October/7th/2021

Audio dynamic range compressor

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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.

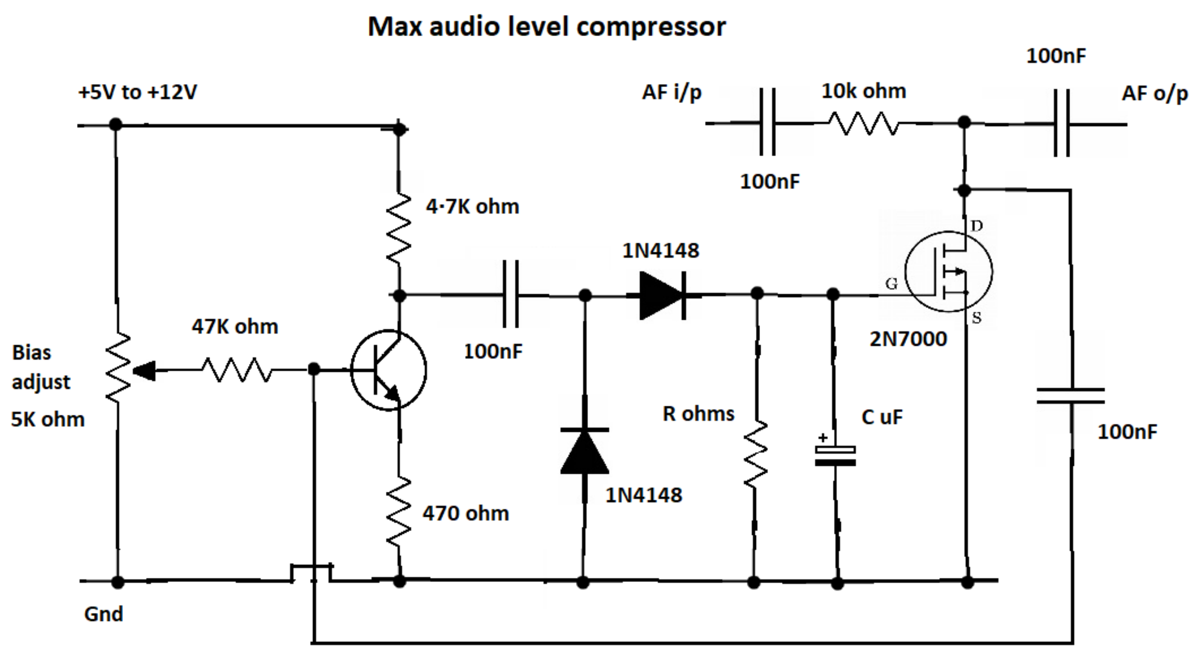
A use of a noise gate is general adopted, but this can lead to a clipped audio as the gate opens up late, in accordance to the spoken voice timing.

Solution.

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.

A circuit that reduce the voice audio would be required, “diagram 1”, the “max audio level compressor”, illustrates this point.

