

ARC-1.1 Analog Recording Channel

Users Manual

ARC 1.1 USERS MANUAL VERSION-1, AUG 2004. CONTENTS COPYRIGHT BUZZ AUDIO LTD 2004.
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Welcome…

Thank you for choosing the Buzz Audio ARC1.1 Analog Recording Channel. In this manual you will find important information regarding the use of the ARC and we suggest you do read it before using the unit to become familiar with all the controls and routing options.

If after unpacking the ARC you find any damage you should contact your dealer or supplier immediately for advice on what to do.

We also suggest you retain the original packaging at least during the warranty period in case you need to return the unit for service, however we are confident this will not be necessary!

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1] The Mains Input.

As a safety precaution your ARC maybe shipped without a mains fuse fitted. Before use, you must select the correct mains voltage for your local supply on the rear panel and fit the correct mains fuse type into the fuse draw for that voltage.

- **Setting the Voltage Selector…**

You will need a medium size flat blade screwdriver.

If your local mains voltage is 110V to 120V set the rear panel voltage selector to 110V.
If your local mains voltage is 220V to 240V set the rear panel voltage selector to 220V.

POWERING UP THE ARC 1.1 WITH THE WRONG VOLTAGE SETTING MAY CAUSE SEVERE INTERNAL DAMAGE!

- **Selecting the Correct Fuse…**

In the supplied accessories bag you will find;

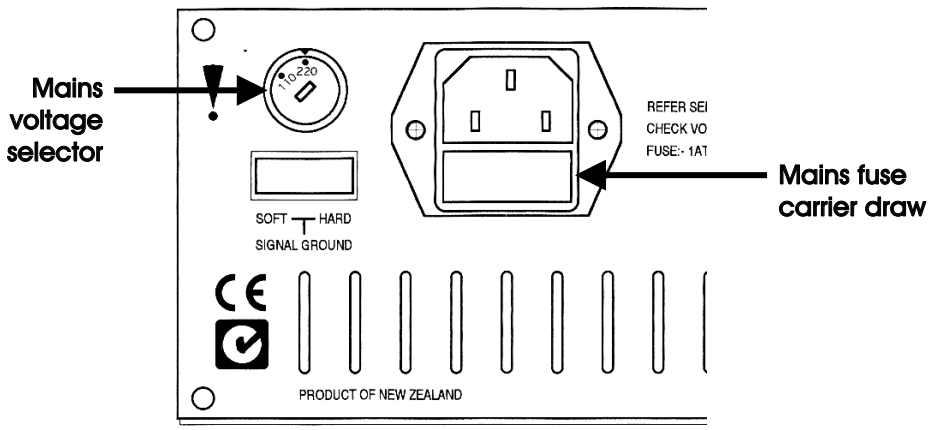
2x 1 Amp slow blow fuses, FIT for 220V to 240V.
2x 2 Amp slow blow fuses, FIT for 110V to 120V.

Slide out the fuse draw below the IEC power inlet using a small flat blade screwdriver and insert the appropriate fuse into the carrier clip. Note there is also a position in the carrier for a spare fuse (the square tube bit) and we suggest you store the second supplied appropriate fuse here.

- **Please Note…**

Fitting the wrong fuse may result in the fuse blowing on power up or inadequate protection. Fitting a fast blow type fuse may also result in the fuse blowing on power up.

Your dealer may have already set all this up for you but it pays to check it before powering up the ARC.

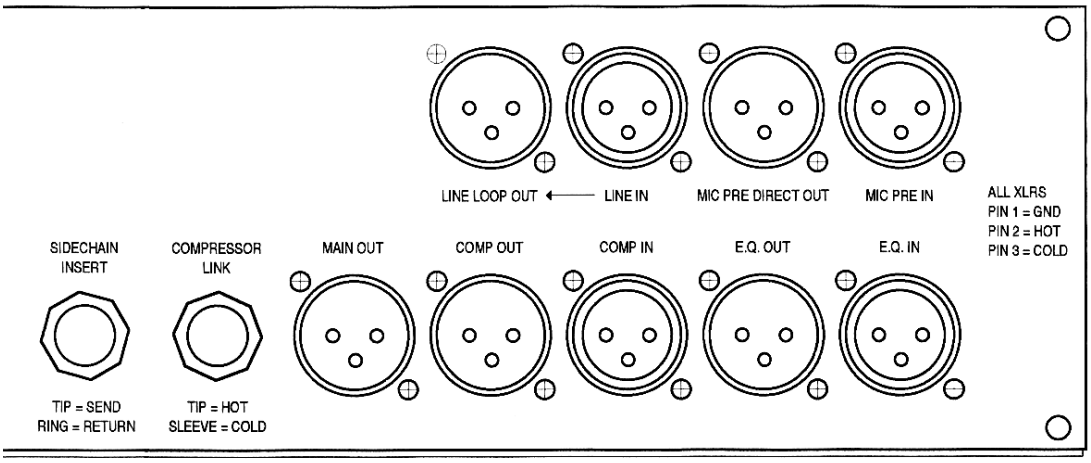


2] Connections…

All the XLR connectors on the rear of the ARC1.1 are wired as follows;

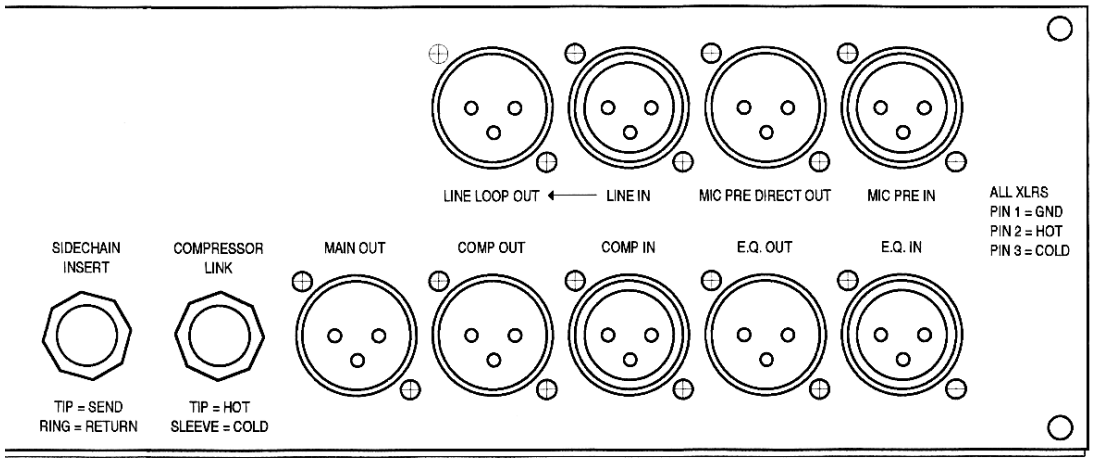
Pin 1 = Chassis Ground
Pin 2 = Signal Hot (+)
Pin 3 = Signal Cold (-)

All XLR inputs and outputs are balanced, except for the Mic Pre Direct Out, see below for further detail on that.



- **MIC PRE IN** – obviously, this is the input to the ARC Microphone Preamplifier section. Connection of any source other than a balanced microphone is not really recommended. Note that with the **+48V** front panel switch **ON** (down), 48V phantom power is applied to pins 2 and 3 in the normal fashion.
- **MIC PRE DIRECT OUT** – this output is a DC coupled direct feed from the ARC microphone preamplifier and is active at all times. Although this output is unbalanced (pin 2 HOT), it should be treated as balanced to avoid any earthing issues.
- **LINE IN** – this is the Balanced input to the ARC Line Preamplifier section and is active when **BAL** is selected on the front panel.
- **LINE LOOP OUT** – this connector is hard wired to the Line In and carries the exact same signal. It is provided for ease of connectivity where the source may need to feed another device in daisy chain.
- **E.Q. IN** – this connector is the External pathway in to the ARC1.1 Equaliser section. Any Equaliser band switched to **EXT** (external) will receive its signal from this input.
- **E.Q. OUT** – this connector is the External pathway out of the ARC1.1 Equaliser section. Any Equaliser band switched to **EXT** will output via this connector.

