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Audio dynamic range compressor design.

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Updated article.

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.

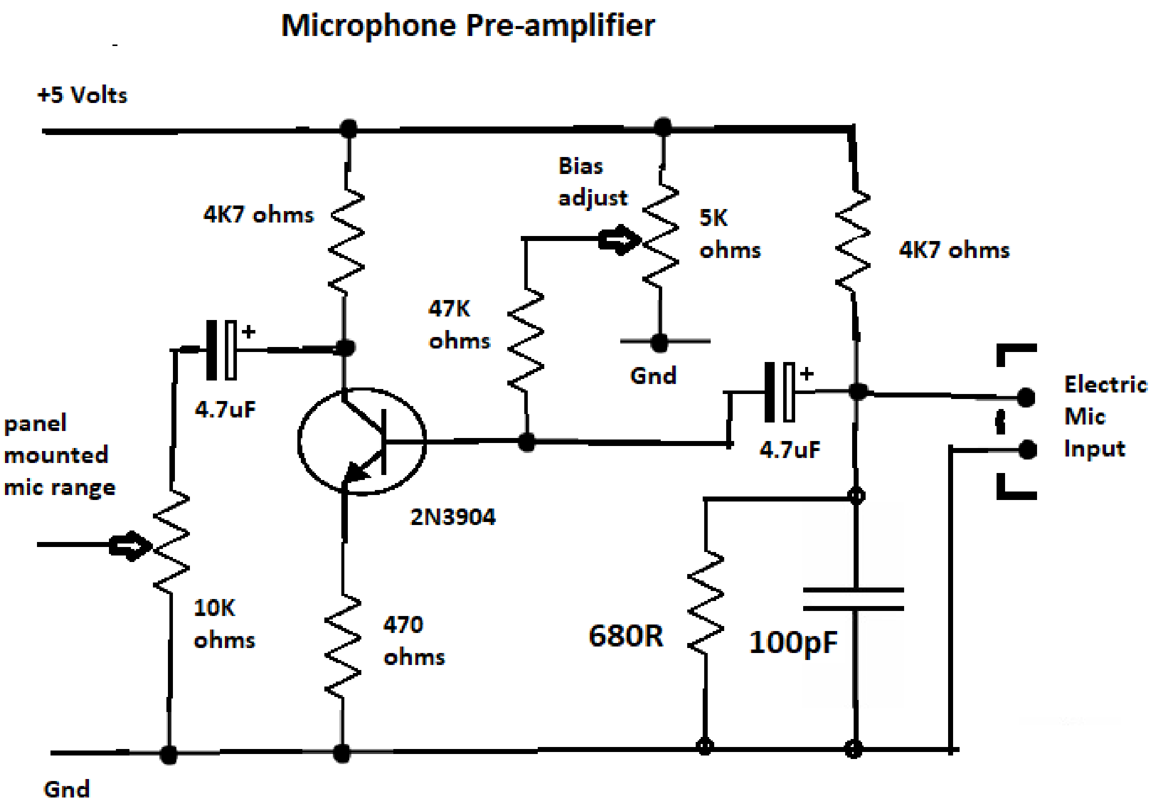
A use of a noise gate is general adopted, but this can lead to a clipped audio as the gate opens up late, in accordance to the spoken voice timing.

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:

