracks, your L and R channels are still similar volumes.

Using EQ

As mentioned in the **Filtering** chapter, 2 instruments that are both vying for the same frequencies tend to get in each other's way and 'muddy' your track. *1.1* in the appendix shows how various instruments can 'clog up' the frequency spectrum even with frequencies that offer nothing to the musicality of the instrument.

When this occurs it is important to use **filtering** effectively and remove these unnecessary frequencies so that each instrument has their own place on the spectrum.

1) The first thing is to build up the tracks in your mix in order of importance. So if your vocal is the most important sound in the mix, then you should use **creative EQ** to get the perfect sound and then you should be trying to preserve as much of that sound in the final mix as possible.

2) Then add in the next most important track. If it hides anything important in the vocal, that second track needs to be **cut with EQ** so that it doesn't compromise the vocal whilst still generally keeping the sound you want of that second track. The trick is to mute and unmute the second track while actually concentrating on the vocal sound.

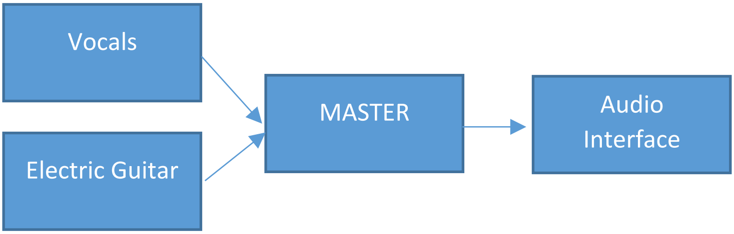
3) This same principle can then be applied as each subsequent track is added in. If you've got a very busy arrangement, this EQ approach means that the filtering will usually become more and more severe as tracks get less important.

The bottom line is that if you bypass the EQ cuts and the sound itself doesn't appear to lose anything important, the sound you're cutting away is expendable, and getting rid of it will only improve the clarity of other, more important instruments in the mix.

Buses and Side-chain Compression

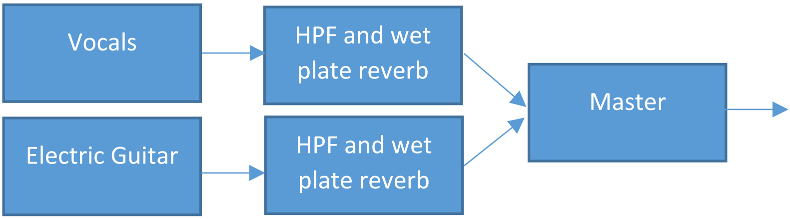
Bus

A bus is essentially a pathway which allows you to route one or more audio signals to a particular location. For example when Sonar opens all your tracks are already 'routed' to a bus called **Master** which then has your Soundcard as an output.

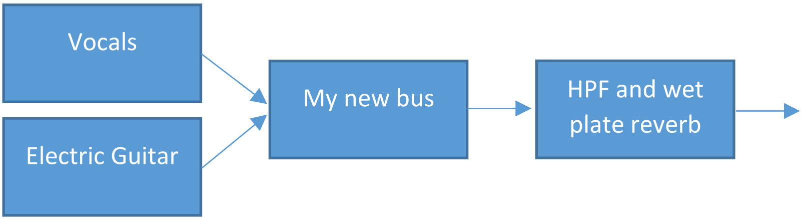


You can create buses and 'group' together certain tracks. This is useful if you want to apply the same filtering and effects to multiple tracks.

So instead of adding it to each individual track;



You can route your instruments to a **bus** and apply your filtering and effects to the bus instead.



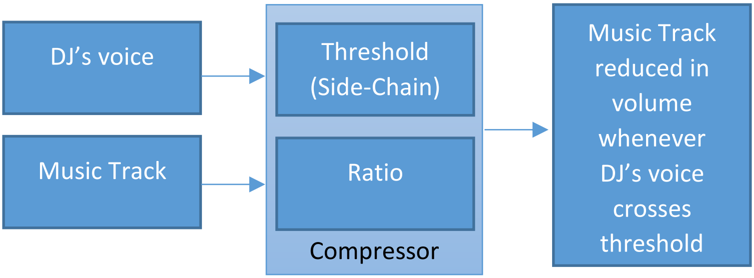
Side-Chain Compression

We were introduced to **compression** in the Dynamics Processing chapter; which is when a sound's output volume is reduced when it **passes the threshold**.

