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Audio dynamic range compressor design.

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Problem.

There have been many different designs of audio compressor used for both ham radio and broadcasting.

Mostly the designs related around a degree of audio clipping, to then filter the distortion from the audio signal transmission.

An alternative to the clipping compressor, is the auto matic level gain microphoen amplifier. This design has its problems, in that the background noise level falls in and out as the microphone amplifier alters its gain according to the microphone input audio level.

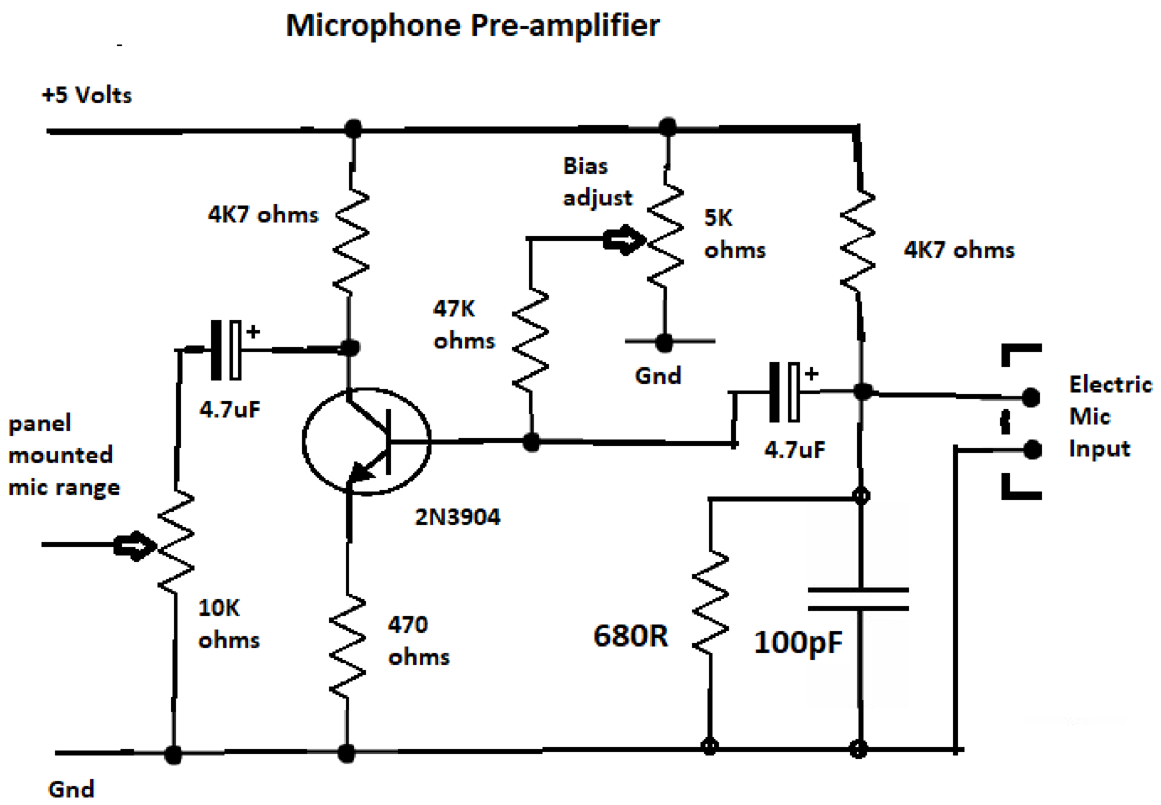
A use of a noise gate is general adopted, but this can lead to a clipped audio as the gate opens up late, in accordance to the spoken voice timing.

Solution.