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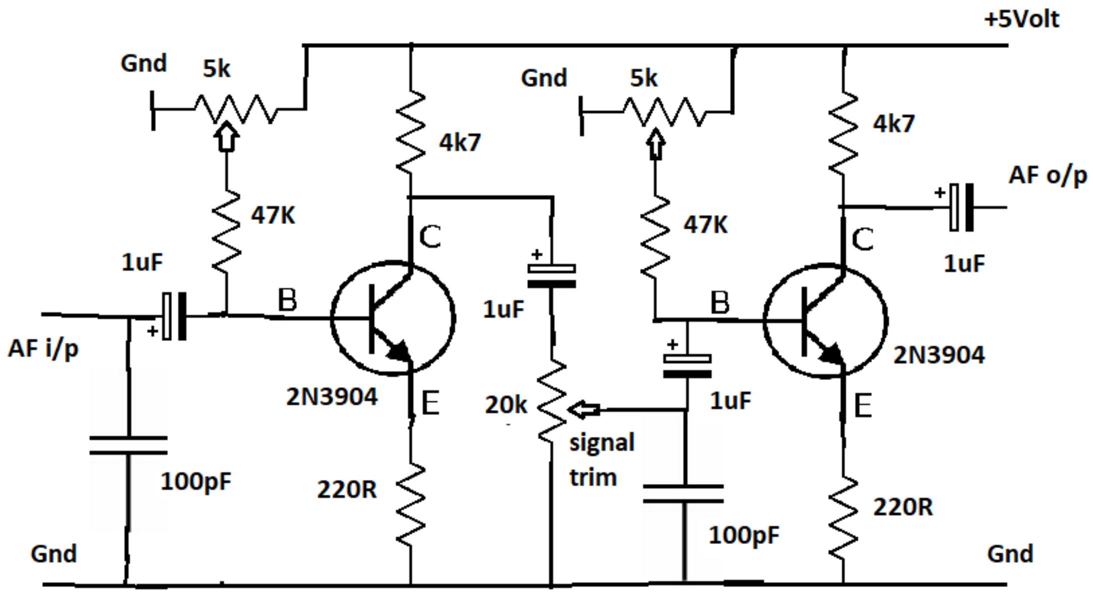
Microphone mid stage amplifier



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”.

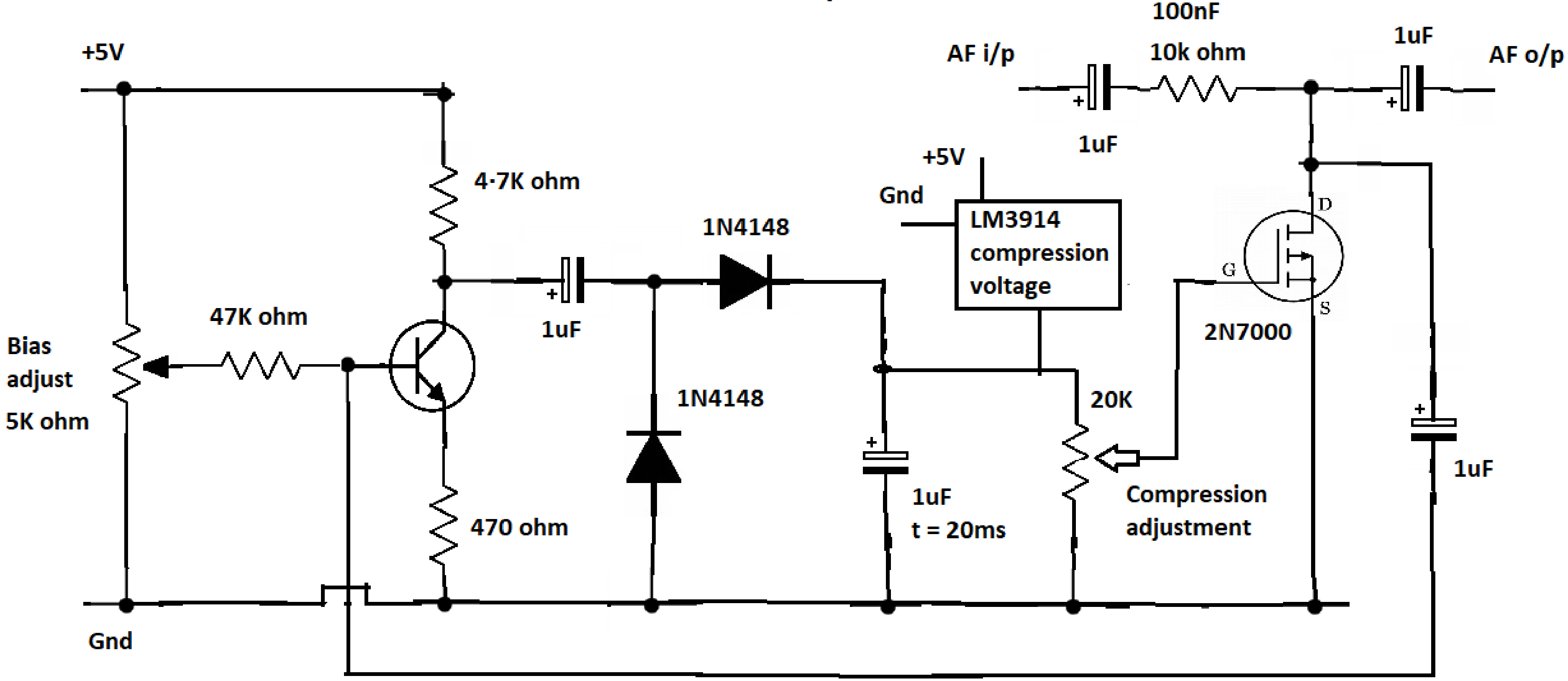
A max audio level compressor, once the spoken voice audio peek above a minimum limit, the degree of automatic gain control adjustment will reduce the audio amplitude compression, by virtue of the 20k ohm trim variable and the mosfet 2N7000, half “T section” attenuator circuit.