Side-chain compression follows this principle, but reduces a sound's output when a **different sound** passes the threshold.

The early application of this was with radio DJs and a technique called **ducking**. A compressor is applied to the music track with a **low threshold** and a **high ratio**, and then separately the DJ's voice is 'bussed' to the compressor.

Whenever the DJ spoke, his voice would cross the **threshold** of the compressor on the music track and the music would turn down, 'ducking' out of the way of the DJ's voice.



This principle can be applied to your mixing if, for example, you have a synth pad and a kick drum 'muddying' each other.

a) First you would need to add a compressor to your **synth pad** track with a low threshold and high ratio.

b) Then you would need to route (bus) your kick drum's output to the compressor.

c) Now whenever your kick drum plays, your synth pad will be compressed and reduce in volume to duck out of the way.

Mastering

After you have mixed your tracks, you need to **bounce down** your tracks to a single stereo track. Once you have it is still possible to add filtering as well as effects and dynamics processing to the **entire mix**. This is known as mastering.

As well as the tools we have covered already, there are also 2 tools that are particularly useful in the mastering stage; **normalizing** and **multi-band compression**.

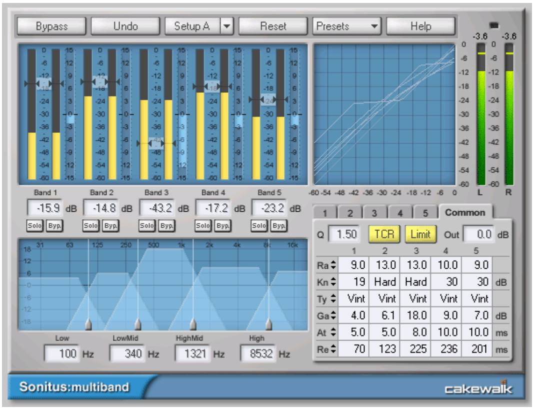
Peak Normalisation/Normalising

When you have finished your track it is the industry ‘norm’ for it to **peak** (be at its loudest) at 0db. A normalizer finds your track’s peak and then turns up the **entire track** to make it peak at 0db without altering the balance of your tracks.

Multi-band Compression

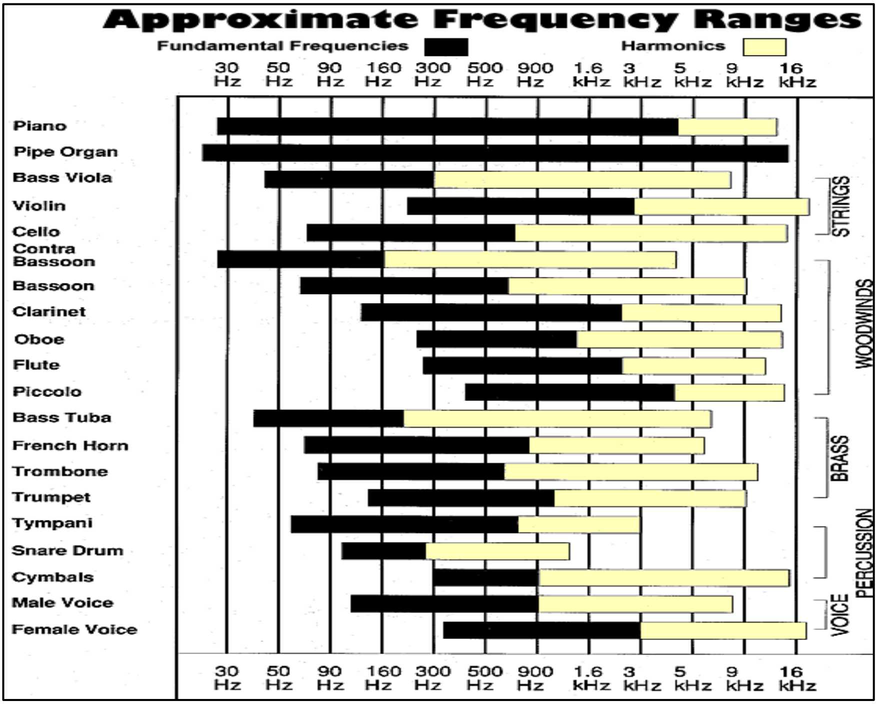
If EQ and compressor had a baby, you’d get a multi-band compressor. This tool allows you to apply different compressors to different ranges of frequencies on one track.