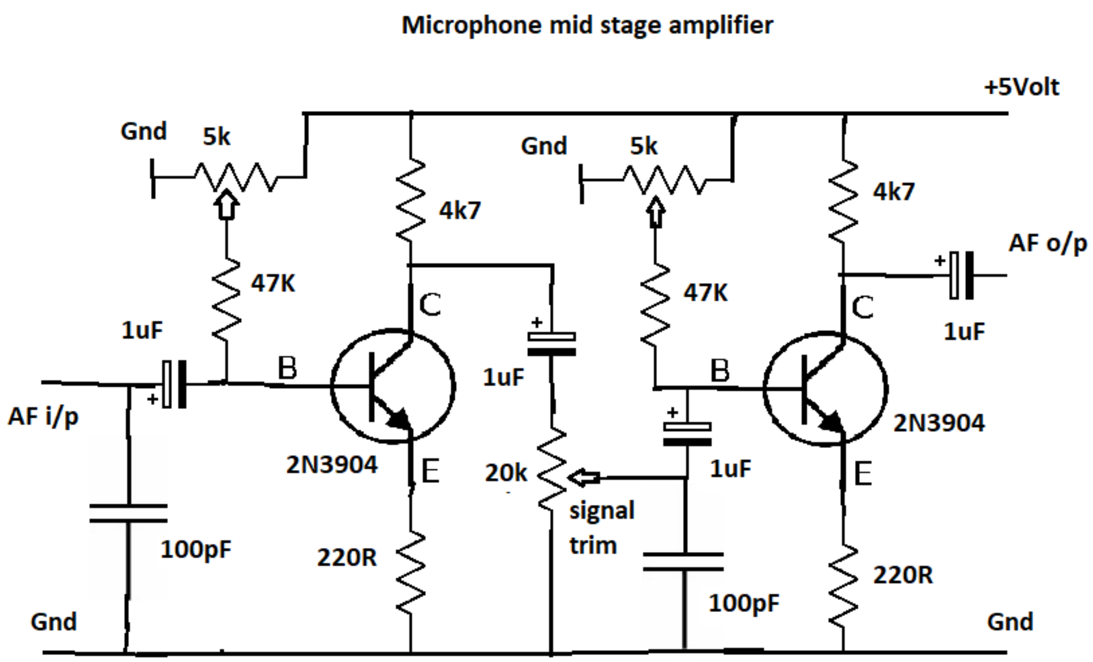
Input and Output Circuits.

Perhaps the microphone audio gain could have a manual adjustment to allow a set to a pre-determined distance away from the microphone. Once set, the spoken voice away from the microphone would not then peek the transmitter signal, but if closer than the operating distance from the microphone, would then overload the transmitter output.

The first thing that was required is a microphone pre-amplifier stage, an amplifier with an adjustment mic range setting, shown below:



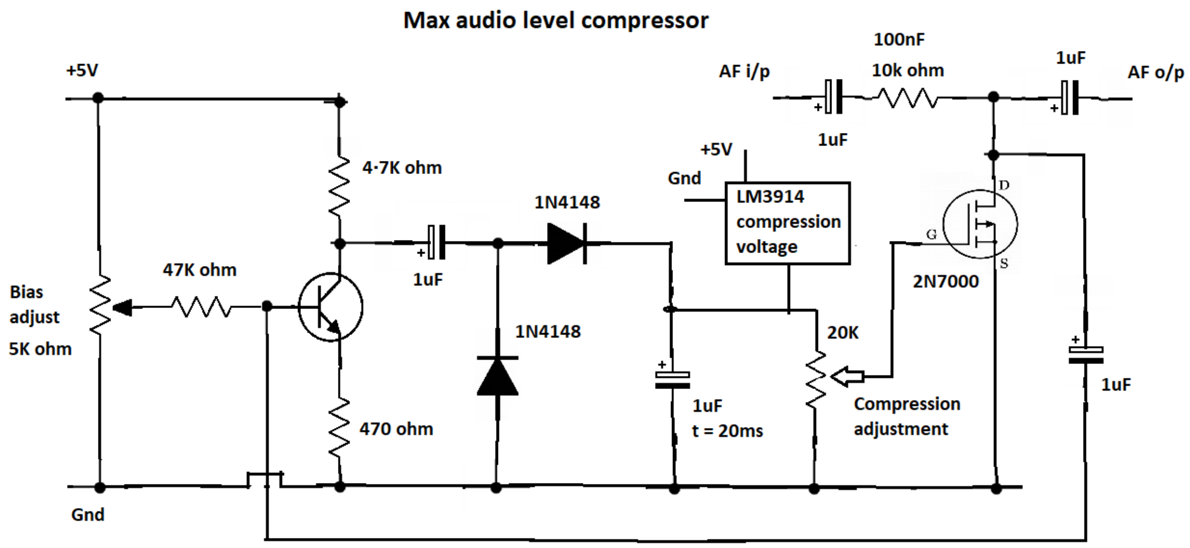
A mid stage audio amplifier was found to be required, so the below circuit has been constructed for the microphone audio amplification, a mid stage amplification trim is used, setting the overall max level of voice audio signal gain, shown below:



The emitter circuit DC bias setting across the emitter resistor, does not use a by-pass capacitor for an AC signal gain setting. It was found that the capacitive reactance of the emitter capacitor, shunted the DC bias current, thus causing audio distortion within the audio signal. The higher the sound signal frequency, the worst the problem became.

By changing the emitter capacitor, the problem varied, but the signal gain due to the capacitor reactance kept on varying across the audible sound range. The solution was to only use the DC bias setting as the amplifier circuit signal gain, and in the process avoid any further problems.

A circuit that reduce the voice audio level would be required, the “max audio level compressor”.



A max audio level compressor, once the spoken voice audio peek above a minimum limit, the automatic gain control adjustment will reduce the audio amplitude, by virtue of the 20k ohm variable and the mosfet 2N7000, half “T section” attenuator circuit.

An in-cirucit trim or a panel mounted control could either be used, thus providing either an in-circuit or panel mounted compression control adjustment.

The AGC time constant would as be prefered, shown here as a 20ms time constant, the principle to follow the voice signal itself to keep the overall voice AC signal to within a set limiting level.

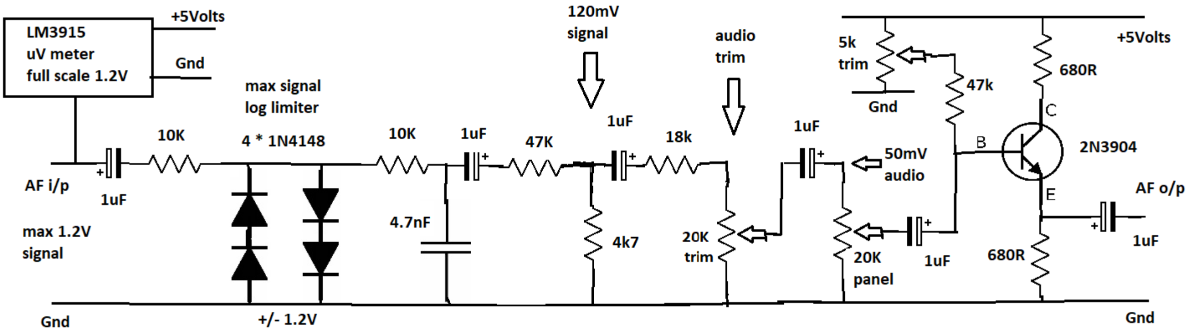
The controlled level of output audio is relatively held within a +/- 0·7V limit, but high speech volume can increase the level up to +/- 1·2V signal. To cater for a high audio volume, a double “back to back” diode limiter is used, interfaced with 10K resistors to give a similar logarithmic characteristic attenuation curve.

Note the LM3914 circuit box, this to visualise the AGC or compression generated voltage.

The spoken voice audio content below the limit threshold, is not afffected by this circuit, compression is only added when the audio level is breached.

The audio output stage to return the microphone signal back to the radio mic input, is shown below.

Note the LM3915 circuit box, this to visualise the “uV meter” audio level, into the output stage interface circuit.