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 longer the time constant, the less lilely the agc will be able to directly follow the peeks of the voice signal. A long time constant will follow the voice envelope, but here it is needed to follow the voice sinewave as to the envelope. However, at the end of the day, the time constant used will be a comprimise.

Max audio level compressor



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 over-shoot, 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.

If the spoken voice audio content below the limit threshold, the audio is not afffected by this circuit, forced compression is only added when the audio level has breached the +/- 1·2V signal limit.

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

However, if as I have found that the out-going audio is a bit small, so some additional gain from the output of the circuit below could be useful. The alter the gain, reduce the emitter resistor value to say half, from 680ohms to say 330 ohms. This circuit output is used as an emitter follower circuit design.

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

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 type connected cable as the microphone connection, with the mic pluged into the voice conditioner box.

The article has been filed as a pdf document, which will then allow the largement of the text and drawings by the usual way with a pdf document file.

In the next section, are designs for a “S curve” compression circuit. Now I have used built these circuits, and it took part within the original design construction. I found that the overall sensitivity of the complete project was fab, even brill, even so as the radio PSU fan jumped into life, the base line sensitivity of the “S curve” compression gain transfer curve, was a bit too sensitive.

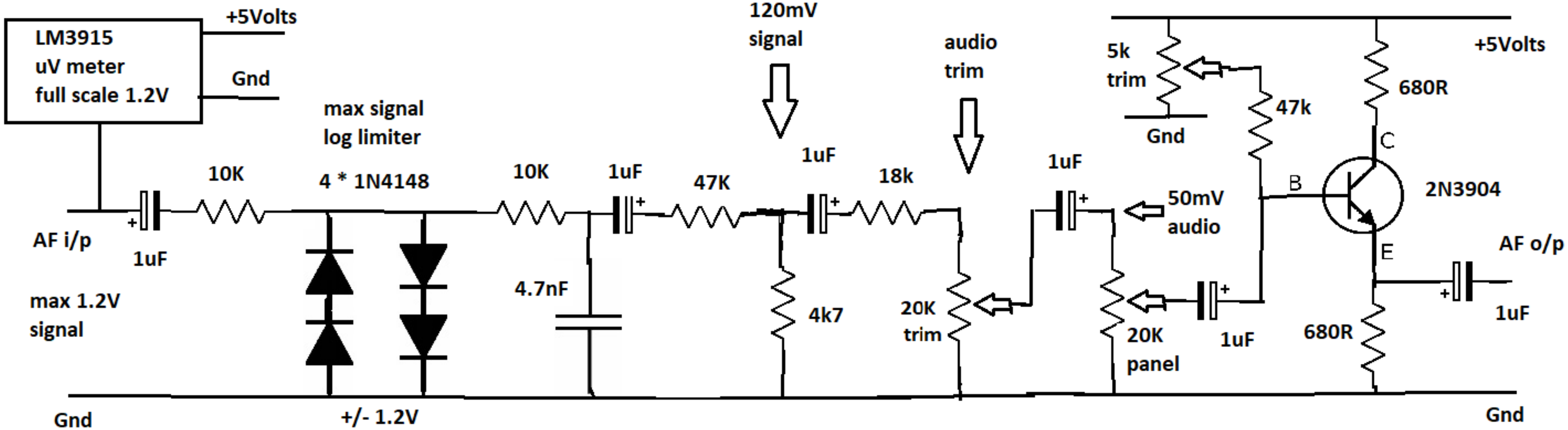
However, studio wise, it would be, just brill.

