EW - 2 Project 1

Aim -

The aim of the project is to build an audio amplifier with the following specifications -

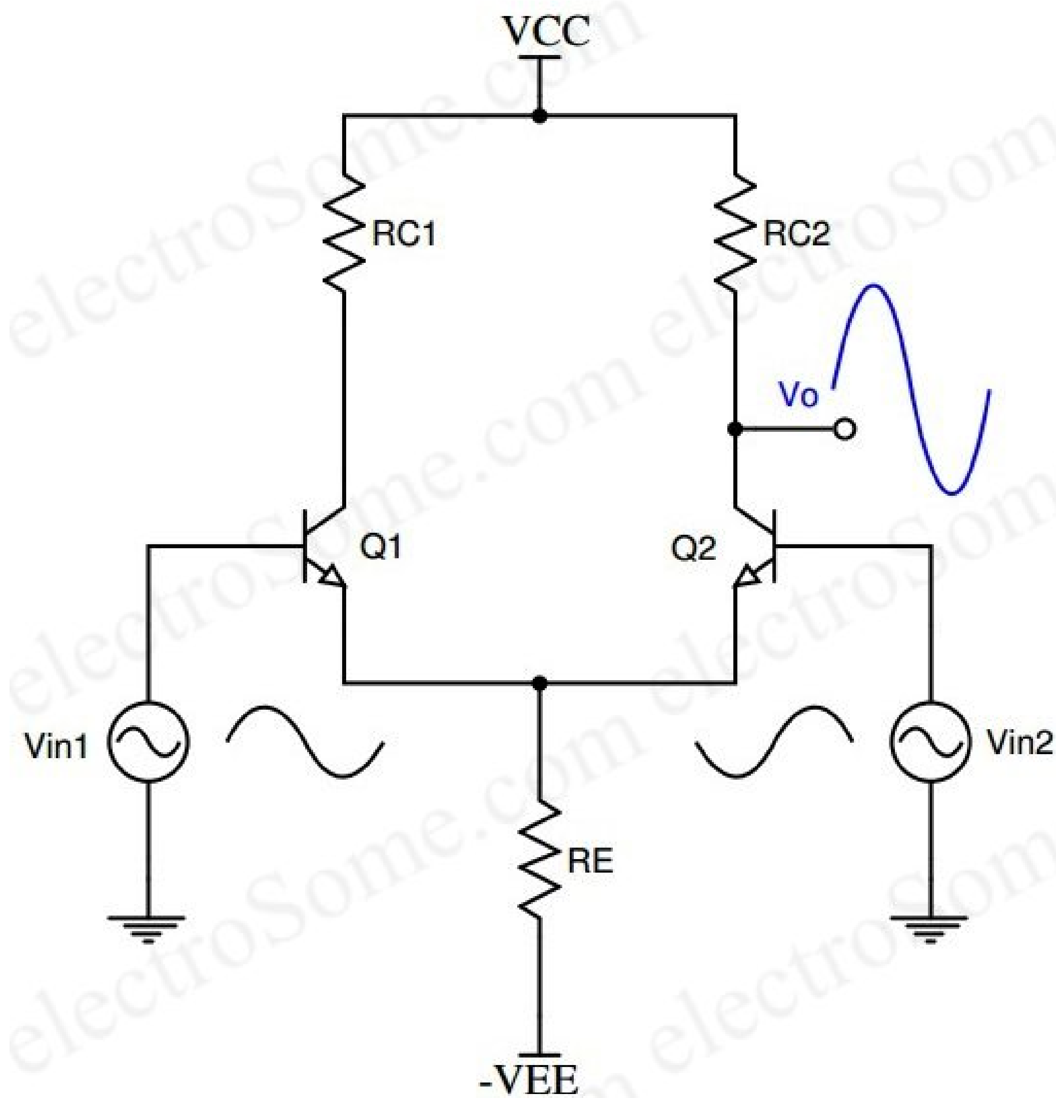
1. Supply Voltage = 0 to 5 V
2. Input small signal voltage = 10-40 mV peak-to-peak
3. Gain = $G_1 \times G_2 \geq$ 500 (Pre-amp and Gain stage)
4. Frequency = Audible Range (20 Hz to 20 kHz)
5. Power $\geq$ 1.5 W
6. Filter **should not** attenuate the input signal
7. Power Amplifier **should not** provide voltage gain
8. Load = 10 Ω

Stages

1. Pre-amp

- The pre-amp stage is required for initial amplification.
- Ideally, the input resistance should not be low as this will cause the amplifier to draw high current, which the microphone cannot supply, leading to ineffective operation of the amplifier.
- For this reason, the common-emitter differential amplifier can be used as it has a high input and output impedance, with good noise performance.
- If the noise performance of a pre-amp is bad, the already weak signal ($10mV - 40mV)$ could be completely overpowered by noise
- In summation, the pre-amp's main purpose is to provide initial amplification to the signal to send it to the gain stage, while also preventing noise from

entering the system.