Audio output stage to radio mic input

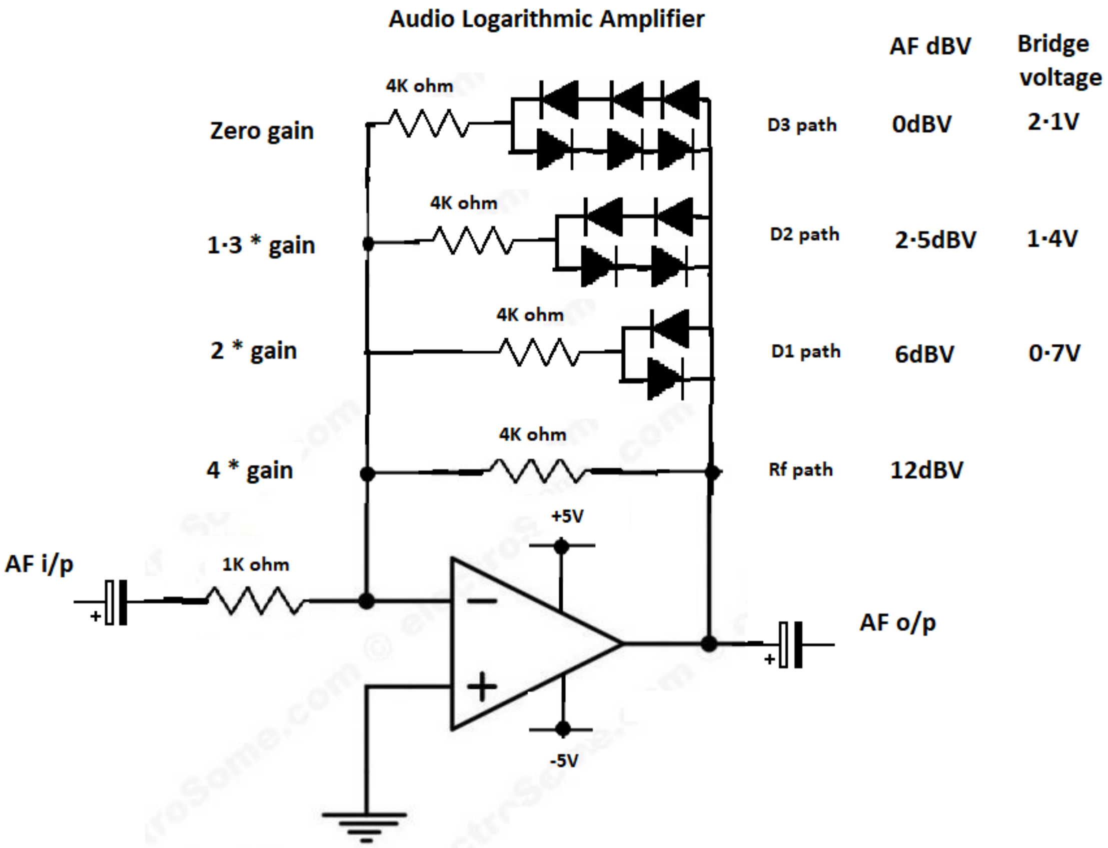
Using the above circuits, an audio compressor circuit can be simply constructed. I have completed my own construction, just using the 5Volt supply from my FT450d microphone connection.

This has then allowed me to just connect the audio compressor as an inline unit, running from the ethernet connected microphone connection, with the mic pluged into the voice conditioner box.

Sorry about the smallness of the output stage circuit diagram, but to overcome this snag, I have used a pdf document file to allow the diagrams to enlaregd in size to make things easier to understand.

Base level signal boost

The diagram below, the "audio logarithmic amplifier", is a logarithimic amplifier design to provide a logarithmic amplification to the lower level audio signal. Once the input audio signal reaches a set level, the opamp circuit gain self adjusts to reduce the audio signal gain, providing in principle a compession curve to the output audio relative to the input audio signal.