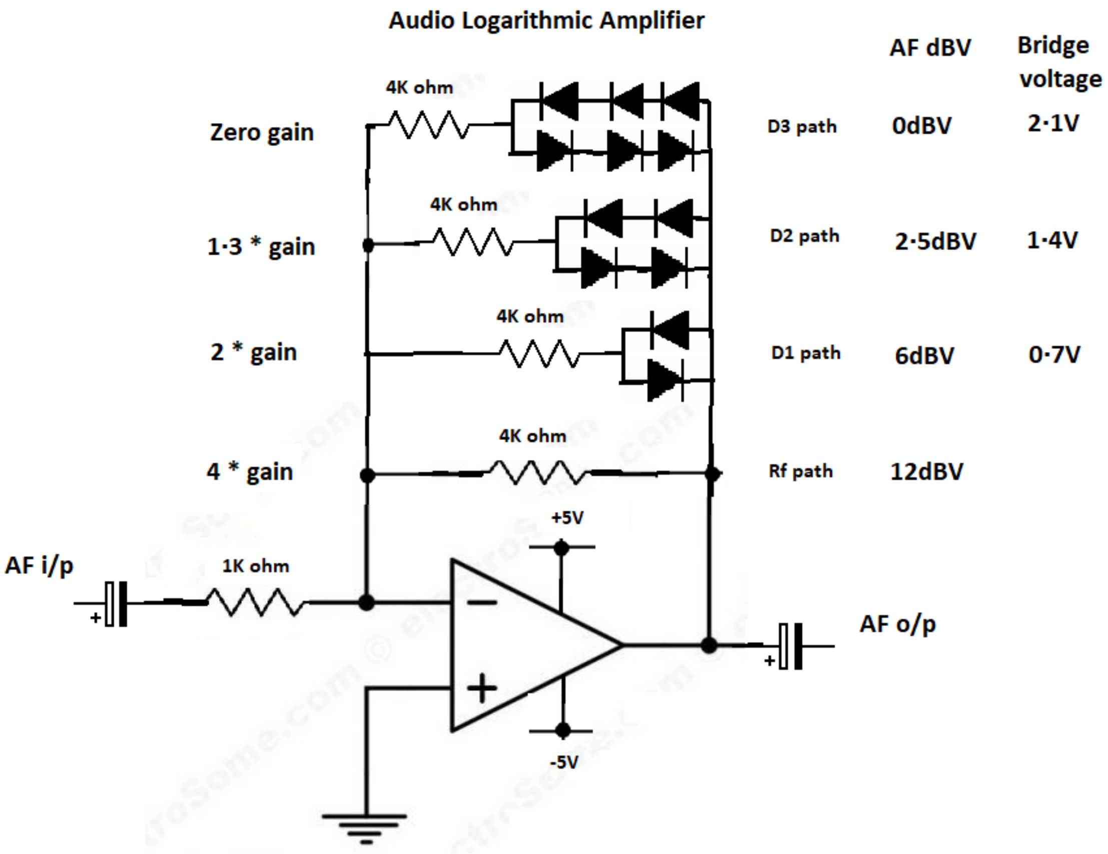
With this max audio level compressor, once the spoken voice audio peek above a minimum limit, the automatic gain control adjustment will reduce the audio amplitude, by virtue of the 10k ohm resitsor and the mosfet 2N7000, half “T section” attenuator circuit. The AGC time constant would as be prefered. The agc circuit also used within the bitx40 radio.



Notice however, that the spoken voice audio content below the limit threshold, is not afffected by this circuit, as this part of the low level voice audio, it base level needs to be boosted up to bring up the low level audio signal.

The accomplishment on the combination of both circuits, is the top level compression, and the low level boost, thereby compressing overall the original dynamic range of the spoken audio from the microphone pickup.

The diagram below, "diagram 2", 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 "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.