- For proper operation, the two NPN BJTs must be in active mode ($i.e.$ Base-Emitter junction in forward bias, Base-Collector junction in Reverse Bias).
- The input can be applied to either transistor's base with the choice of either grounding the other transistor's base or applying an equal and opposite input to the other transistor's base (effectively twice amplification).
- In this case, a resistor (R_e) is used instead of an independent current source and $I_{e_1} + I_{e_2}$ is the current that flows through it.

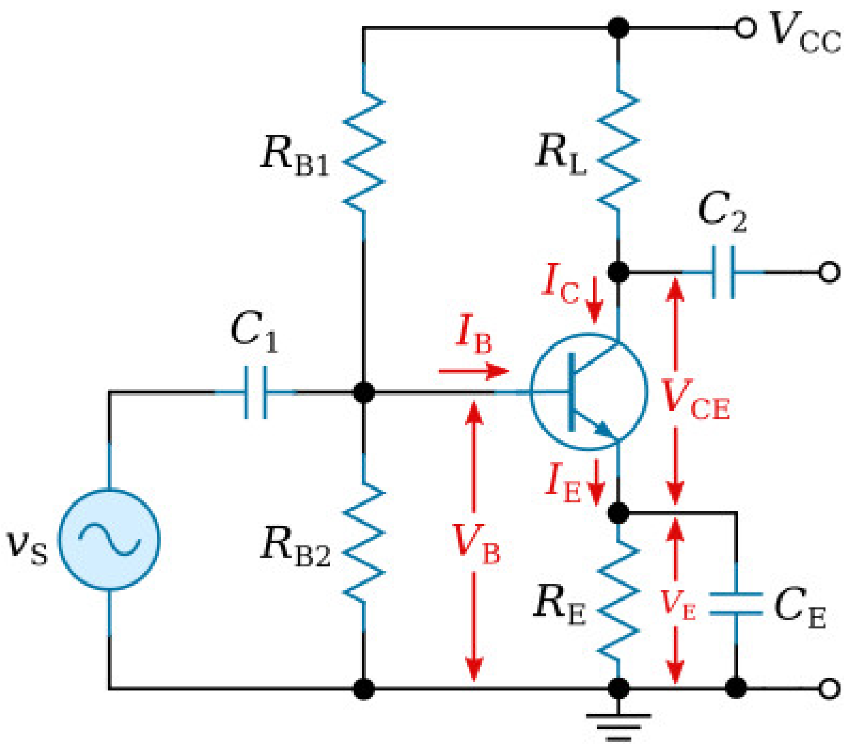
- The bias current values were obtained by the above method, and these were used in the small signal analysis of the circuit.
- To find input resistance, the output must be grounded and a test voltage should be given at the output. The ratio of the input voltage to input current (test) is the input resistance.
- To find output resistance, the differential inputs are grounded and a test voltage is to be given at the output.
- The CMRR is calculated by supplying a common mode signal to the differential amplifier and measuring common mode gain (A_c) and by supplying another differential mode signal to the diff amp and measuring the differential gain (A_d).

  Then, the CMRR is found using:

$$CMRR = 20\log\left(\frac{A_d}{A_c}\right)$$

2. Gain

- For this stage, a CE amplifier can be used as it has low input impedance and high output impedance. It also has high current and voltage gain.



- The input capacitor C_1 serves to block the DC components of the input signal V_s and contributes a pole to the system.

  In this case, the frequency lies between $20Hz$ and $20kHz$, so C_1 needs to be set such that it allows everything below $20kHz$ to pass.