Connections cont'd…



- **COMP IN** – this connector is the External pathway into the ARC1.1 Compressor/Limiter section. If the Compressor and/or Limiter are switched to **EXT**, that (those) section (sections) will receive its input from this connector.
- **COMP OUT** – this connector is the External pathway out of the ARC1.1 Compressor/Limiter section when either section is switched to **EXT**.
- **MAIN OUT** – as the name suggests, this is the Main line level output from the ARC. It will carry either the MIC, LINE or S/C (sidechain) signals depending on the from panel switch settings.
- **COMPRESSOR LINK** – this ¼" jack socket is used to link two ARC Compressor sections for stereo operation. A normal mono ¼" jack lead is all that is required to connect the two together, however, a balanced wired (TRS) jack lead will also work. It becomes active when **LINK** is selected on the front panel, note that if using two ARC's, both units need to be switched into **LINK** mode.
- **SIDECHAIN INSERT** – this TRS ¼" jack socket provides access to the Sidechain signal of the Opto Compressor (but not the Limiter). It is similar to an insert on a mixing console and allows for the connection of an external device (such as an equaliser) to "modify" the signal sent to the sidechain of the Compressor. Note that it is post any ARC Equaliser section that is switched to **S/C** mode. This connector is wired;

  Tip = signal send (or input to device)
  Ring = signal return (or output of device)
  Sleeve = signal common or ground.

Signals returning via this connector can be monitored with the **S/C MON** (sidechain monitor) function.

3] System Overview

The ARC1.1 has been designed to offer the user a high degree of flexibility via various routing options. The unit can be used as an individual mic preamp, line preamp, equaliser and compressor/limiter, all at the same time. To achieve this the architecture consists of 4 main signal paths;

1. Main Path (**IN**)
2. Equaliser External Path (**EXT**)
3. Compressor External Path (**EXT**)
4. Sidechain Path (**S/C**)

The Main Path is either the Mic or Line input as selected on the front panel. Any or all of the Equaliser sections (bands) may be switched into the Main Path to equalise the mic or line signal. The Compressor and/or Limiter may also be switched into the Main Path, pre or post the Equaliser. Additionally, an audio transformer may be switched into the Main Path to add some further "colour" to the mic or line signal.

The Equaliser External Path allows the user to access any or all Equaliser bands independently from the Main Path. Similarly, the Compressor External Path permits separate operation of the Compressor and/or Limiter.

Any or all Equaliser bands can also be switched into the Sidechain Path, which is the audio control circuit of the Opto Compressor (does not affect the Limiter control circuit). This feature allows the user to increase or decrease the sensitivity of the Compressor at certain frequencies (more on this later). Additionally, an external device may be patched into the Sidechain Path via the **SIDECHAIN INSERT** rear panel connectors.

The Sidechain Path can be auditioned using the **NORMAL<>S/C MON** switch, where the Main Path signal is replaced with the Sidechain Path signal, appearing on the rear panel **MAIN OUT** connector.

> **Designers Note**
> To achieve all this signal routing and maintain a clean signal path, the ARC utilises a switching matrix consisting of 22 DPCO relays functioning like an electronic patch bay. Each "element" of the ARC is individually and directly connected to this matrix via high quality OFC cable rather than a myriad of PCB tracking, eliminating cross talk and bleed that could occur at high frequencies, and keeping the signal paths as short as possible. In addition, each ARC element has it's own signal earth return back to the power supply forming a star earth system to further reduce individual "section" interaction problems.

4] The Mic Preamp…

To use the ARC Mic Preamp separately from other elements of the ARC, connect to the **MIC PRE DIRECT OUT** rear panel connector. This output carries the amplified mic signal at all times.



- **MIC GAIN** – adjusts the gain or volume of the mic preamp stage from +9 to +50dB (or +24 to +65dB when the +15dB mode is engaged, see next).

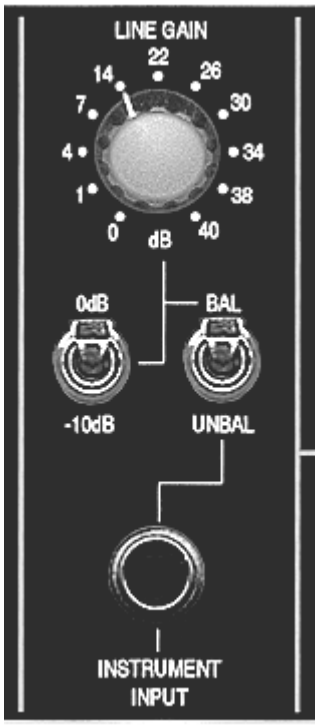
- **0dB<>+15dB** – this switch adds an extra 15 dB of gain to the mic preamp stage. Generally the preamp can be operated at the 0dB setting, but the extra gain is useful for low output microphones such as ribbon types or when you are recording a pin drop or other quiet sound.

- **+48V<>ON** – turns the 48V phantom power necessary for condenser mics and active DI boxes on or off, with the down position being **ON**. A special soft start/stop circuit minimises thumps and bangs normally associated with switching phantom power, but it is still a good idea to **MUTE** the ARC output before operating this switch.

- **MIC LOAD** – this is a very interesting control in that the load presented to the microphone at the ARC **MIC** input can be continuously varied from 220 ohms through to 5500 (5.5k) ohms. The purpose of this control is to provide yet another sound shaping facility, there is no wrong or right position for this control. Some microphones (transformer coupled types for example) may exhibit a significant sound change at the extremes of this control, whilst others (transformer-less types for example) may not. A good starting point in either case would be mid way at 1.4k (1400) ohms, the user can then fiddle and hear what differences are to be discovered.

Designer Note
The mic preamp stage in the ARC1.1 utilises the proprietary Buzz Audio BE40 True Class A amplifier modules. These modules offer exceptionally low noise coupled with very high speed, ensuring signals are amplified without loss of transient detail. The input to the BE40 modules is via high quality WIMA brand coupling capacitors. The pad-less input design ensures that the mic signal hits the front end directly no matter what gain you require. This means the preamp will sound the same at low gains as well as high gain, which is important when recording loud sources, such as close mic-ed vocals and drums.

5] The Line Preamp…



- **LINE GAIN** – adjusts the gain or volume of the selected line input source from 0dB through to +40dB. If the **–10dB** mode is engaged (**BAL** input only, see below), the gain range is –10dB through to +30dB.

- **BAL<>UNBAL** – this switch selects either the rear panel balanced **LINE INPUT** or the front panel unbalanced **INSTRUMENT INPUT** as the line preamp source.

- **0dB<>-10dB** – this switch is only active when using the **BAL** (rear panel) input. It is provided to prevent overload of the balanced input from high level signals (such as those from DAW outputs), and brings the signal down to manageable levels.

- **INSTRUMENT INPUT** – use this input for high impedance inputs such as acoustic pick-ups, bass and electric guitars, keyboards or any other source that requires minimum loading. The input impedance of this input is approx 1Meg (that's 1 million) ohms. This input is active when **UNBAL** is selected.

> **Designers Note**
> In addition to attenuating the balanced input by 10dB, the **0dB/-10dB** switch raises the **LINE IN** input impedance from 20k ohms to approx 70k ohms. This feature will help the driving device deliver a cleaner signal at high levels due to less current being demanded from the device output stage and consequently less generated distortion.

6] Main Output Section…



- **MIC<>LINE** – selects which input preamp will feed the **MAIN OUT** rear panel connector via the Main Path. **MIC** selects the mic preamp (obviously) and **LINE** selects either the **BAL** or **UNBAL** line inputs, as above.

- **OUTPUT GAIN** – this control adjusts the signal level at the **MAIN OUT** connector and provides an additional 10dB of gain to the system if required. It can also be used to

6] Main Output Section continued…



attenuate the output down to match lower levels that may be encountered with semi pro sound cards or –10dBv equipment, without sacrificing noise performance. The **OUTPUT GAIN** control is always post any Equaliser or Compressor elements that may be switched into the Main Path.

- **ϕ NORM<>MUTE<>ϕ REV** – this switch is used to invert the polarity (or phase) of the Main Path signal. In the up position, the output signal is "in phase" with the input signals. In the down position, the output signal is "out of phase" with the input signal. In the centre position, **MUTE**, the output is switched off or muted. The MUTE function is handy when turning on/off the phantom power to avoid thumps or when changing microphones or to mute the signal between takes etc.
- **NORMAL<>S/C MON** – permits the user to listen to the Sidechain Path signal via the ARC1.1 **MAIN OUTPUT**. In the up position, the output is from the Main Path, in the down position, the output is from the Sidechain Path. This function is useful for listening to any equalisation that may be applied to the Compressor control circuit either via the ARC equaliser sections of via the rear panel **SIDECHAIN INSERT** connector.
- **CLEAN<>TRANNY** – this control switches a custom made audio transformer into the Main Path when in the down position. The transformer adds some low frequency harmonic distortion to the audio signal, offering yet another colour option to the ARC1.1. The transformer is inserted into the Main Path directly ahead of the **OUTPUT GAIN** control so it is possible to turn up the input level and reduce the output level for increased colour from the transformer. Very nice.