As each bridge voltage is breached, an addition feedback resistor comes into play, creating a parallel resistance with resistor "RF", the 4K ohm resistor to give the four times original signal gain.

By changing the new bridge voltage resistor feedback values, a compression curve of one choosing can thus be created.

Expanding the number of feedback diodes with increasing bridge voltages, a greater number of compression curve reference points can thus be created. The expander or decompression curve would have the same number of bridge voltage levels. This would then create an even transfer of the overall sound quality.

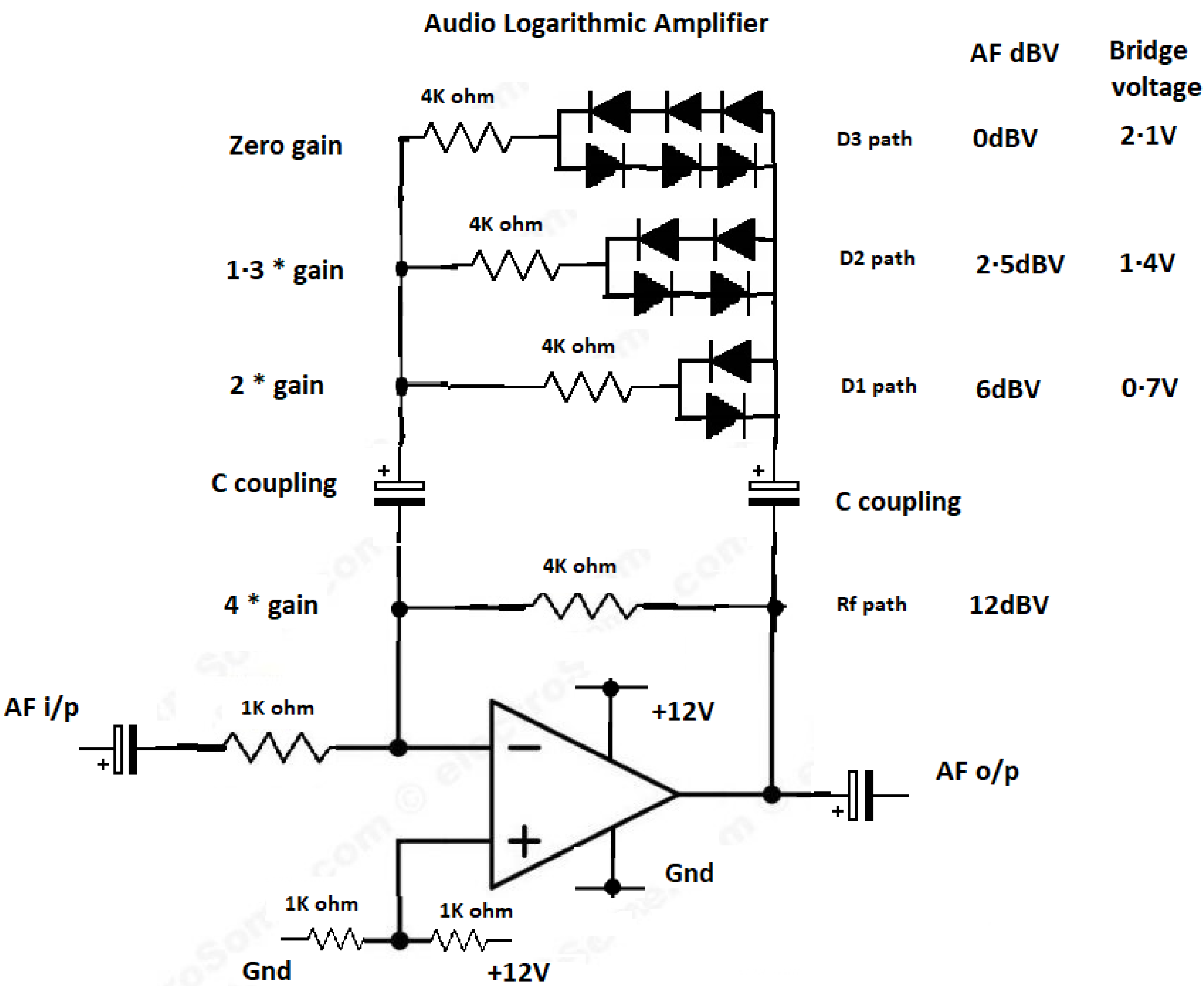
For completeness of the combined audio compression circuit, and an additional max audio level compressor could be added after the audio logarithmic amplifier, to in-sure that the input audio into the radio does not overdrive the radio transmitter.