  Additionally, if C_1 is too small, the low frequency components of the input will be lost and the it'll allow more current to pass, causing a larger I_B, which will decrease the current gain.
- C_E serves to block DC current flowing to ground.
- C_2 provides the second pole to the system and provides AC coupling to the filtering stage.
- R_{B1} and R_{B2} are bias resistors, which form a voltage divider with respect to V_{CC}. This is used to be bias the transistor (Base-emitter junction in forward bias and Base-Collector junction in negative bias).
- The stability factor can be found using $S = \frac{\delta I_c}{\delta I_{CBO}}|_{v_{BE},\ \beta}$

Specifically -

$$I_C = \beta I_B + (\beta + 1) I_{CBO}$$

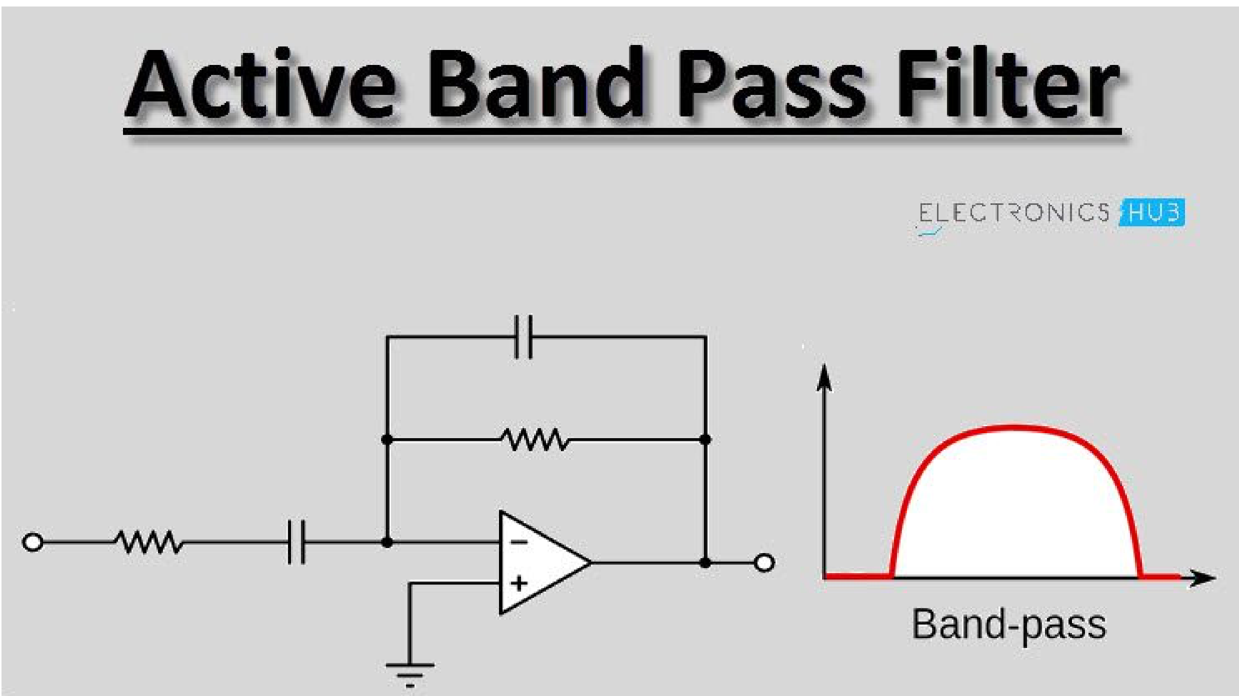
$$\therefore \frac{\delta I_C}{\delta I_C} = \beta \frac{\delta I_B}{\delta I_C} + \frac{(\beta+1)}{S}$$