Designers Note
The audio transformer inserted by the CLEAN-TRANNY switch has been designed to sound like transformers from the early years of audio and is made with a fairly low grade steel core. Whilst it has a fairly flat frequency response from 20Hz to 100kHz, it is intentionally not "pristine" in sound quality. The amount of harmonic distortion added to the signal will vary with how hard it is pushed. At lower levels (0dBu) the THD (Total Harmonic Distortion) at 50Hz is in the order of 0.5%. At higher levels (+15dBu) this rises to approx 2% as the transformer begins to saturate. The distortion falls with increasing frequency, and measures approx 0.1% at 1kHz right up to +15dBu.

7] Equaliser Routing…

You will notice that every Equaliser band on the ARC1.1 has a switch labelled **S/C<>EXT<>IN** so rather than repeat this 5 times, we will explain it just once here! This switch is used to route the related band into the Sidechain Path (**S/C**), Equaliser External Path (**EXT**) or Main Path (**IN**) as described in the preceding System Overview section.

Your next question might be "what if I do not want the Equaliser in any Path". Lets look at a possible scenario…

Lets say you have set up some EQ for a vocal recording with the Mid Band 1 controls and that is switched to **IN** (Main Path). You have also have set up some EQ with the High Shelf to brighten an instrument running through the External Path, so that band is switched to **EXT**. You are also using the Opto Compressor for the vocal recording and you have used Mid Band 2 switched into the Sidechain Path (**S/C**) to boost around 2kHz and make the Compressor work harder around this frequency.

What do you do with the other bands you are not using and therefore want them out of circuit? The best thing to do is to centre the **CUT/BOOST** controls of the unused bands (ie 0dB) and switch them into the **S/C** path where the effect of extra electronic stages will make no difference to the sound quality of the vocal recording or instrument. In the case of the High Pass Filter, if you don't want that in the Main or External Path, set the **FREQ** control fully counter clock wise (25Hz) and switch that into the **S/C** path where it's effect will be minimal.

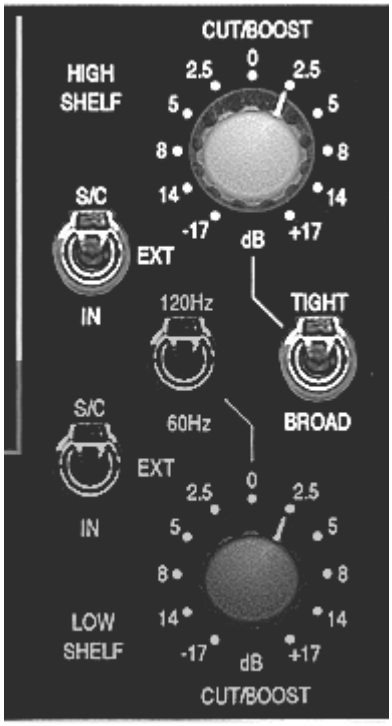
8] High Pass Filter…

High Pass Filters (HPF) are generally used to eliminate unwanted low frequency sounds from the audio signal. These sounds may be traffic rumble picked up in the recording or excessive low frequency energy in a vocal recording. When used in the Sidechain Path (S/C), the HPF can be used to make the Opto Compressor less responsive to low frequencies.



- **FREQ** – apart from the routing switch described above the HPF consists of only this single knob. The **FREQ** control sets the frequency in Hertz (**Hz**) at which signal level is attenuated by 3dB, sometimes referred to as the cut off frequency. Frequencies below the cut off point are then attenuated at the rate of 12dB per octave, where one octave equals half (or double) the frequency. For example, with the **FREQ** control set to 200Hz, signal level will be –3dB at 200Hz, -12dB at 100Hz and –24dB at 50Hz and virtually non-existent at 25Hz!

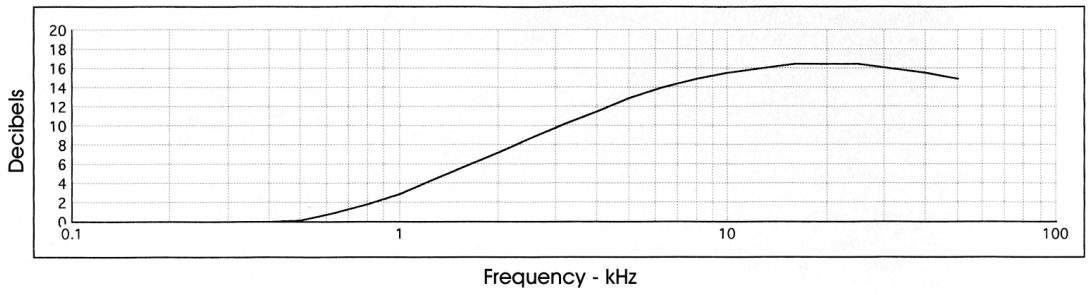
9] High Shelf EQ…



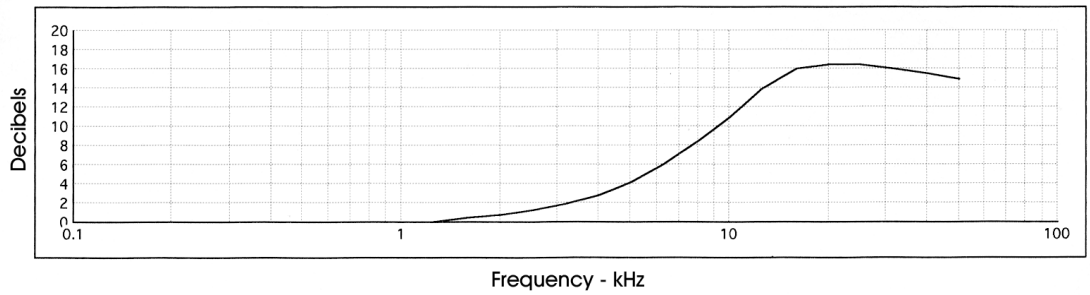
- **CUT/BOOST** – as the name suggests, this control varies the amount of attenuation or gain applied to the upper part of the audio spectrum and is calibrated in dB (decibels). A light detent in the centre of travel marks the 0dB point.

- **TIGHT<>BROAD** – selects the type of shelving curve. The **BROAD** position is very similar to the type of treble control you will find on a stereo hifi amplifier. The **TIGHT** position allows for the control of very high frequencies without less effect to the upper mid frequencies compared to the **BROAD** setting. The graphs below show the measured response curves of the two settings with the **CUT/BOOST** control fully clockwise (max boost). Note that the **TIGHT** position could also be used as a Low Pass Filter if desired.

High Frequency Shelf in Broad Mode



High Frequency Shelf in Tight Mode



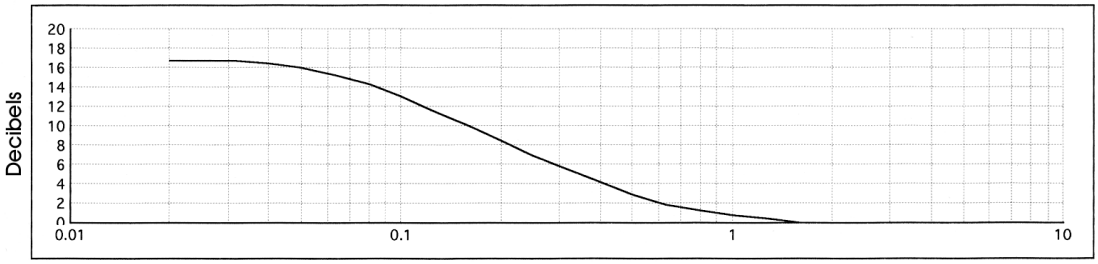
10] Low Shelf EQ…



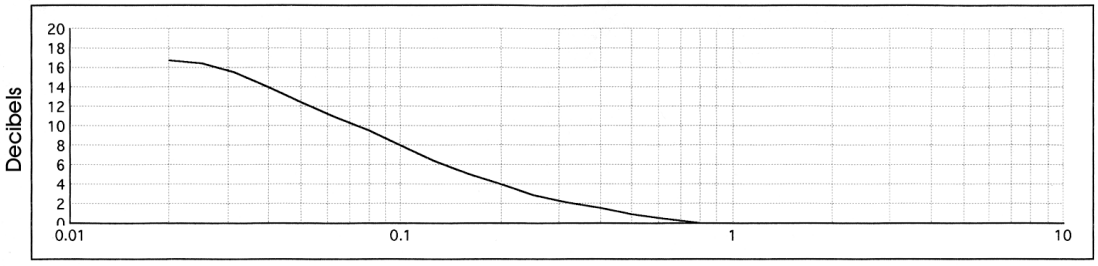
- **120Hz<>60Hz** – selects the turnover frequency of the Low Shelf control. The 60Hz position will add real "sub" to the signal, whilst the 120Hz position affects more of the upper bass frequencies as well. The graphs below show measured performance of the two settings with the **CUT/BOOST** control fully clockwise.
- **CUT/BOOST** – varies the amount of attenuation or gain at the selected turnover frequency, calibrated in dB.