An addition to the overall application, maybe include a parametric amplifier, between the "audio logarithmic amplifier", and the second "max audio compressor". As the parametric amplifier peeks its output, the second max audio compressor would contain the maximum peek audio level output to the radio transmitter microphone input.

The “dynamic range compressor”, would in theory just require a “mic distance” adjustment for the microphone sensitivity, as well as a “audio drive” adjustment into the radio. The parametric amplifier adjustments would as per-design.

By the way, I have built the compressor unit, apart from a parametric amplifier construction fault, the dynamic range compressor works quite well indeed.

To use the “audio logarithmic amplifier” with a single supply, two additional capacitors need to be added into the diode circuit feedback, shown below.

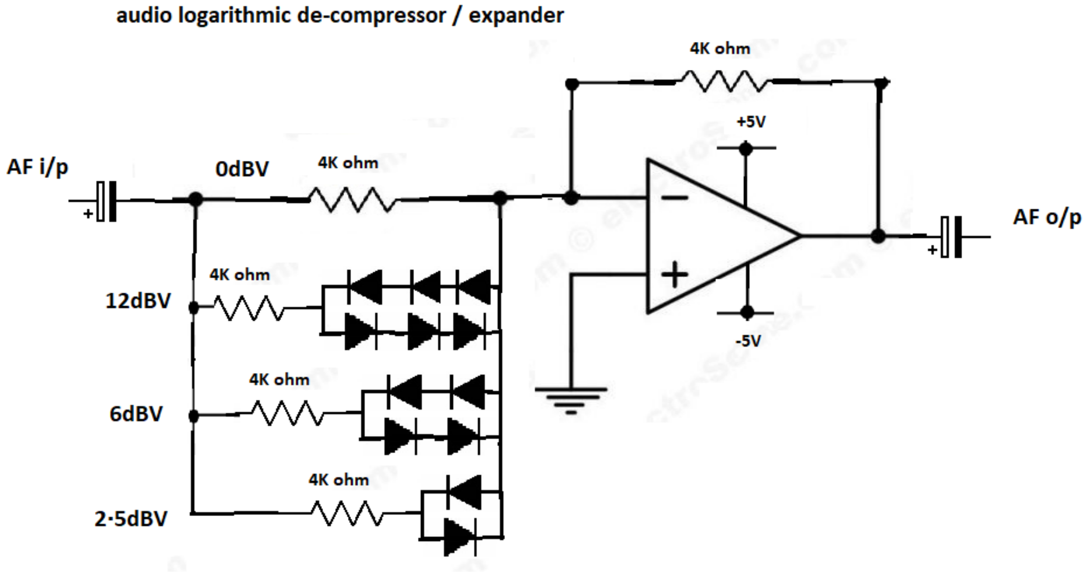


The addition of the “C coupling” capacitors, is to provide a ground voltage reference for the diode feedback circuit to function.

The value of “C coupling” would be best suited with a low capacitive reactance, to the audio bandwidth range from the microphone, at the lowest audio frequency response.

In case anyone is wondering, I did also twig how to design a log amp de-compressor circuit as well.

The illustrated diagram below, shows the de-compressor or expander circuit implementation.