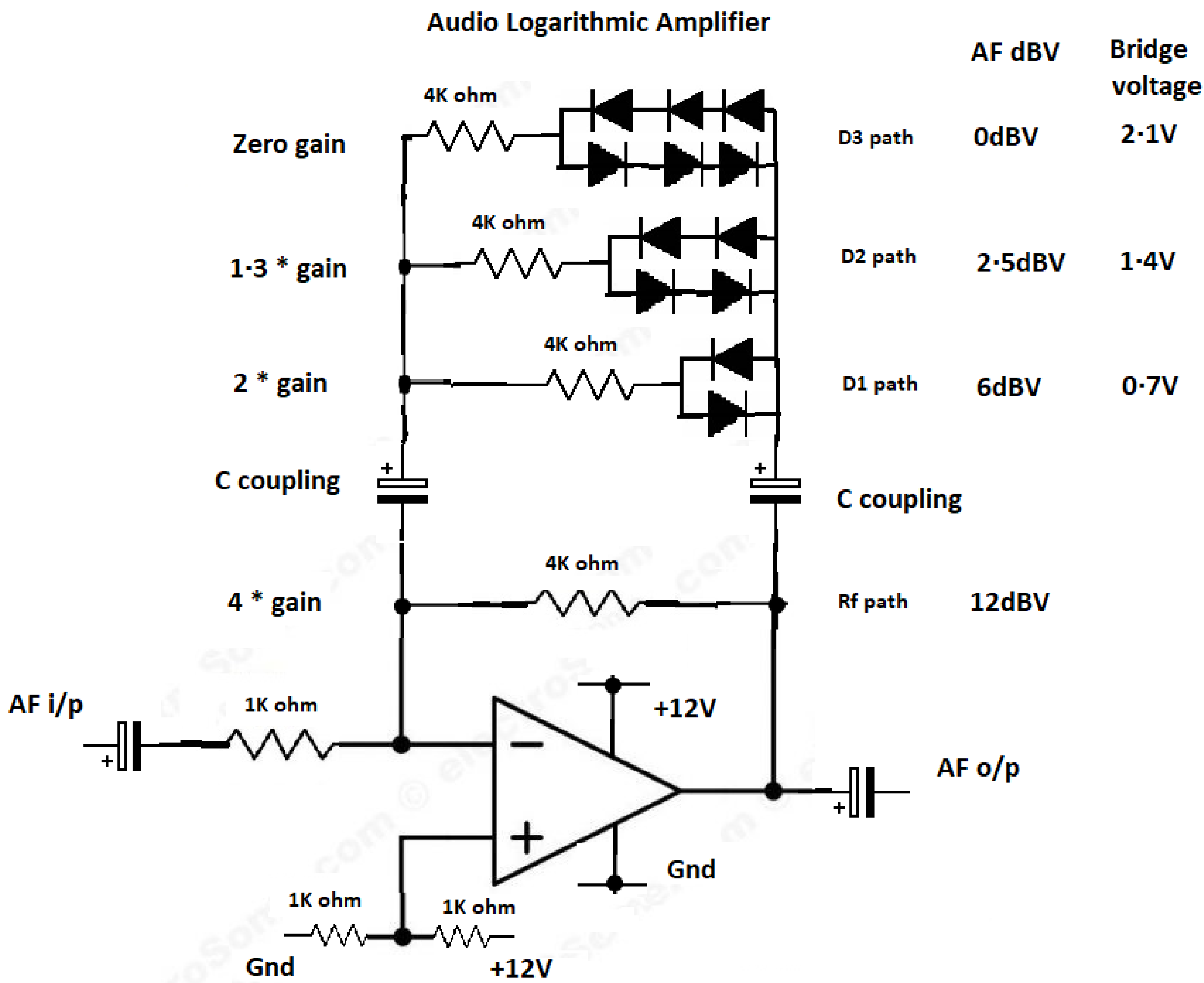
For completeness of the combined audio compression circuit, and an additional max audio level compressor could be added after the audio logarithmic amplifier, to in-sure that the input audio into the radio does not overdrive the radio transmitter.

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, “diagram 3”.

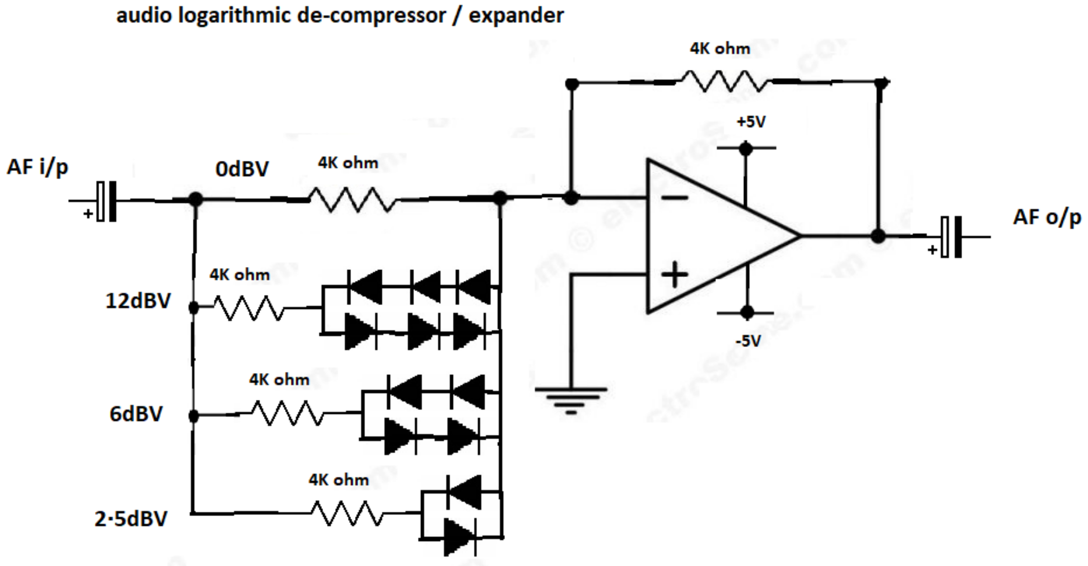


The addition of the “C coupling” capacitors, is to provide a ground voltage reference for the diode feedback circuit to function.