Designers Note
The Low Shelf EQ section utilises a real choke (inductor) as the reactive element rather than an electronically generated one (gyrator). Although a lot more expensive than a gryrator, listening tests have shown the choke to be far superior with a much "tighter" bass sound.

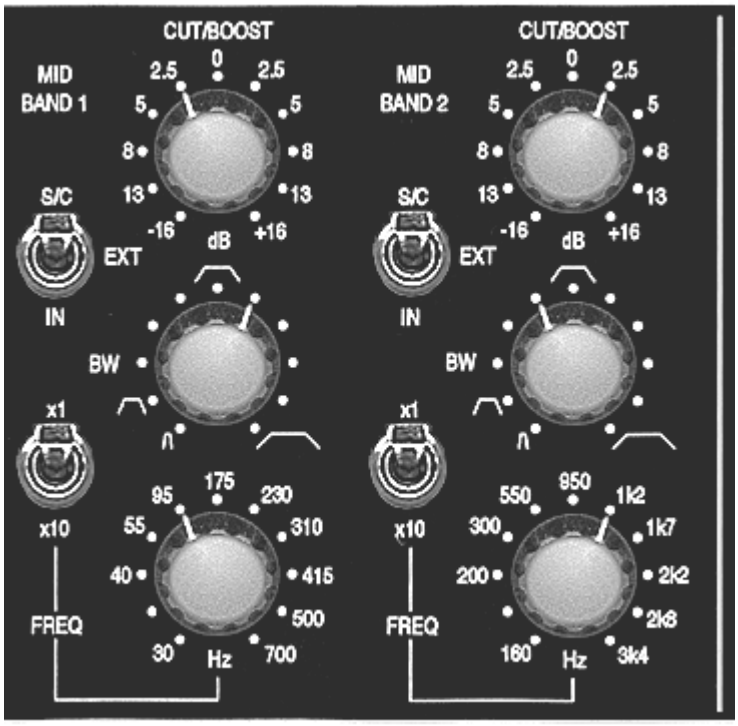
Low Frequency Shelf in 120Hz Mode



Low Frequency Shelf in 60Hz Mode



11] Mid Band EQ's…



- **CUT/BOOST** – similar to the Shelving sections, these controls vary the attenuation or gain of the signal at the frequency set by the **FREQ** control.

- **BW** – sometimes referred to as "Q", this control varies the width of the reject/pass band (**CUT/BOOST**) at the selected frequency (see **FREQ** below). Fully clockwise, the bandwidth is approx 1.7 octaves (wide). Fully counter clockwise, the bandwidth is approx 0.25 octaves (narrow). This is equivalent to a Q of 0.6 and 4.0 respectively. See the next page for graphs.

- **FREQ** – the Mid Band EQ sections are Band Pass (**BOOST**) or Band Reject (**CUT**) filters. The **FREQ** control defines the "centre" frequency at which the Pass or Reject Band is centred. The **BW** control is used to define the "width" of the Pass or Reject Band. Mid Band 1 has a centre frequency range of 30Hz to 700Hz (or 300Hz to 7kHz in **x10** mode). Mid Band 2 has a range of 160Hz to 3.4kHz (or 1600Hz to 34kHz in **x10** mode).

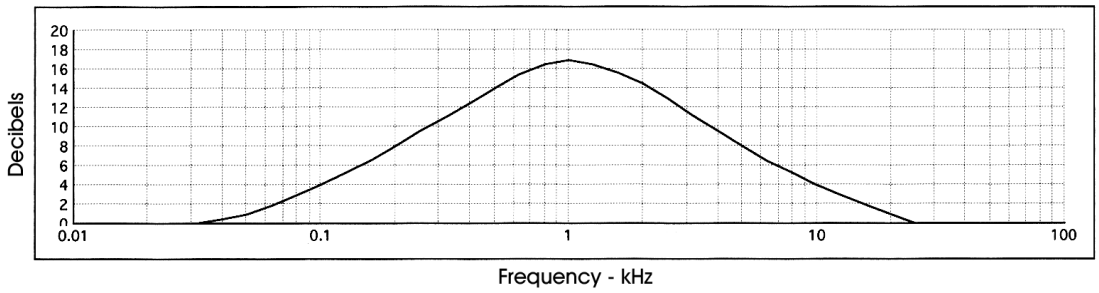
- **x1<>x10** – multiplies the range of the **FREQ** control by a factor of **1** or **10**, as explained in the preceding paragraph. Combining the two Mid Bands, this feature enables the user to choose any centre frequency within the entire audio band (and beyond!).

Designers Note
The actual filter utilised in the ARC1.1 Mid Band EQ design is based on a circuit topology called the Constant Amplitude Phase Shift network (CAPS). Back in the early 1980's, the CAPS network was suggested as an improvement over the more common State Variable Filter by well know UK audio designer Mr Steve Dove. I have always been intrigued by this topology and have pleasure in presenting it here within the ARC 1.1.

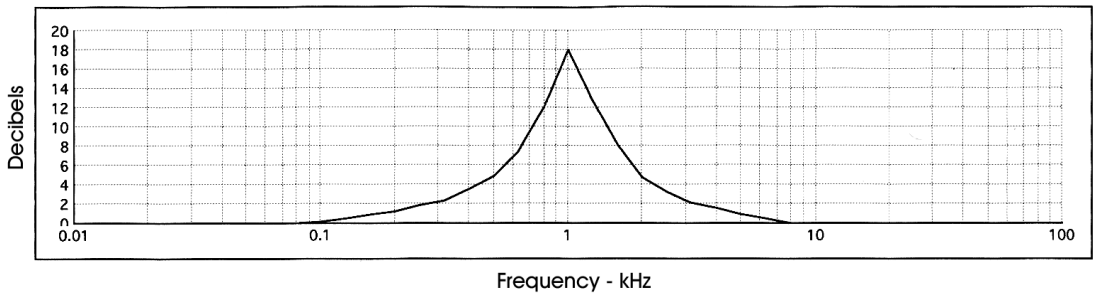
11] Mid Band EQ's continued…

The graphs below show the measured performance of the two Mid Band **BW** control for the two extremes, centred at 1kHz with **CUT/BOOST** fully clockwise.

BW Control in Wide Position (CW)



BW Control in Narrow Position (CCW)



12] Equaliser Glossary…

Bandwidth – is the difference between the upper and lower points where the filter response falls to 3dB below it's peak value on the way out of the pass band.

Centre Frequency – the centre frequency of the pass band of any type of filter or combination of filters is the geometric mean of the lowest and highest frequency of the pass band.

Equaliser – a device consisting of reactive elements that may be connected into an electrical circuit for the purpose of altering the frequency characteristics of that circuit either up or down.