$$\therefore\ S = \frac{\beta+1}{1-\beta \frac{\delta I_B}{\delta I_C}}$$

3. Filter Stage

The simulated values were used and the result was found to be very close to 1

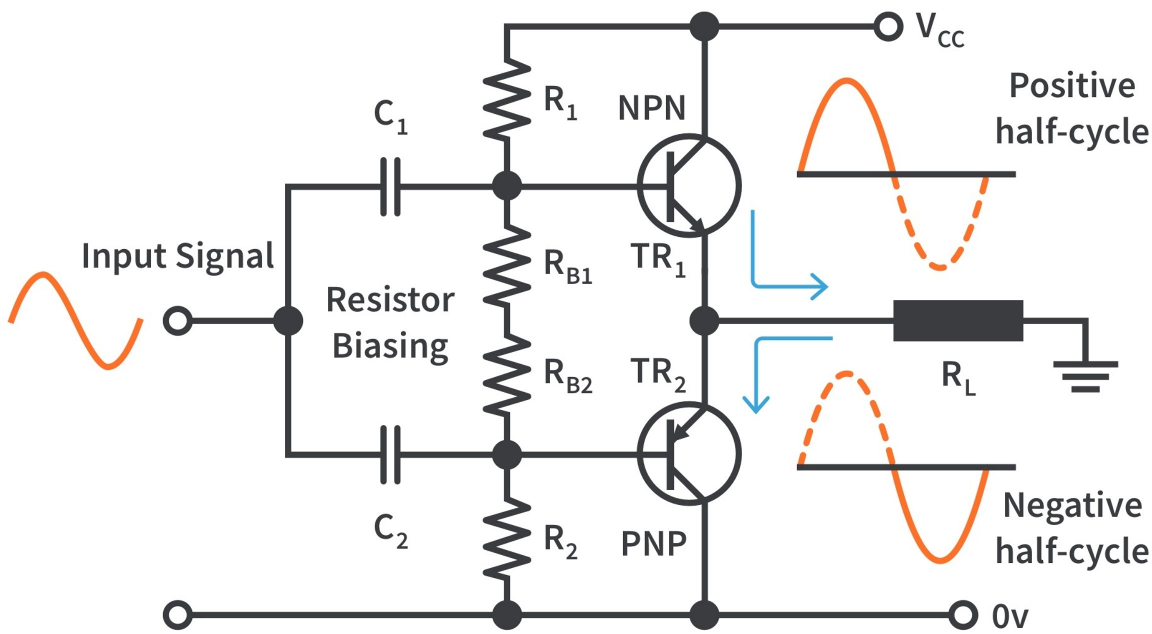
- For the filter stage, a simple active low pass filter can be used.
- **It is recommended to use an active band-pass filter.**



4. Power amplifier

- You are free to use an class of amplifiers, but a class AB amplifier is recommended.
- A class AB power amplifier is an improvement on class A and class B amplifiers for the following reasons -

- class A amplifiers are inefficient, wasting both power and operational cost. For higher output amplification, a higher DC supply is required, which leads to more power dissipation.
- class B amplifiers have high cross distortion due to their configuration. This could be disastrous for an audio signal.
- class AB amplifiers lack the above faults and are more resistant to cross distortion. It also has linearity, which class B amplifiers lack.



- The function of $R_1,\ R_{B1},\ R_{B2}, and\ R_2$ is to bias the bases of the transistors to put them in the appropriate regions of operation in the positive and negative cycles of the input.
- For the NPN transistor to conduct, the Base-Emitter junction must be in forward bias and the Base-Collector junction must be in reverse bias
- The opposite is true for PNP transistors.
- R_1 and R_2 must be equal, and the separation of R_{B1} and R_{B2} has been done to show symmetry in the configuration.
- C_1 and C_2 are used to minimise the base current for both the transistors, which will increase the current gain, further increasing the power gain.

- Additionally, no gain was desired, which was achieved by setting the bias resistances aptly.

Distortion Analysis

Distortion Analysis is done to figure out the amount of unwanted harmonic content in the frequency content of a signal. It can further elaborate on the behaviour of the circuit at different frequencies.

The main formula used was -

$$THD = \frac{\sqrt{\sum_{n=2}^{\infty} V_{n\_RMS}^2}}{V_{f\_RMS}}$$

- THD is the total harmonic distortion present in the signal
- V_{n_RMS} is the RMS voltage of the nth harmonic
- V_{f_RMS} is the RMS voltage of the fundamental frequency

The voltages can be found using the FFT of a sine signal in the *wavegen*, which can be plugged into the formula accordingly.

Having a large THD leads to distortion in the output amplified signal, which can be manifested in the form of skewed or flat regions in the signal. This is disastrous for an audio signal as any skewing or slewing in the input signal can lead to complete destruction of the quality of the audio of the output.

A lower THD also means higher efficiency magnification of the signal, and a higher quality output.

THD can be improved by trying to minimise the circuit interference and by adding filters to shut down higher unwanted harmonic currents in the system. Resistive-capacitive filters (low pass in our case) effectively kill the higher order harmonics, improving the THD of the circuit.

Slew Rate

Slew Rate is defined as the maximum change in output voltage divided by the change in time.

Ideally, the slew rate for a circuit should be infinitely high, so that even for high frequency applications, the circuit does not produce a distorted output. If the slew rate is not high enough, it will not be able to catch up with the input (say high frequency) which will cause distortion and the output will not be as required.

$$S.R. = max(\frac{\delta V_{out}}{\delta t})$$

Slew Rate can be increased by increasing the maximum operating voltage or by making the capacitive impedances of the subcircuits smaller. This will allow that particular subcircuit to operate at a higher frequency.

$$S = 2\pi f_m V_m,\ V_m = 15mV,\ f \text{ varies according to input audio}$$