Practically this means that your bass and treble frequencies can have less dynamics variation than your middle.



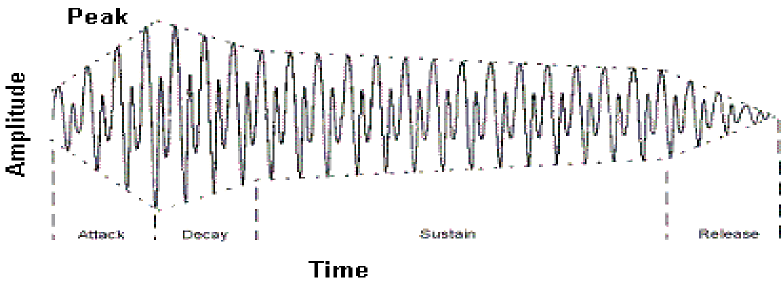
Appendix

1.1



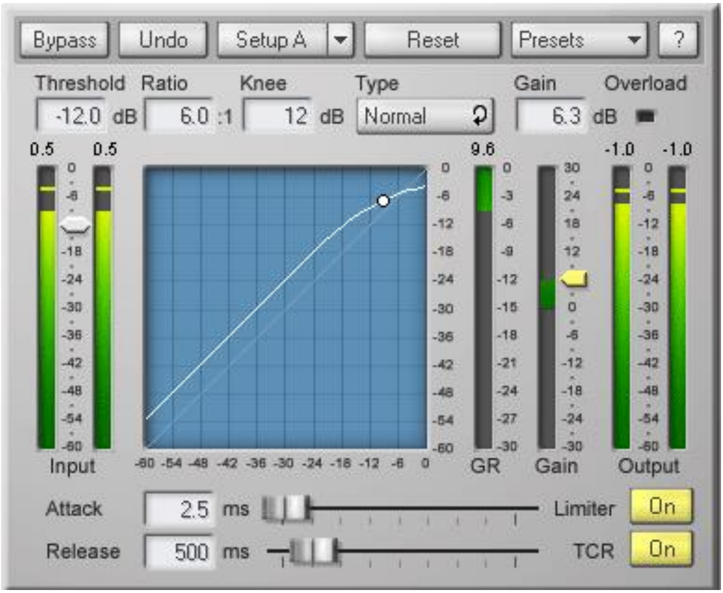
An image showing the approximate range of frequencies that various instruments can occupy.

1.2



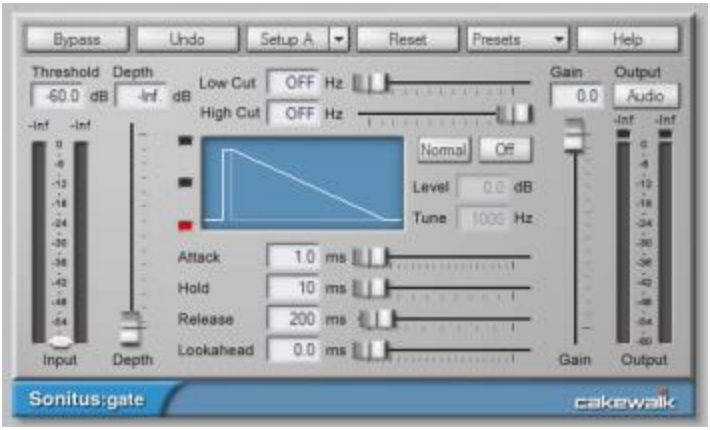
An image showing the different stages of a sound wave (ADSR)