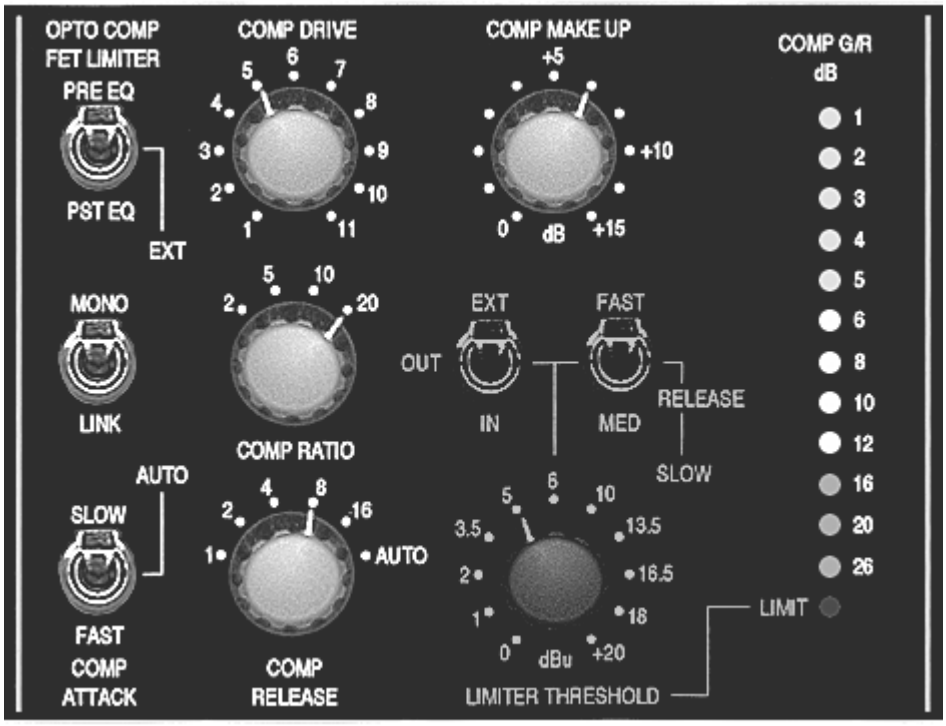
Filter – a device or network that favours certain frequencies at the expense of others. Filters can either pass or reject bands of frequencies (band pass or band reject) or can pass only high frequencies or low frequencies (high pass or low pass).

Frequency – indication of how many cycles of a repetitive waveform occur in 1 second. A waveform that has a repetition cycle of once per second has a frequency of 1Hz (Hertz).

Inductor – a reactive component that presents an increasing impedance with increasing frequency, often referred to as a "choke", and usually constructed with a coil of wire on a metal or air filled former.

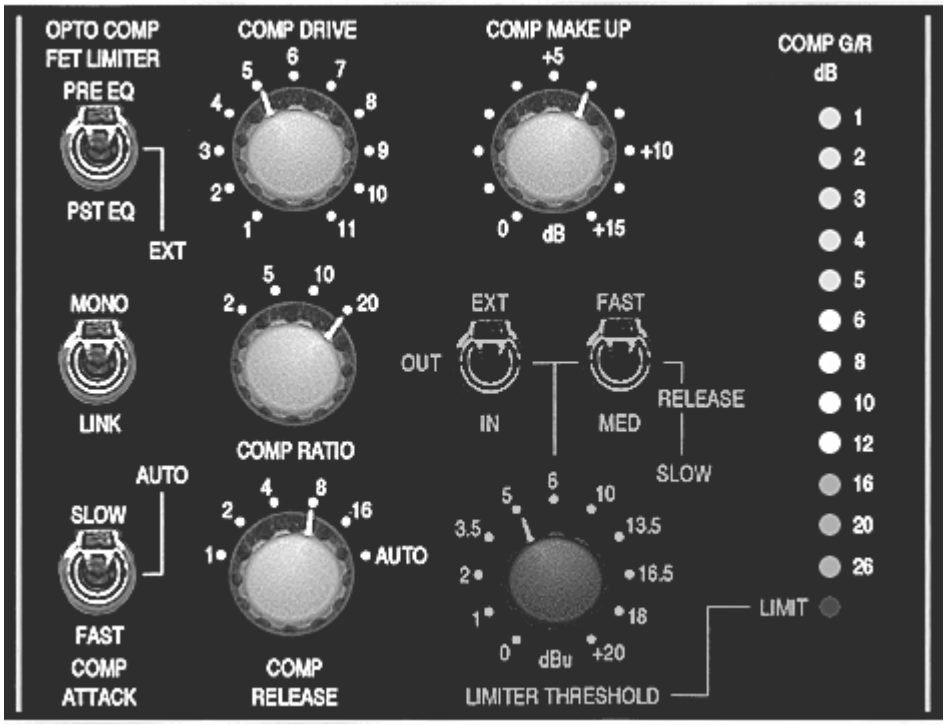
Octave – doubling or halving of a given frequency (or pitch). When a frequency is transposed up by one octave, its frequency is doubled.

13] Opto Compressor…



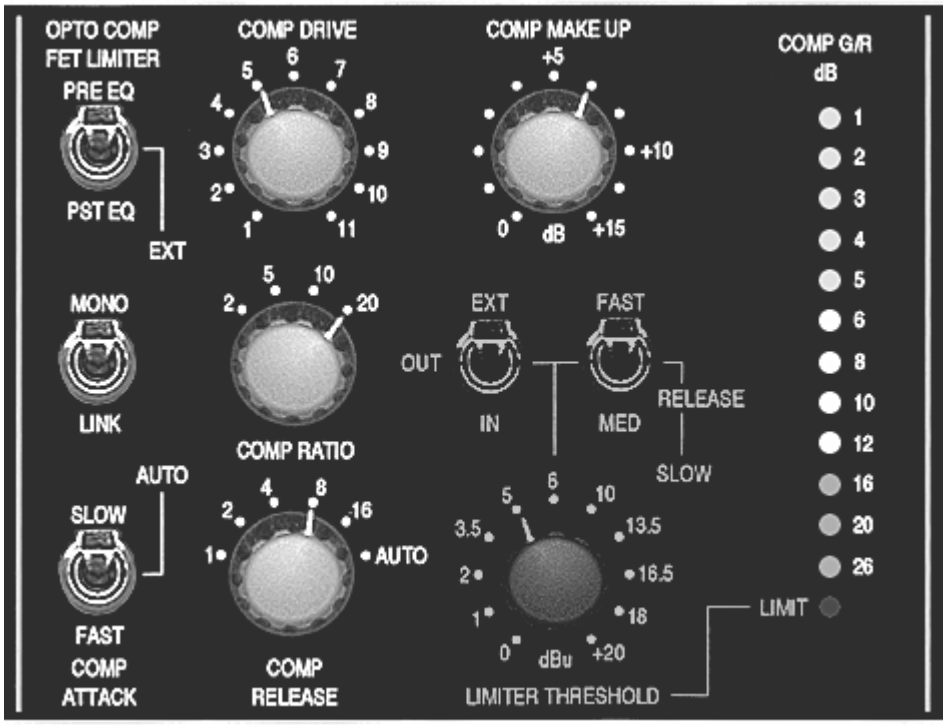
- **PRE EQ<>EXT<>PST EQ** – this switch controls the routing of the Opto Compressor section (and in some cases the Limiter section, see Section 15 for further detail on this). In the **PRE EQ** position, this switch places the Opto Comp into the Main Path ahead of any Equaliser sections that may also be switched into the Main Path. In the **EXT** position, the Opto Comp is switched into the Compressor External Path (accessed via rear panel connectors). In the **PST EQ** position, the Opto Comp is inserted into the Main Path after any Equaliser sections that may also be switched into the Main Path.
- **MONO<>LINK** – if using two ARC1.1 Compressors on stereo program material it is best to **LINK** the Sidechains via the rear panel LINK connectors to ensure proper stereo tracking. In **MONO**, the link is ignored, in **LINK** (must be selected on both ARC's) the link is activated.
- **ATTACK** – This switch has 3 positions.
  SLOW – The attack time of the compressor is approx 70mS and varies slightly depending on the amount of gain reduction. The attack time speeds up a little with greater gain reduction. This setting is great for getting extra "front" on a percussive sound.
  AUTO – Attack time of the Compressor is totally dependent on the program material and the amount of gain reduction occurring. This is perhaps the most "musical" mode when using low Ratio settings and could also be named the "Classic Sound" setting.
  FAST – The attack time of the compressor is 1mS or less, and also varies with program material. Percussive signals will provide a further increase in speed, the Opto Compressor will attack faster.

