

An AM Radio Realized with Only Discrete Bipolar Transistors

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1 Introduction:

In this project, we developed an all-analog AM radio receiver, which operates at an average power dissipation of about 70mW. Some of the primary features in our design include automatic gain control, a pre-amplifier stage for improved reception and pseudo-stereo for more pleasant audio output. The radio is realized on a prototype board.

The term modulation refers to a process which causes a shift in frequency contents of a signal. The idea of using modulation is two-fold: first, transmitting audio signal directly would require prohibitively large size and power requirements at the antenna. Second, modulation can be put to use to take advantage of full bandwidth of the channel (e.g. in frequency division multiplexing). There are several types of modulation; however our interest is only in amplitude modulation.

In amplitude modulation, the shift in frequency spectrum is achieved as a result of multiplying the baseband signal (audio in our case) with another signal (called a carrier signal). Let the carrier be a cosine of frequency ω_c , amplitude A and phase θ_c . Then in Fourier domain:

$$m(t) \Leftrightarrow M(\omega) \quad (1)$$

$$A \cos(\omega_c + \theta_c) \Leftrightarrow A/2 [\delta(\omega + \omega_c) + \delta(\omega - \omega_c)] \quad (2)$$

$$m(t)A \cos(\omega_c + \theta_c) \Leftrightarrow A/2 [M(\omega + \omega_c) + M(\omega - \omega_c)] \quad (3)$$

Thus the baseband signal gets translated to ω_c and hence can be transmitted easily through an antenna. The recovery of this signal is called demodulation or detection. At the receiver end, in order to recover the original signal from the modulated signal the frequency spectrum has to be shifted back to its original position. Thus, it has to again be multiplied by a sinusoid. In frequency domain:

$$[m(t)A \cos(\omega_c + \theta_c)]/A \cos(\omega_c + \theta_c) \Leftrightarrow M(\omega) + 1/2 [M(\omega + 2\omega_c) + M(\omega - 2\omega_c)] \quad (4)$$

The components at $2\omega_c$ can be eliminated by low-pass filtering the output. Hence demodulation is almost identical to modulation.

The block diagram of our radio is shown in Figure 1. The design is based on the superhetrodyne principle, according to which the receiver has a local oscillator whose frequency can be changed to pick up different radio stations. In traditional AM design, the local oscillator and received signal are mixed such that the difference between oscillator and signal frequencies is always 455 kHz. Therefore if the received signal is in range of 500 kHz to 1700 kHz, the bandwidth of the local oscillator must be from 995

kHz to 2105 kHz. The FCC has assigned a separation of 10 kHz between adjacent radio stations.

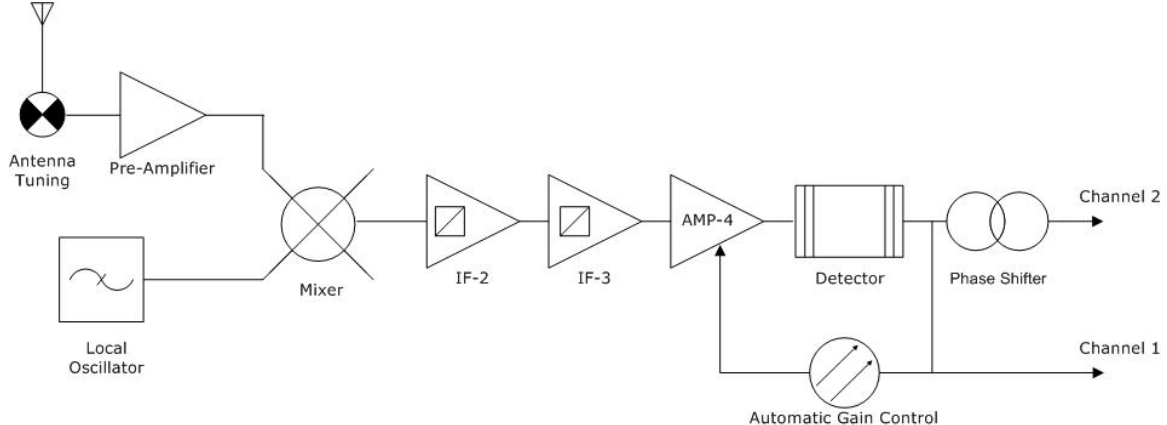


Figure 1: Block diagram of the implemented AM radio.