2.1



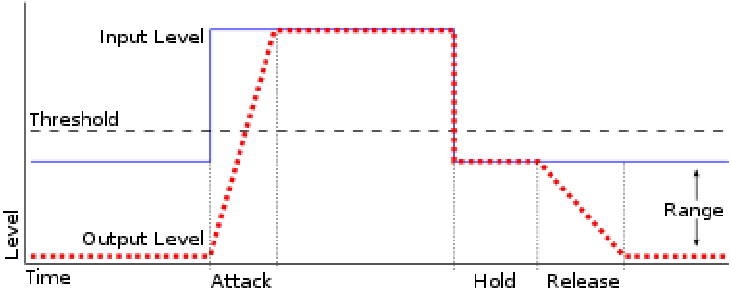
An image showing the Sonitus:FX Compressor in Sonar 8.5.

2.2



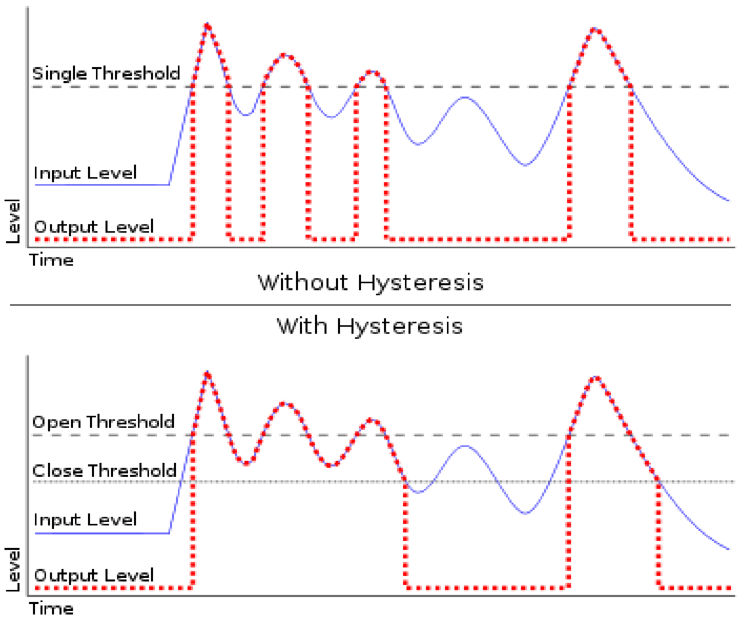
An image showing the Sonitus:FX Gate in Sonar 8.5.

2.3



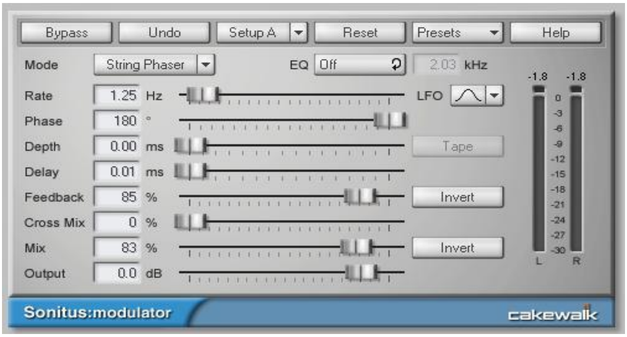
An image showing the impact of attack, release and hold on an audio signal. The input level is the steady line, the output is the broken line.

2.4



An image showing the impact of hysterisis on an audio signal.
The input level is the steady line, the output is the broken line.
Note the reduction in erratic opening and closing of gate in the lower image.

3.1

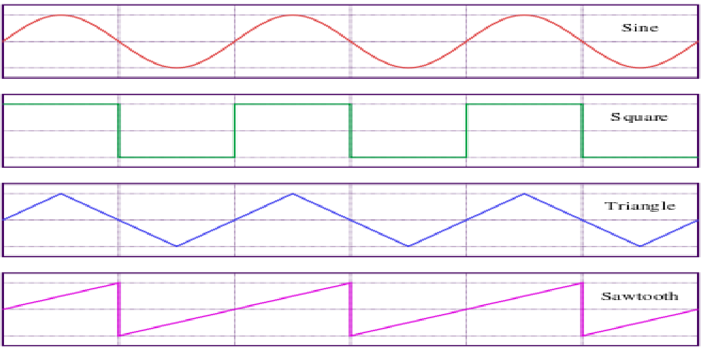


An image showing the Sonitus:FX Modulator in Sonar 8.5.
*This plugin can act as both a phaser and a flanger by changing the **mode**.*