13] Opto Compressor continued…



- **COMP DRIVE** – this control adjusts the amount of signal sent to the Sidechain circuit and therefore the amount of gain reduction applied to the passing audio signal. It is similar to a Threshold control found on other compressors. Clockwise rotation increases the amount of gain reduction.
- **COMP RATIO** – This control adjusts the change in gain reduction for a given change in input signal level. For example, in position 5 a 5dB increase in input signal level will yield a 1dB increase in output signal level, being a Ratio of 5:1. Because the Opto Compressor has a soft knee characteristic, the Ratio value selected is the ultimate achieved and will appear to be softer at threshold. See page 20 for more detail on this subject.
- **RELEASE** – this switch has 6 positions.
  1-2-4-8-16 – sets the approx time it will take for the Compressor to recover from 20dB of gain reduction in 100's of mS. For example, selecting setting 4 means the compressor will take 400mS to return to 0dB gain reduction from an initial 20dB of gain reduction. Note however the actual release time will vary slightly depending on the amount of gain reduction with the Compressor taking longer to recover from heavy gain reduction.
  AUTO – in this position, the release is program dependent. The Compressor will release from fast transient signals quickly, but will maintain a slower release time with more continuous signals.

13] Opto Compressor continued…



- **COMP MAKEUP** – adjusts the gain of the post Compression amplifier to compensate for lost level in the Compression process. Variable from 0dB to +15dB gain.
- **COMP G/R** – this string of LED's to the right of the controls displays the amount of gain reduction imposed onto the audio signal calibrated in dB (decibels). The meter accurately displays the attack and release timing of the Opto Compressor control circuitry enabling the user to get a much clearer picture of gain reduction dynamics than what a conventional mechanical VU meter provides.

14] Tips for Using the Compressor…

Unlike some other audio compressors that are designed to "text book" parameters, the Opto Compressor incorporated into the ARC1.1 possesses a unique "musically transparent" soft knee compression characteristic. It is equally at home providing gentle level control of acoustic sources through to full on squash for "in your face" lead vocals. The following operational tips will help in getting the best out of your unit, but the final word has to be – use your ears!

- **Slow Attack** – The **SLOW** position is useful in obtaining more fronts from percussive sounds. For example, a snare that has no "stick" sound may be improved by gentle compression with a **SLOW** attack and quick (**1** or **2**) release time. The initial transient of the snare is allowed to pass unaffected but compression is applied to the "tail" of the sound, bringing the "stick" forward. For vocal recording, the **SLOW** position will

14] Tips for Using the Compressor continued…

bring forward the beginning of phrases and words, adding "crunch" to the vocal sound, especially useful on male rock vocals.

- **Auto Attack**- Best choice for low ratio light compression of sources like acoustic guitar, bass guitar, vocals and even stereo mixes. The adaptive nature of the **AUTO** position prevents "pumping" and really is the "classic" sound of LDR type compressors.
- **Fast Attack** – Generally, when high ratios and heavy compression are applied to vocals, the **FAST** attack position may be the best choice to prevent sibilance problems. When compressing a stereo mix, the **FAST** position will help control peaks in the program material getting through.
- **Release Timing** – Selecting the most appropriate compressor release time is best achieved by ear but here are some guidelines.

  Vocals & Voice Overs – under heavy gain reduction, 100mS release (number **1** on the dial) will prevent "pumping" and missing phrases due to the compressor still releasing from a loud patch. If using a low ratio and light compression, slower release or the **AUTO** position may sound warmer.

  Bass Sounds – generally speaking a slow release time will provide the least uncoloured sound. Faster release times tend to distort the bass register as the sidechain control circuitry actually starts to track the bass frequencies and "crushes" the sound. However the effect of a fast release may be the sound you're looking for!

  Stereo Mixes – generally a release time of 400mS (number **4** on the dial) or **AUTO** is appropriate, but really does depend on the type of program material. Fast release times will tend to distort the lower frequencies of the mix, if the release time is too slow, then the mix may become "lumpy" due to the slow recovery of gain reduction after a high peak.

  Auto Mode – a clever little circuit inside the ARC 1.1 provides a quick release from transients and peaks whilst maintaining a slow release on overall program. This setting works well with complex program material and vocals, and often is the best place to start in choosing a release time.

- **Using EQ in the Compressor Sidechain** – the ability to easily equalise the sidechain of the ARC Compressor forms a very powerful and creative dynamics tool. The main thing to remember when applying EQ to the sidechain is if you apply a **BOOST** at any given frequency, the Compressor will apply more gain reduction around that frequency, if you apply a **CUT**, then the Compressor will tend to ignore those frequencies. Using the **HIGH PASS FILTER** in the sidechain of the ARC is similar to applying a **CUT**.

14] Tips for Using the Compressor continued…

The possibilities are endless but following is a list of suggestions and possible applications for sidechain equalisation. (Remember that you can monitor the EQ applied to the Sidechain Path with the **S/C MON** switch).

Vocals & Voice Overs – when close miking the human voice, the proximity effect of most microphones can result in a unnaturally bass heavy vocal sound and in some cases, low frequency "pops". If you apply a low frequency **BOOST** (either with the **LOW SHELF** or **MID BAND 1** controls), the Compressor will apply more gain reduction when these frequencies are present, thereby "flattening out" the sound.

If you encounter a voice with excessive brightness, applying a **BOOST** at high frequencies (maybe with the **HIGH SHELF** in **TIGHT** mode) the Compressor will tend to pull back any passages containing the brightness.

If the voice recording contains sibilance, it is possible to operate the ARC1.1 Compressor section as a de-esser. This is best achieved by "tuning" **MID BAND 2** into the problem area, and applying a **BOOST** into the Sidechain Path. To further enhance de-esser action, rolling off the low frequencies (**LOW SHELF** at **120Hz** on full **CUT** or using the **HIGH PASS FILTER** set to **450Hz**) should do the trick. Note that for the Compressor to work as a de-esser, **FAST** attack and fast **RELEASE** (number **1** on the dial) will be required.

If you have a voice where the low and high frequencies could be enhanced or if the voice is very "middy", then applying a broad **MID BAND BOOST** around 1-2kHz into the sidechain may help it.

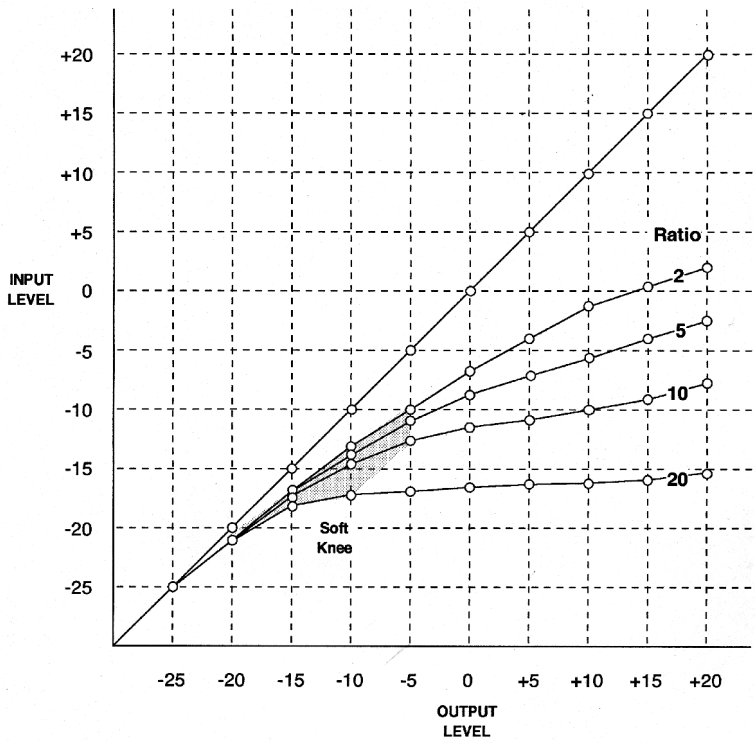
Bass Sounds – often when recording bass instruments wether it be an mic-ed up acoustic bass or DI-ed electric bass, certain notes tend to be louder than others resulting in a "lumpy" sounding track. With the ARC Equaliser it is possible to tune into the notes that are dominant, **BOOST** into the Sidechain thereby applying more gain reduction to level them out.

To emphasise the upper register of a bass sound, apply a **LOW SHELF BOOST** into the Sidechain. To emphasise the lower register, apply a **LOW SHELF CUT** into the sidechain.

Full Mixes – by applying equalisation into the Sidechain of the Compressor, it is possible to change the "spectra" of a full mix. The mix may be too "peaky" in the mid range (snare drum too loud for example) or the mix may only require Compression at low frequencies. All these things are possible to achieve with the ARC 1.1, and by now you should have some idea on how to achieve the results you require.

14] Tips for Using the Compressor continued…

- **Compressor Ratios** – the following graph shows the input to output relationship of the four ratios with the **COMP DRIVE** control set fully clockwise.