I came to the conclusion, that the “S curve” designs could be made human ear sensitive, would be just brill for the movie industry, as I came to the end thought, that the “S curve” compressor with its up to three diodes “back to back” design, was on the way to be human ear sensitive.

Adding even more diodes, say a four “back to back” diode network, would also improve the circuit object of the overall design.

The LM3914 of the agc circuit, uses a linear measurement, but the LM3916 uses a “UV” measurement. In both cases, from either Amazon or from Ebay, bar graph meter purchase was constructed with a 5 Volt supply line in mind.

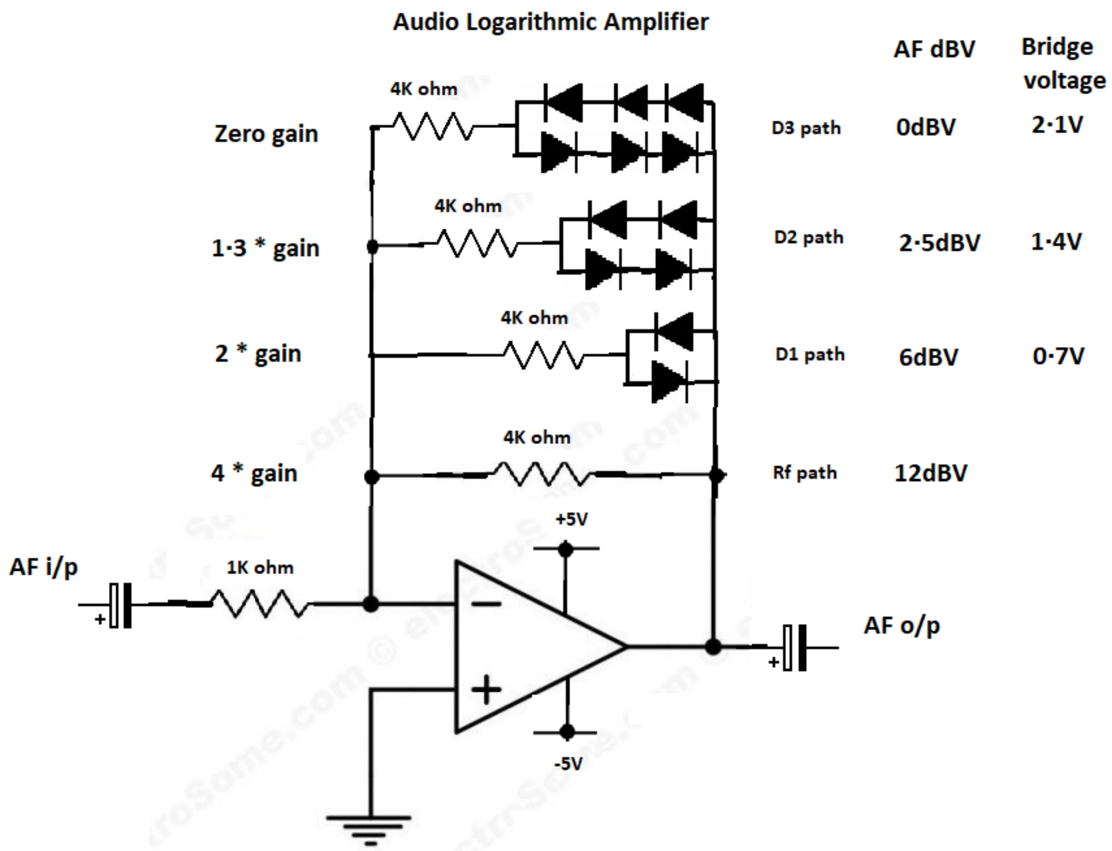
The on-board variable trimmer would scale the measurement signal amplitude. For the LM3914, the on-board trimmer resistor, would affect the time constant, so the AGC timing capacitor, may needed to adjusted in value.