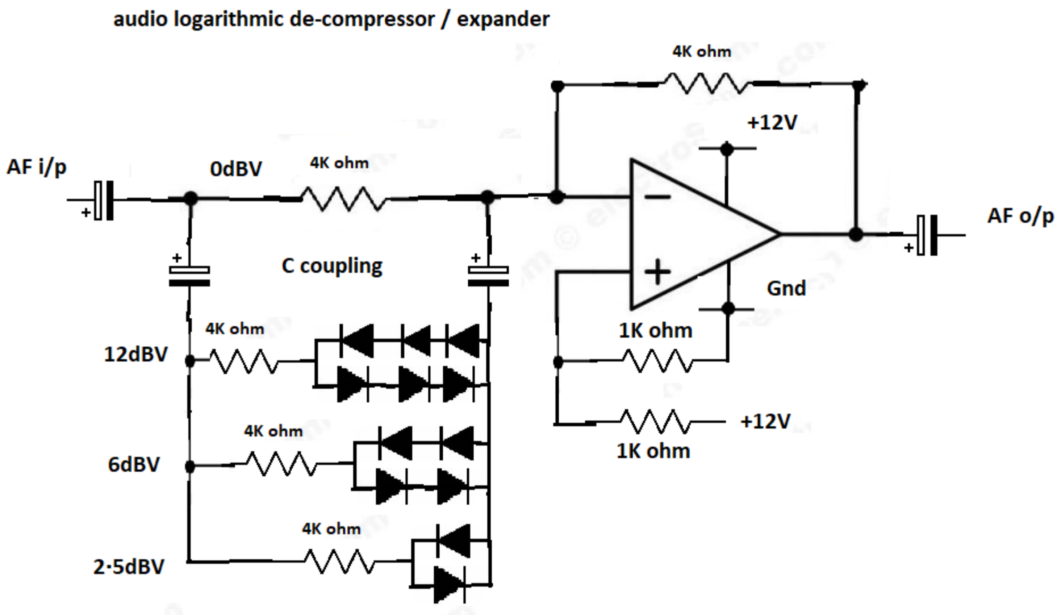
For the logarithmic expander circuit, it is perhaps best to use Shockley diodes due to their low forward voltage, otherwise referred to as the "barrier potential" voltage.

The reason is that the 12dBV gain input circuit amounts to a forward voltage of 2·1 Volts, using silicon diodes, the 12dBV audio signal circuit gain, relates to an output voltage signal of 8·4 Volts.

With Shockley diodes, the combined input threshold of 0·6 Volts for 12dBv gain, relates to a 2·4Volts op-amp output signal.

Using a low supply voltage, the Shockley diodes maybe perhaps the preferred option.

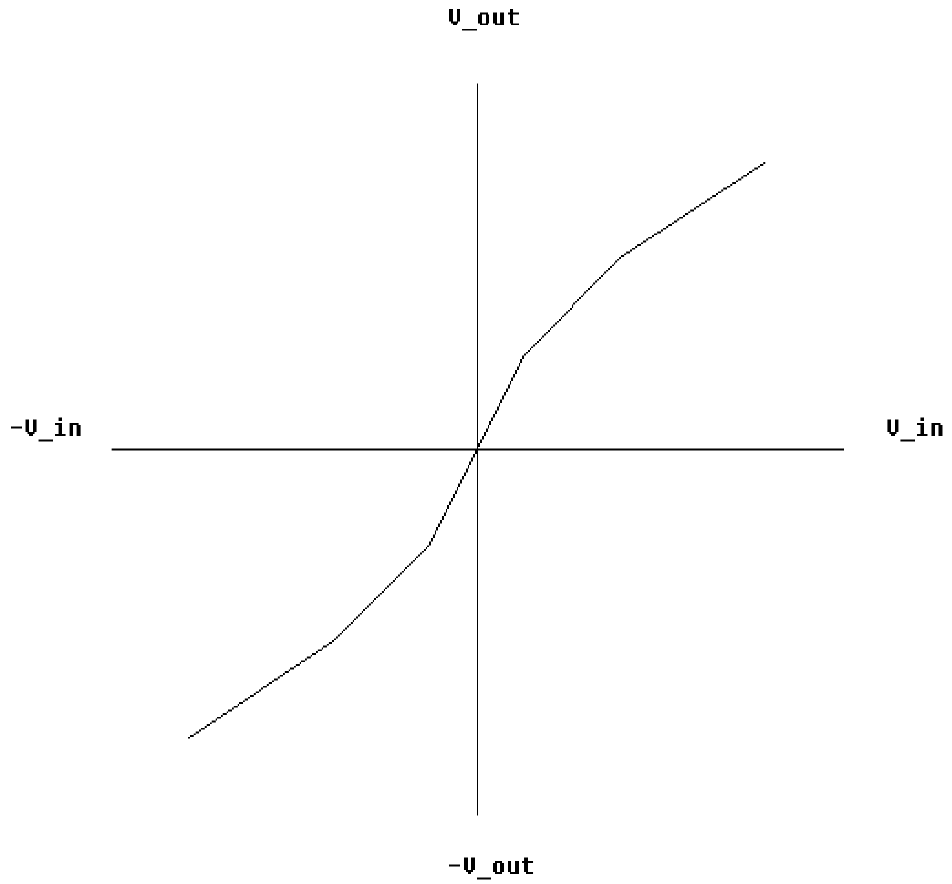
Below, illustrates a single supply voltage circuit version of the logarithmic expander.