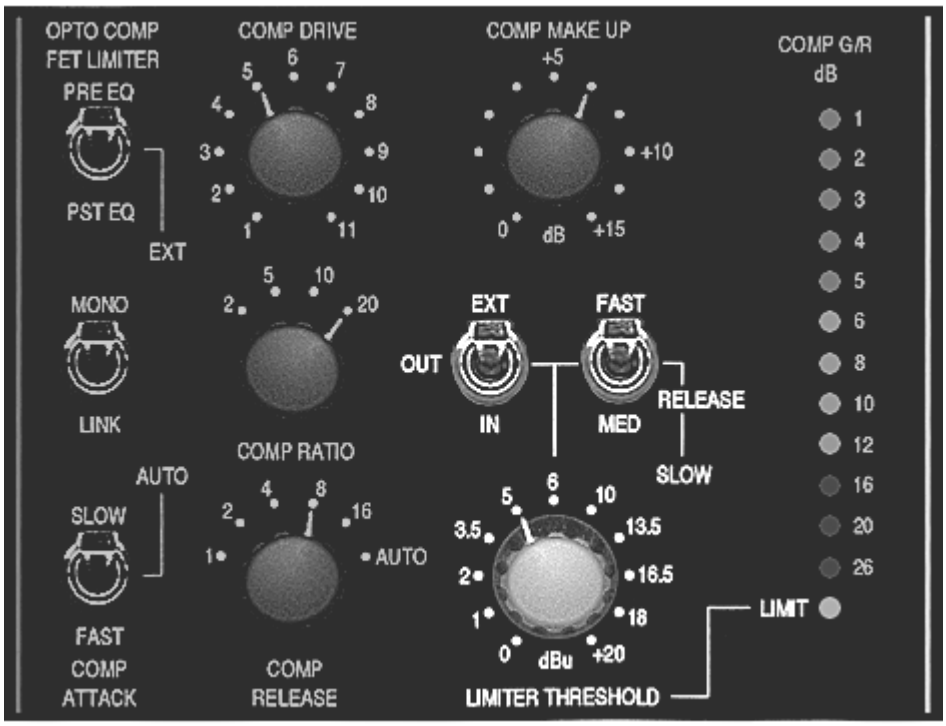
From this graph we see that for lower ratios settings (2:1 and 5:1), the ratio is ultimately achieved only after another further 10dB of gain reduction from threshold, (threshold meaning 0dB gain reduction). For higher ratios, (10:1 and 20:1) the ultimate ratio is achieved within approx. 10dB and 5dB respectively.

This area of the graph represents the "soft knee" of the Opto Comp compression characteristic and is an important feature of its design.

Choosing a low ratio settings for gentle control of level and thereby increasing the loudness of a sound within a mix is possible without signal degradation and minimal colouration. On the other hand, with higher ratio settings, a sound may be "clamped" or levelled whilst still retaining some audible dynamic.

Note that the 20:1 ratio setting is virtually hard limiting in that a 20dB input change only results in a 1dB output change.

15] Peak Limiter…



- **EXT<>OUT<>IN** – the function of this switch is a bit complicated so please read this carefully – hopefully it will make sense to you!

  When selected to **EXT**, the Peak Limiter is switched into the Compressor External Path and can be used separately. However, if the Opto Comp is also switched to **EXT**, then the Opto Comp and Limiter are both in the Compressor External Path with Limiter following (post) the Comp in the chain.

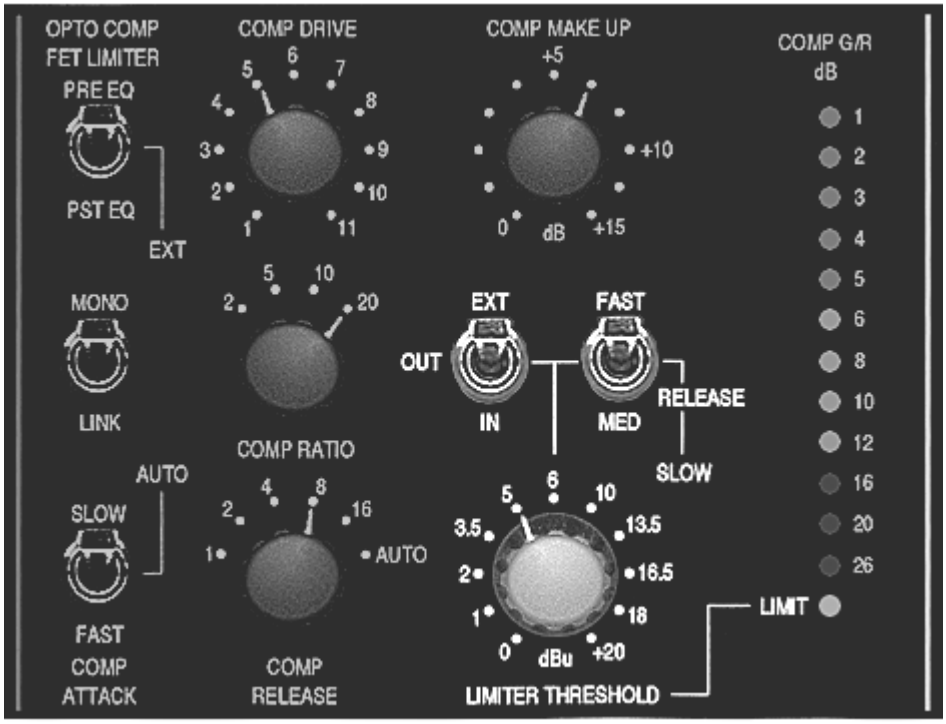
  When selected to **OUT**, the Limiter is in no Path and does not function. That was easy!

  When switched to **IN**, the Peak Limiter is in the Main Path, always post the Opto Comp but **PRE** (before) or **PST** (after) any Equaliser sections that may also be in the Main Path, as selected by the **PRE EQ<>EXT<>PST EQ** switch. In other words, it travels with the Comp section. If the Opto Comp is switched to **EXT** and you switch the Limiter **IN**, then the Limiter will still be in the Main Path but always after (post) any Equaliser sections also in the Main Path.

- **RELEASE FAST<>MED<>SLOW** – selects the recovery time of the Peak Limiter once the signal has fallen below threshold. The actual recovery time from 10dB of gain reduction is as follows;

  FAST = 100mS (0.1 second)
  MED = 750mS (0.75 seconds)
  SLOW = 2000mS (2 seconds)

15] Peak Limiter continued…



- **THRESHOLD** – sets the point at which limiting action will occur. This control is calibrated in dBu and signals above this level will not exist, that is, the audio is "clamped" at this point.

- **LIMIT LED** – limiting action is indicated by a single red LED which will light once Threshold has been exceeded by the audio signal and extinguish when the Limiter has released fully from gain reduction.

16] Tips for Using the Peak Limiter…

Intended Purpose - The Peak Limiter incorporated into the ARC has been designed primarily to "catch" short duration audio peaks and remove them with little colouration to the signal. When used in conjunction with the Opto Compressor, the Peak Limiter (always post the Compressor) has the ability to catch transients missed by the slower optical action thereby providing a near zero attack time compressor system.

When used in this fashion, the best setting for the **RELEASE** control is **FAST**, so that the Limiter will recover quickly to avoid any audible "pumping" effects.

The attack time of the Limiter is not adjustable and for all intents and purposes is instantaneous. The ratio is infinity to 1 - signal level cannot exceed the Threshold point.

16] Tips for Using the Peak Limiter continued…

Heavy Limiting Effects - The Limiter can also be used for special effects where the sound of a heavily limited signal may be desirable and slower release times (**MED** and **SLOW**) have been provided for this application. When limiting low frequency signals (bass), the **FAST** position will generate quite a lot of audible distortion, an effect that can be used creatively, but if you want to keep it clean, then **MED** or **SLOW** would be better choices.

Stereo Operation - if you have two ARC1.1 units and are using them in stereo, note that the Peak Limiter sidechains are NOT linked via the **COMPRESSOR LINK** connector. Whilst this may at first appear to be an oversight in design of the ARC, in practical use it turns out that the Limiters function better if left in dual mono mode bearing in mind the intended purpose of catching short duration peaks.

If you need to heavily limit a stereo mix, use the Opto Compressors (linked) with a **RATIO** of **10:1** or **20:1**, followed by the Peak Limiter with the **THRESHOLD** set to catch any residual transients with the release set to **FAST**. Because the action of the Limiter is so quick, there is no audible shifting of the stereo image.