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, “diagram 3”, shows the de-compressor or expander circuit implementation.



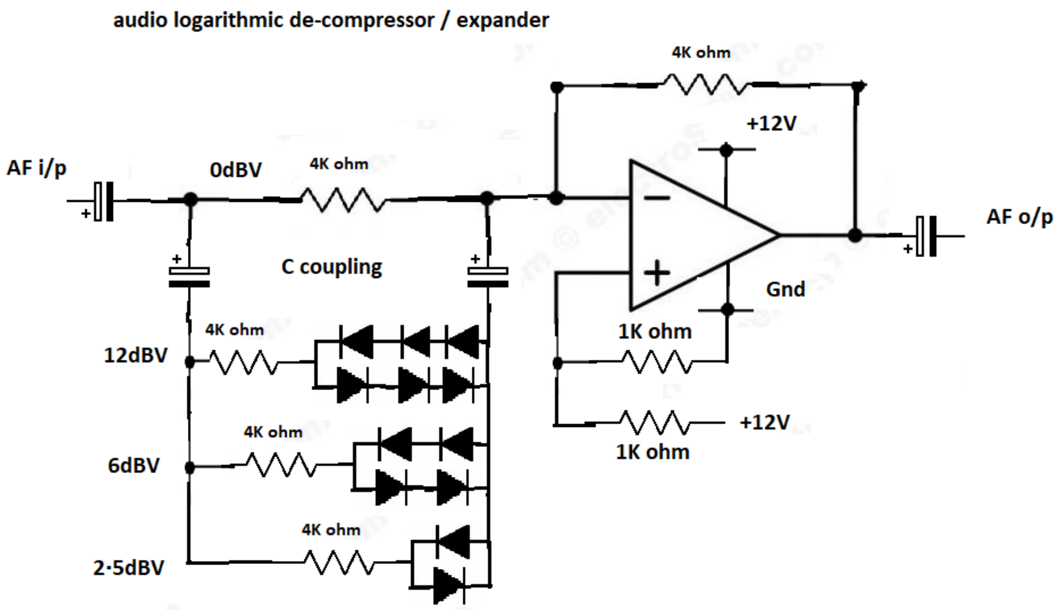
For the logarithmic expander circuit, it is perhaps best to use Shockley diodes due to their low forward voltage, otherwise referred to as the “barrier potential” voltage.

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.