Audio output stage to radio mic input

Base level signal boost "S curve"

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. The curve can be adjusted by replacing the 1Kohm with a variable adjustment. The greater in resistance is the 1Kohm input resistor, the less acute will be the compression curve.

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.

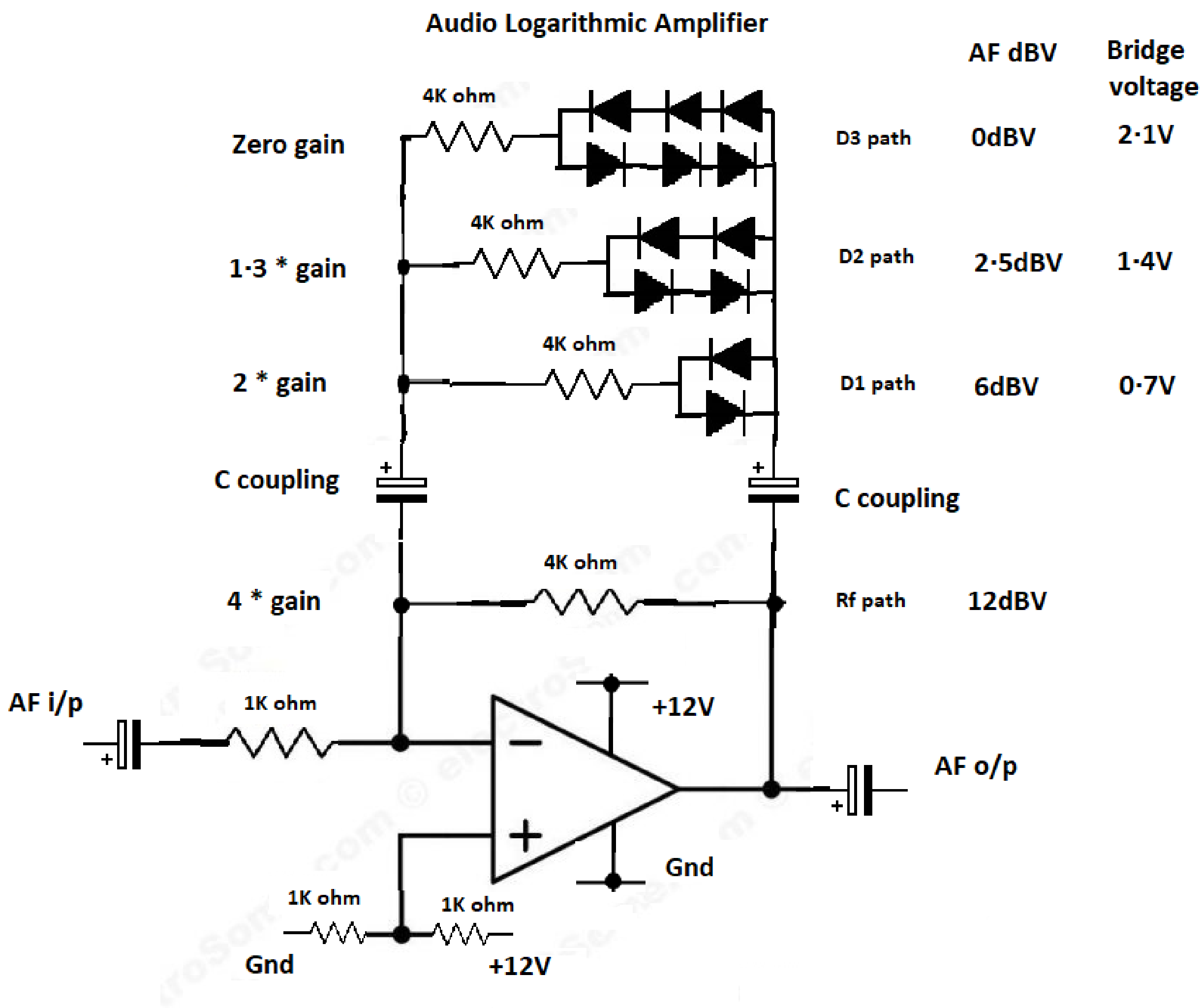
For completeness of the combined audio compression circuit, and an additional max audio level compressor could be added after the audio logarithmic amplifier, to in-sure that the input audio into the radio does not overdrive the radio transmitter.