Designers Note
The ARC 1.1 Peak Limiter uses a very special Field Effect Transistor (FET) as the gain reduction element in that the device has been specifically constructed to act as a variable resistor whilst most other FETs are designed to be amplifiers or switches. The sidechain circuit of the Peak Limiter uses a combination of feed-forward and feed-back topology's along with very high speed signal processing to achieve the near zero attack time. Whilst the resulting circuitry is quite complex, the action of the Peak Limiter is quite transparent when used to limit short duration peaks that may otherwise cause overs on digital recording systems.

17] Compressor/Limiter Glossary…

Attack – the time taken for a compressor or limiter to reduce it's gain in response to an increase of signal level.

Compressor – a device designed to reduce the dynamic range of audio signals by reducing the level of high signals or by increasing the level of low signals.

De-esser - a device for reducing the effect of sibilance in voice recordings.

Gain – the amount of positive amplification applied to the audio signal, usually expressed in decibels.

Gain Reduction – the amount of negative amplification imposed on the audio signal.

Limiter - a device that controls the gain of a signal so as to prevent it from ever exceeding a preset level. A limiter is essentially a fast acting compressor with an infinite compression ratio.

Peak – the maximum instantaneous level of an audio signal.

17] Compressor/Limiter Glossary continued

Ratio – the change in output signal level for a given change of input signal level through a compressor or limiter. A true limiter has an infinite ratio.

Release - the time taken for a compressor or limiter to recover from a specified amount of gain reduction.

Sidechain – a circuit that splits off and processes the audio signal to form a suitable signal to control the gain reduction element in a compressor or limiter.

Threshold – the audio level at which a compressor or limiter will begin to impose gain reduction on the passing audio signal.

18] Level Meter…



- On the far right hand side of the ARC front panel resides the **LEVEL METER**. The meter is calibrated in dBu and indicates peak level with a fast attack and fairly slow decay time, similar to a Peak Program Meter. The **OVER** LED monitors audio levels at critical points throughout the ARC circuitry and will light if any level exceeds +20dBu, indicating a possible clipping situation. Note that even if the Equaliser and Compressor are switched into their External Paths, this circuit is still monitoring them.

- **INPUT<>OFF<>OUTPUT** – when switched to **INPUT**, the level meter will display the signal level of the mic or line input, as selected by the **MIC<>LINE** switch.

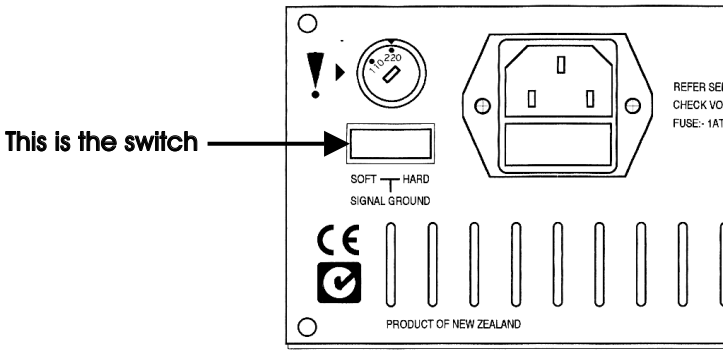
In the **OFF** position, well it's off, it displays nothing, which is handy if you'd prefer to have not too many flashing lights. Note that this does not turn off the **OVER** LED.

In the **OUTPUT** position, the meter displays signal level leaving the Main Output connector, post the **OUTPUT GAIN** control.

- **POWER<>ON** – this switch turns **ON** and **OFF** the mains power to the ARC 1.1. Note that because the ARC is manufactured in New Zealand which is in the Southern Hemisphere, down position is **ON** as opposed to equipment manufactured in the Northern Hemisphere, where normally the up position is **ON**. Just in case you maybe confused by this, a yellow LED above the switch should light when you've got it right.



19] Signal Ground Hard/Soft Switch…



- **SIGNAL GROUND HARD<>SOFT** - this rocker switch on the rear panel selects **HARD** or **SOFT** internal signal grounding. In the **HARD** position, the ARC 1.1 internal signal ground is connected directly to the mains earth and chassis. In the **SOFT** position, the signal ground is lifted from the mains earth via a 10 ohm resistor. Normally the **SIGNAL GROUND** switch should be set to **HARD**. If you experience problems with "hum" when connecting the ARC 1.1 to other devices, try the **SOFT** position.

  Please Note - The mains earth to ARC 1.1 chassis connection is unaffected by this switch. Also, the Chassis Ground connection to the input and output connectors is unaffected by this switch.

20] Specifications…

Main Path Frequency Response (measured without EQ, Comp/Limiter or Tranny switched IN)
Mic Input to Mic Direct Out .....3Hz to 400kHz (-3dB)
Mic Input to Main Out.....4Hz to 400kHz (-3dB)
Bal Line Input to Main Out .....3Hz to 250Khz (-3dB)
Bal Line In with Tranny selected..... 12Hz to 100kHz (+/-1dB),+5 @ 250Khz, -3dB @ 400Khz
Instrument Input to Main Out .....9Hz to 300kHz (-3dB)

External Path Frequency Response
Compressor External Path (no Limiter) .....5Hz to 250kHz (-3dB)
Compressor External Path (with Limiter) .....5Hz to 80kHz (-3dB)
Equaliser External Path (with all EQ zeroed and no HPF).....5Hz to 200kHz (-3dB)

Total Harmonic Distortion
Mic Input to Mic Direct Out .....100Hz=0.005% 1kHz=0.005% 10kHz=0.015%
Compressor External Path (no Limit, no GR).....100Hz=0.01% 1kHz=0.004% 10kHz=0.006%
Compressor External Path (with Limit, no GR).....100Hz=0.01% 1kHz=0.01% 10kHz=0.02%
Compressor External Path (10dB of GR @ 20:1, Release at16, +10dB Make Up Gain).....
.....100Hz=0.15% 1kHz=0.15 10kHz=0.18%
Limiter in External Path (10dB GR, Release Slow).....100Hz=0.2% 1kHz=0.08% 10kHz=0.18%
Bal Line Input to Main Out .....100Hz=0.0015% 1kHz=0.001% 10kHz=0.001%
Bal Line Input to Main Out with Tranny.....100Hz=0.3% 1kHz=0.05% 10kHz=0.015
Instrument Input to Main Out .....100Hz=0.08% 1kHz=0.025% 10kHz=0.025%

20] Specifications continued…

Noise (all measured A weighted ref to 0dBu=0.775mV RMS)
Mic Input (65dB gain, 150 ohm source impedance).....-71dBu (EIN 132.5dB)
Bal Line In (10dB gain, 150 ohm source impedance).....-95dBu
Unbal Line In (10dB gain, 0 ohm source impedance).....-100dBu
Compressor External Path (no Limiter) .....-90dBu
Compressor External Path (with Limiter) .....-80dBu
Equaliser External Path (no cut/boost).....-90dBu

General
Power requirements..... 115V/230V selectable on rear panel
Dimensions .....2U rack mount, 300mm deep

21] Warranty/Service Information.

We are confident that you will receive many years of trouble free operation from your unit. If however you experience any technical problem with your ARC 1.1, contact your dealer or Buzz Audio for recommendations on what to do. The modular nature of the ARC construction means most electronic faults can be easily repaired by swapping circuit boards.

For on line support visit our web site; www.buzzaudio.com

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New Zealand. Voice/Fax 64+4+385-2478.
Email; support@buzzaudio.co.nz

- **Disclaimer**

Buzz Audio is not liable for any damage to microphones, amplifiers, consoles, speakers or any other equipment and/or electric shock to humans that is caused by negligence or improper installation and/or use of the ARC 1.1 Analog Recording Channel.

- **Product Warranty**

Buzz Audio guarantees the ARC 1.1 Analog Recording Channel to be free of defective materials and/or workmanship for a period of 2 years from the date of sale, and will replace defective parts and repair malfunctioning products under this warranty when the defect occurs under normal installation and use – provided the unit is returned to our factory (or duly authorised service centre) via prepaid transportation with a copy of the proof of purchase, ie, sales receipt. This warranty provides that examination of the returned product must indicate, in our judgement, a manufacturing defect. This warranty does not extend to any product that has been subjected to misuse, neglect, accident, improper installation, or where the date code has been removed or defaced.