First, the RF modulated signal is captured by an antenna. This is then passed through a pre-amplifier stage and multiplied by a sinusoid generated by a local oscillator. The mixer performs this multiplication. After this, the output of mixer has to be passed through a series of frequency selective amplifiers (called I.F. amplifiers) to eliminate harmonics and noise. The I.F. amplifiers are tuned for frequencies near 455 kHz. A fourth amplifier stage is used for achieving further voltage gain. The original signal can then be recovered by detecting the envelope of the signal at the output of the fourth amplifier stage. This is done via the detector block. A very desirable feature in radios is their ability to automatically adjust so as to keep overall sound intensity constant. This property is referred to as automatic gain control (AGC). A radio with AGC capability is thus able to increase or decrease the gain of various amplifier stages in the receiver so as to keep a constant output even with fluctuations at the input. Another feature which we introduced in the design was to create a second audio channel that is a phase shifted version of the first channel. The resulting sound is more pleasing than a mono speaker output. We call this attribute pseudo-stereo output.

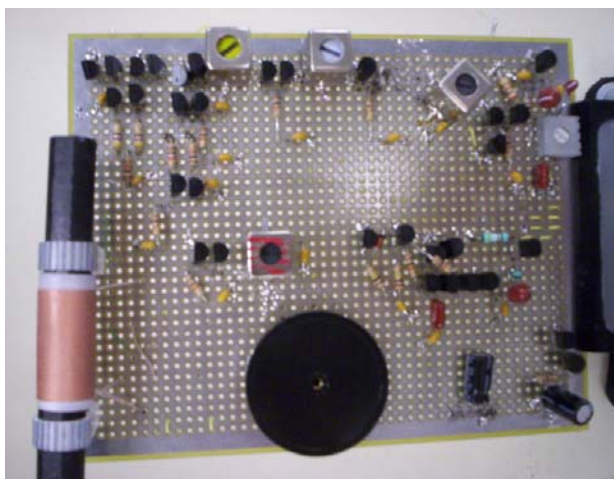


Figure 2: A photograph of the finished AM radio. The only active components used were BJTs.

The radio operates from a 10V power supply (although it can work equally well from a 9V battery). The minimum operating voltage was about 4V. The total current consumption at 10V is about 70mA. A total of 27 bipolar transistors were used in the design. The average amplitude (pk-pk) of the audio output signal for a normal station was around 400mV, so it can be directly connected to powered speakers. Every DC voltage was bypassed to ground with a 0.1 μ F ceramic capacitor. We used only one biasing voltage, called V_{bias}. The

Vbias was generated through two diode drops and for all experiments, was measured to be 1.30V. The finished radio is shown in Figure 2.

2 Antenna Tuning and Pre-amplifier:

In most wireless receivers, the first block immediately after the antenna is some pre-amplifier, usually a low-noise amplifier (LNA). Its noise figure sets the lower bound on the noise performance of the entire receiver.

The pre-amplifier that we designed uses a standard differential pair, as shown in Figure 3. Instead, using a low-noise amplifier could have improved noise performance. However, most LNA designs use inductors, which introduce resonance. Therefore, an inductor with a high bandwidth could have been a better alternative. The outputs from the antenna (*Ant_pos* and *Ant_neg*) tuning are presented differentially to the mixer, and that improves the reception.

From hand calculations, the transistors are biased at a collector current of about 0.4mA. The inputs to the pair are biased around Vbias. The small signal voltage gain of the differential pair is roughly -16.2 . The beta of the transistors used here is 290, and from calculations, the input impedance seen by the antenna is in the order of $36k\Omega$. The output impedance is r_0 in parallel with 1K resistor. The f_t of the transistors used is about 300MHz, while AM frequency range is about 500-1700 kHz. For all practical purposes, we do not consider C_π and C_μ in our small signal hand calculations.

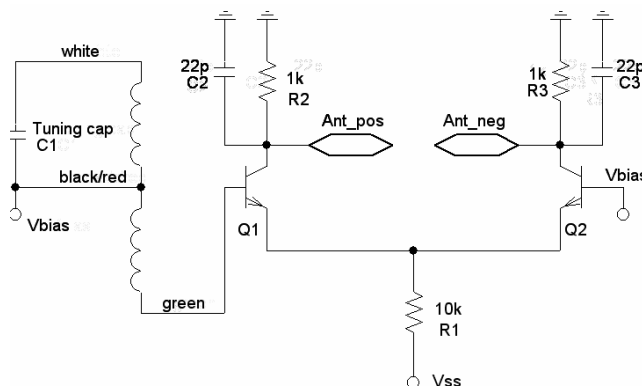


Figure 3: The schematic of the antenna tuning circuit and the preamplifier

of about 7.2MHz, and this helps reduce the noise floor. Using inductors at the output instead of the 1K resistors could have further improved the signal to noise ratio.