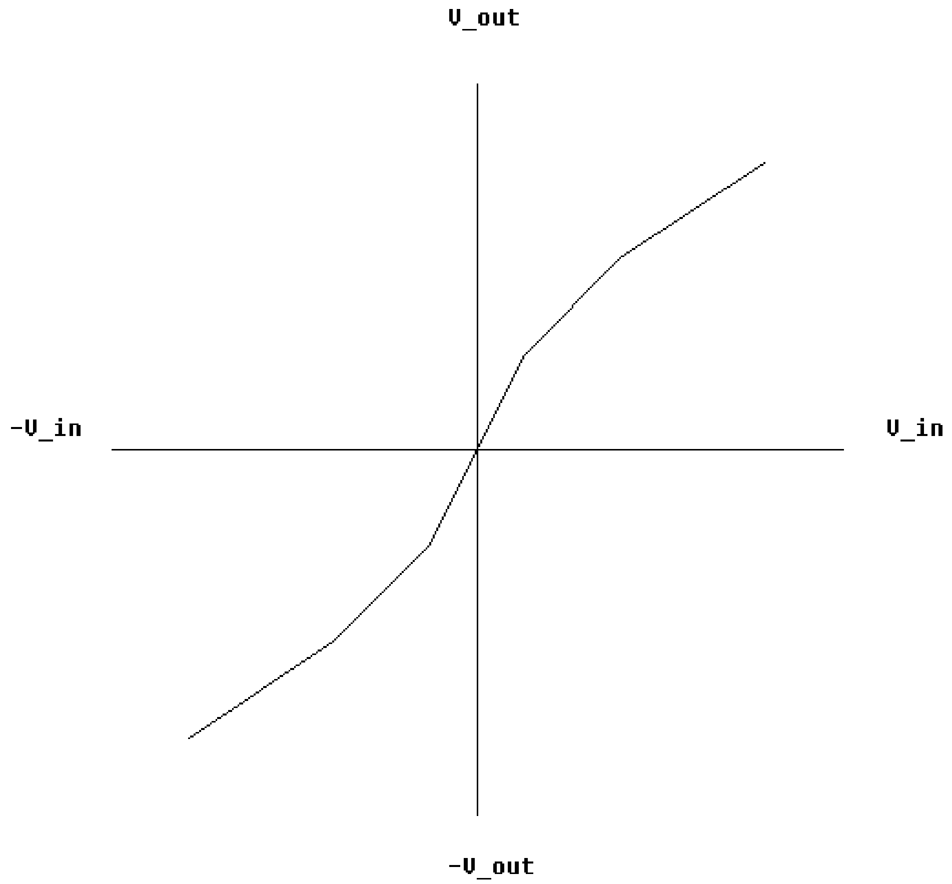
Below, “diagram 4”, illustrates a single supply voltage circuit version of the logarithmic expander.



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”.

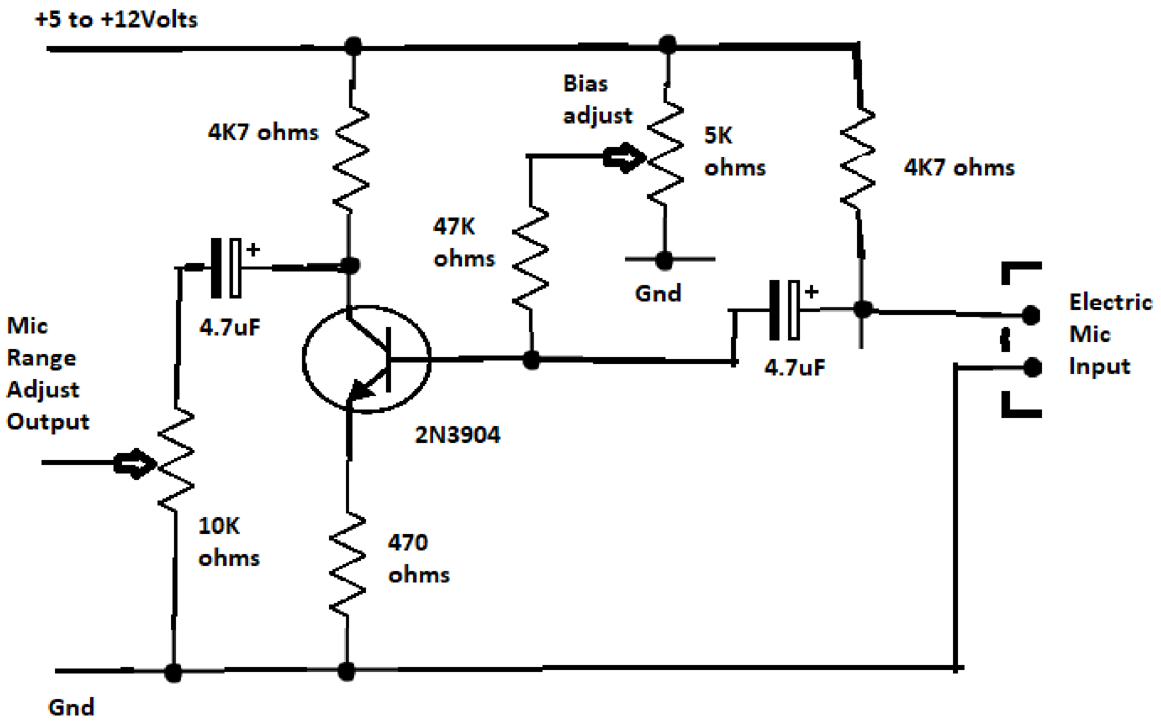
<u>Input and Output Circuits.</u>

The first thing that was required is a microphone pre-amplifier stage, an amplifier with an adjustment mic range setting, below “diagram 6”.