Now to be honest, while yes, I had to redesign the logarithmic circuit, and behind the scenes to this design are the mathematical models to prove the compression curve, using both just the op-amp gain equation.

Now for the bummer, I seen to recall the same log amp circuit was used in a “Sony Betamax HiFi video tape recorder”.

However, the transfer function for the three diode feedback compression curve, is shown below, “diagram 5”.



The mathematics calculation code for the compression curve function, is shown in appendix “A”.

If the signal breaches the 0dBV gain, the transfer curve goes at a 45 degree tangent, thus the max audio compressor circuit would overcome this snag, thus top limiting the tangent curve signal gain control.

If anyone does remember, as I did so, a similar circuit is used within a “Sony HiFi Video tape recorder player”.

Appendix “A”.

```
 10
 20 R_in = 1000
 30 PROC_horz
 40 PROC_vert
 50
 60 MOVE 800,800
 70 FOR V_out = 0 TO 2.1 STEP 0.7
 80   PROC_amp
 90 NEXT V_out
100
110
120 MOVE 800,800
130 FOR V_out = 0 TO 2.1 STEP 0.7
140   PROC_amp_neg
150 NEXT V_out
160
170 END
180
190
200 DEF PROC_amp
210 IF V_out < 0.7 THEN R_fb = 4000
220 IF V_out => 0.7 THEN R_fb = 2000
230 IF V_out => 1.4 THEN R_fb = 1333
240 IF V_out => 2.1 THEN R_fb = 1000
250 V_in = ( R_in / R_fb ) * V_out
260 PLOT 5,(V_in * 150)+800, (V_out * 150) + 800
270 ENDPROC
280
290
300 DEF PROC_amp_neg
310 IF V_out < 0.7 THEN R_fb = 4000
320 IF V_out => 0.7 THEN R_fb = 2000
330 IF V_out => 1.4 THEN R_fb = 1333
340 IF V_out => 2.1 THEN R_fb = 1000
350 V_in = ( R_in / R_fb ) * V_out
360 PLOT 5,800 - (V_in * 150), 800 - (V_out * 150)
370 ENDPROC
380
390
400 DEF PROC_horz
410 PRINT TAB(48,5);"V_out"
420 PRINT TAB(48,36);"-V_out"
430 MOVE 800,800
440 DRAW 800,1200
450 MOVE 800,800
460 DRAW 800,400
470 ENDPROC
480
490 DEF PROC_vert
500 PRINT TAB(78,20);"V_in"
510 PRINT TAB(18,20);"-V_in"
520 MOVE 400,800
530 PLOT5,1200,800
540 ENDPROC
```