4.1

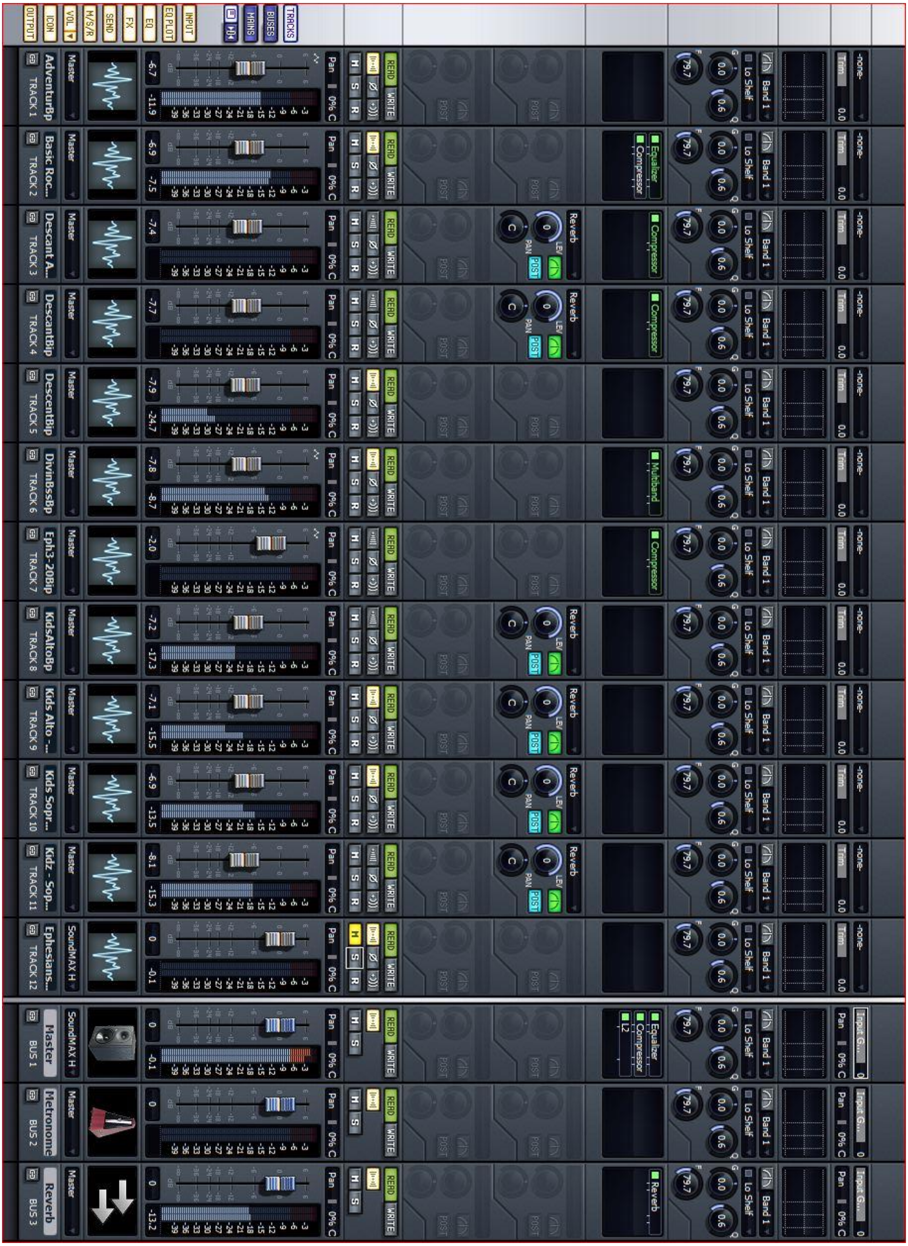
An image showing the Dreamstation DXi2 synthesiser. A good example of a subtractive synthesizer in Sonar 8.5.

4.2



An image showing the different wave types you can select in a synthesizer.

6.1



An image showing Sonar 8.5 ***console view****.*