You may have noticed that a so called “AC gain” is not used, but just the “DC bias” current gain settings as the microphone signal amplification.

The reason for this is a discovery that I then stubbled across while determining why an off the shelve small signal video a sound amplifier module was distorting the sound signal.

Microphone Pre-amplifier



I discovered while searching with an oscilloscope, that while the DC bias was fine, as soon as a sound signal was sent through the sound amplifier circuit, the signal itself was in effectively altering the DC bias settings.

Now one would think this is, and that the AC sound signal winding its way through the amplifier to be amplified, would in doing so alter the DC bias. However, the effect the sound AC signal was having upon the transistor DC bias, was to offset the DC bias, and as a result, causing the offset DC bias to distort the AC sound signal output itself.

The search then was on to why the DC bias was distorting the sound signal output. At a lower signal audio frequency, the distortion was little, but at the top end of the HiFi sound signal, the output signal distortion was quite apparent.

I then lowered the capacitor shunt across the emitter resistor, and the problem resolved itself. Then the penny dropped.

The emitter circuit DC bias setting across the emitter resistor, was in fact shunted across the emitter resistor due to the capacitive reactance of the emitter capacitor due to the sound signal audible frequency. The higher the sound signal frequency, the worst the problem became.

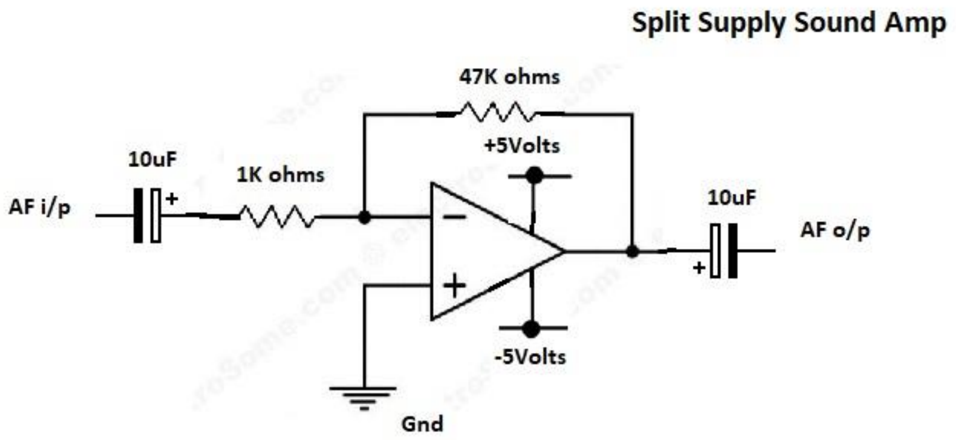
By changing the emitter capacitor, the problem varied, but the signal gain due to the capacitor reactance kept on varying across the audible sound range.

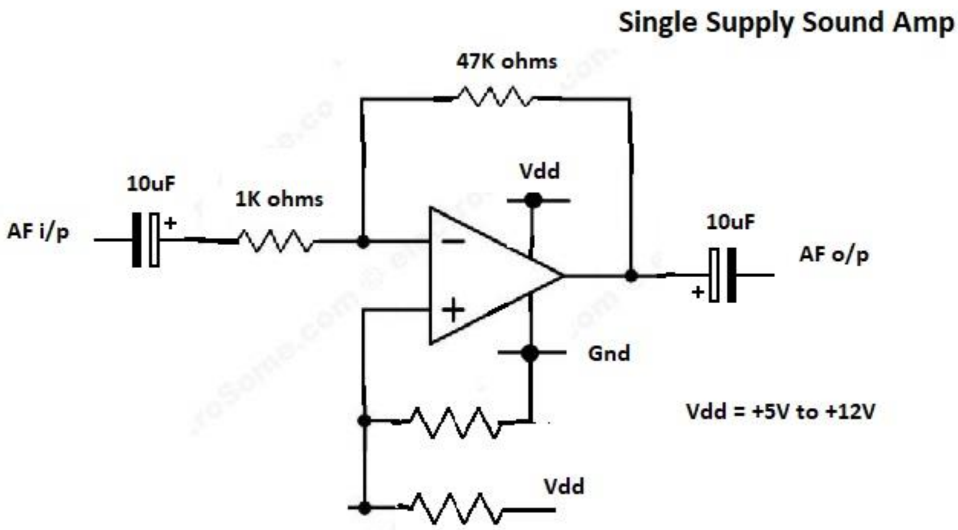
The solution was to only use the DC bias setting as the amplifier circuit signal gain, and in the process avoid any further problems.