3 Local Oscillator:

In order to assure that the proper signal is captured and converted into the intermediate frequency of 455 kHz, the local oscillator must be tuned to a frequency that is 455 kHz higher than the desired frequency. For the valid AM range of 500 kHz to 1700 kHz this requires a range of 955 kHz to 2.155 MHz. In addition, the oscillator should be as close to a sine wave as possible; this limits the number of frequencies input into the mixer.

From measurements, the output biasing voltage on the two outputs was observed to be $-374mV$ and $-432mV$ (were estimated to be 0.4V). This could either be due to resistor mismatch, or beta-mismatch between the transistors. The emitters are near at around $-1.8V$.

The resistors introduce noise, but we consider it to be acceptable for the current design. The 22p capacitors in parallel with resistors give a cut-off frequency

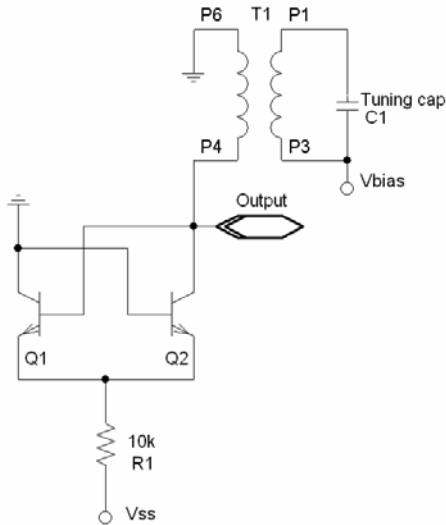


Figure 4: Schematic of the local oscillator.

With a specified turn ratio of 95:9 (the primary is between P1 and P3), the effective inductance and capacitance become 0.32 μH and 15 nF, respectively (the inductance on the secondary coil is found using equation 5). With these values the (minimum) oscillation frequency can be found using equation 6.

$$\sqrt{\frac{L_1}{L_2}} = \frac{N_1}{N_2} \quad (5)$$

$$f_0 = 2\pi \cdot \omega_0 = \frac{1}{\sqrt{LC}} = 720 \text{ kHz} \quad (6)$$

Assuming the ability of the tuning capacitor to decrease to at least 5% of its max value, the theoretical maximum oscillation frequency would be 3.2 MHz. In reality, there will be several capacitances in series with the LC circuit, with C_μ especially prominent. However, the large value of C allowed by the almost 10:1 ratio of the transformer mean that these small values have little effect.

The measured frequency range of the oscillator was 690 kHz to 2.54 MHz. (see Figure 5 for an example of the waveform). These values track fairly well with the predicted value. The lower than expected minimum value could be caused by a tuning capacitor with a higher than expected maximum capacitance or by a capacitance in parallel with the LC circuit, both acting to increase the minimum value (perhaps due to wiring issues). In addition, the lower maximum is probably caused by both a tuning capacitor with a higher

The design chosen for the mixer was very simple (see Figure 4). A negative resistance was formed by the combination of two BJTs. This was then used to complete the loop consisting of an inductor (a transformer was physically used) and a capacitor. The relationship between these two components allowed the oscillation frequency to be set. In addition, the capacitance provided by a tuning capacitor, allowing the frequency to be changed manually to pick up different AM stations. In this circuit, the transformer (T1 in Figure 4) supplied the inductance of 360 μH (measured between P1 and P3). The tuning capacitor had a stated upper value of 141pF. The lower limit was not specified. However, the transformer

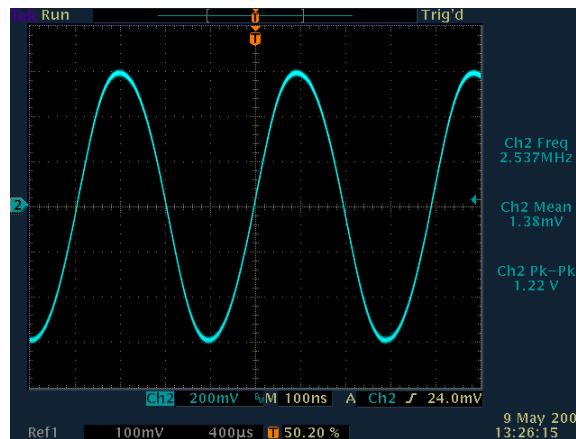


Figure 5: Screen capture from the oscilloscope showing the local oscillator wave form.