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.

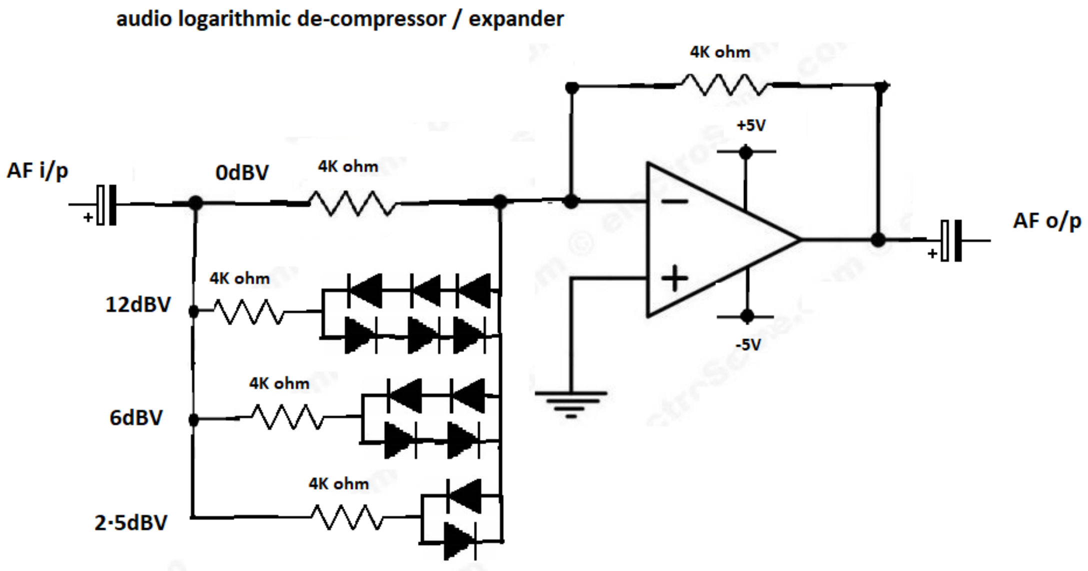


The addition of the "C coupling" capacitors, is to provide a ground voltage reference for the diode feedback circuit to function.

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.



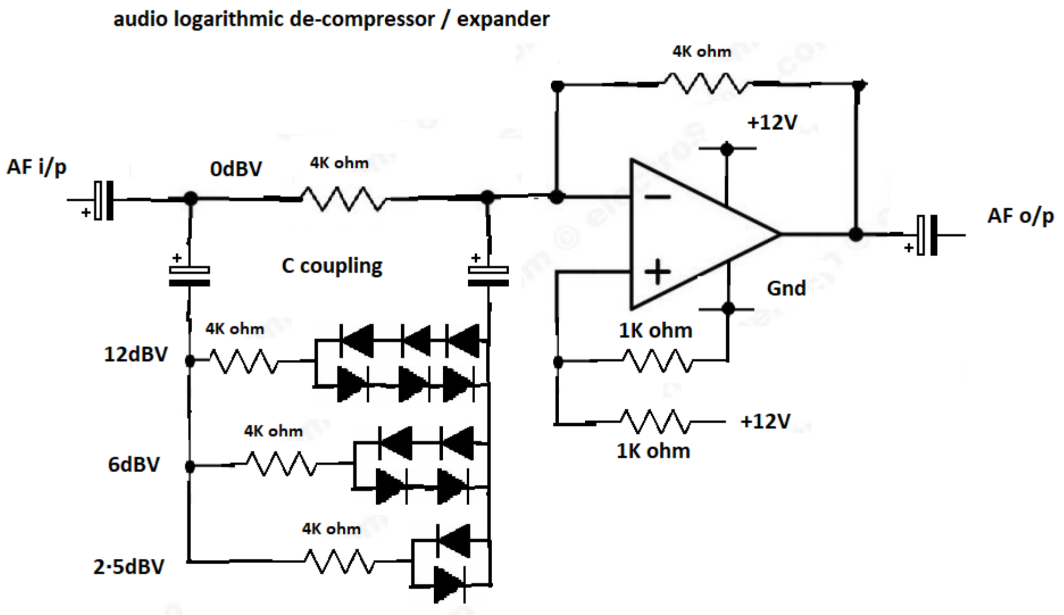
For the logarithmic expander circuit, it is perhaps best to use Shockley diodes due to their low forward voltage, otherwise referred to as the "barrier potential" voltage.