The DC bias is set to a factor of ten as the signal amplification, hence the sound signal amplification is ten times, or a factor of ten.

The next circuit amplifier is the microphone main amplifier circuit, shown below as “diagram 7”.

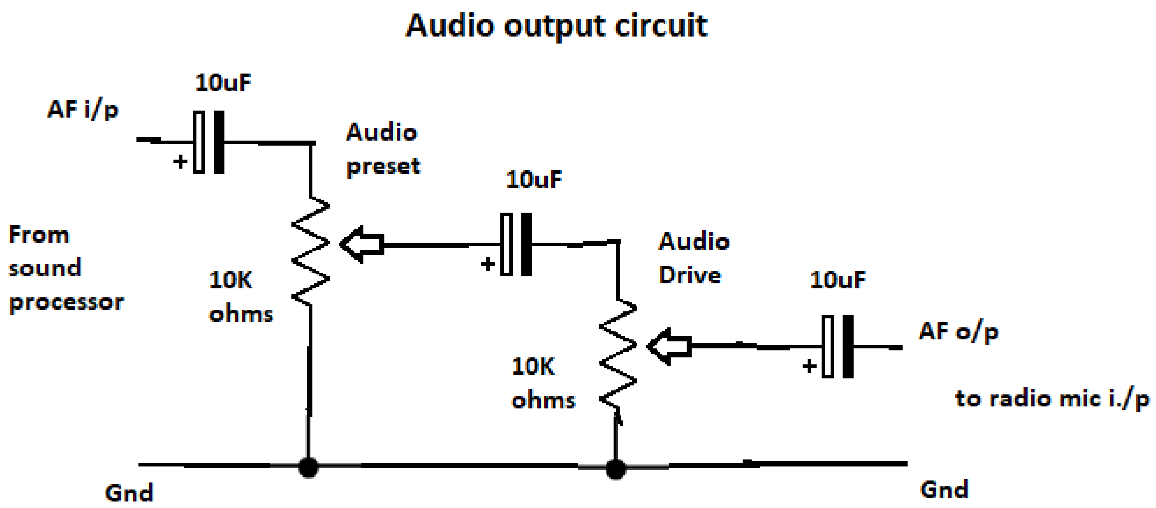
Main Signal Mic Amplifier





The main microphone amplifier has a signal gain of 47 times. With the combination of the microphone pre-amplifier, the overall microphone sound amplification is alterable from 47 times to 470 times.

The audio output from the sound dynamic range compressor is shown below, “diagram 8”.



The audio pre-set control is placed upon the circuit board, but the audio drive is a front panel mounted control, allowing the overall transmitter signal wattage to be altered while using SSB mode. With SSB, the lower the audio signal, the lower the output signal power overall.

The split supply voltage circuits use naturally a positive and a negative voltage supply. The reason for this departure, is to allow the +5Volt supply emanating from the microphone connection output.

However, a negative supply voltage is then required, again 5Volts but in this case a -5Volts supply.

The main reason for this is allow enough voltage supply headroom for the audio signal voltage not to be clipped by the circuits. A split supply 5Volt system would allow an overall 10Volts of audio signal, not including the op-amp head room limits.

After building the circuit, including in addition a supply line lowpass filter on both voltage supply lines, I still got a good 8Volts audio signal range.

The question of generating a negative supply rail, came from the ICL7660 chip, a negative voltage generator. It turns out, that for each op-amp, an individual negative supply voltage generator is required.

A more powerful negative voltage supply line generator may overcome this problem.

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
```