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 forward barrier potential of in essence 0·2 Volts, the combined three diode 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.

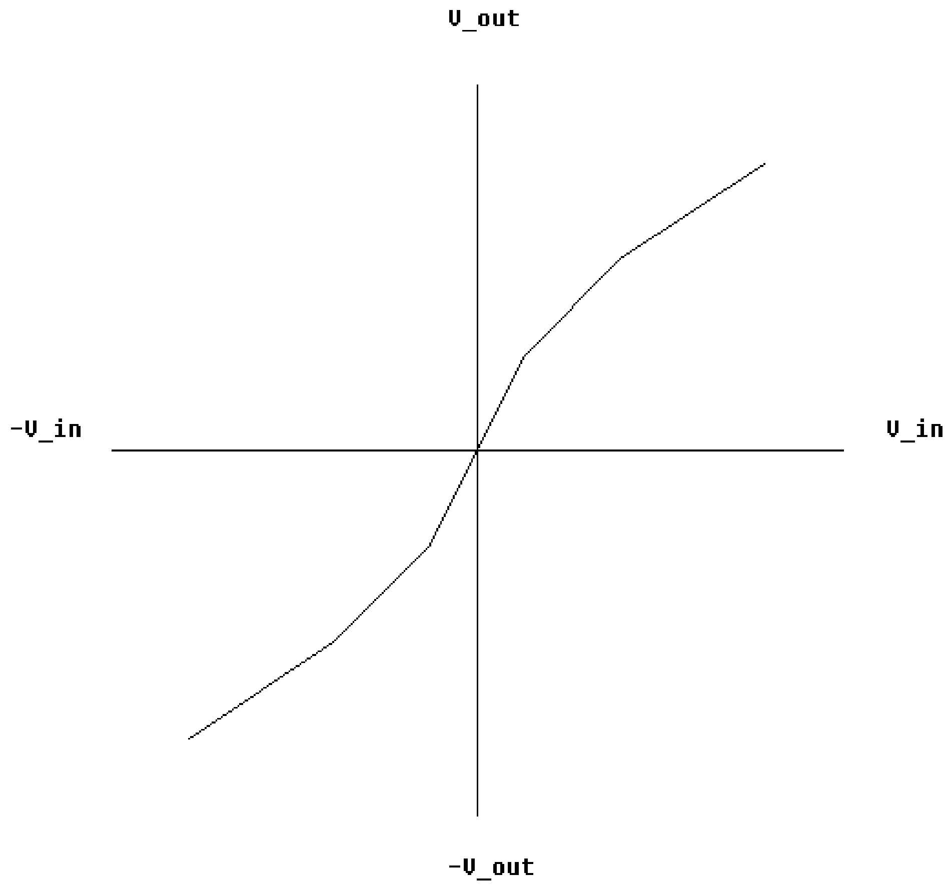
Below, illustrates a single supply voltage circuit version of the logarithmic expander.



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. Here too by replacing the 4Kohm feedback resistor with a variable adjustment resistance, the lesser the value, the less the expansion and hence regained signal from the “S curve” compression.

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

However, the transfer function for the three diode network 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
```