minimum size then the 5% predicted and a parallel capacitance. However, the measured range encompasses the desired range for AM reception totally so that no modifications were made. The measured amplitude of the oscillations (at the output) was 1.22 V from peak to peak. This is a relatively large signal and was adequate for the mixer input. The waveform, while not a perfect sine wave, was close, with the majority of its energy at the oscillation frequency rather than any harmonics.

4 Mixer:

Mixers are one of the most important building blocks of modern radio receivers. They find applications in almost all communication systems, like phase locked loops, modulation, frequency division multiplexing and many others. The essential idea behind mixers is multiplication in time domain (which results in translation in frequency domain). A good mixer should introduce a low number of harmonics at the output. There are two broad types of mixers: fully balanced mixers and unbalanced mixers. Fully balanced mixers are four quadrant multipliers while unbalanced mixers can be two-quadrant or one-quadrant multipliers.

The mixer that we designed is an unbalanced mixer. As shown in the schematics (Figure 6a), it consists of two stages of differential pairs. The first stage (Q1 and Q2) controls the tail current of the second stage (Q3 and Q4). If *local_osc* is active, it takes away all of the current from Q4. Likewise if *Ant_pos* is high and *Ant_neg* low, Q4 does not conduct. Q4 only conducts when Q3 is off and Q1 is off. This amounts to a single-quadrant multiplication. The transformer at the output of the mixer serves as the first IF amplifier. Its turns ratio is 87:7 (~12:1), so it produces a current gain at the output.

The DC tail current for the Q1-Q2 pair is about 0.9mA. The first IF amplifier is reasonably frequency selective. In one of the experiments, 180mV (pk-pk) signal was presented as input to the transformer. The output observed was about 34mV (pk-pk).

In an attempt to gain better performance, we designed a fully balanced mixer as well (Figure 6b). The noise level appeared to be lower, however the overall gain reduced. There are several reasons to explain this, the foremost one being that the matching requirements are more rigorous for a fully balanced design. However, we are reasonably satisfied with the performance of our design and do not feel the need to switch to balanced mixer.

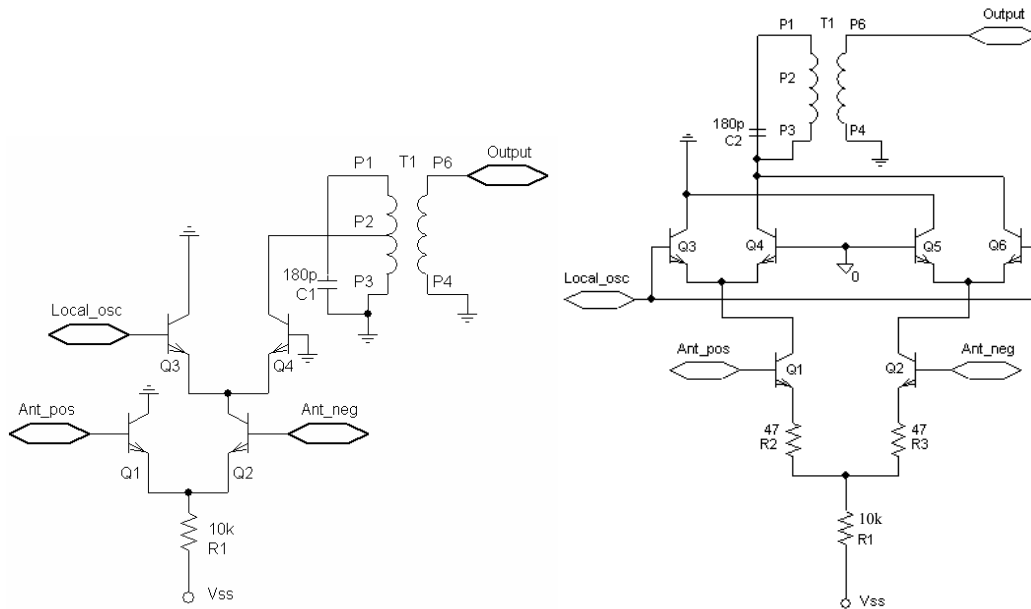


Figure 6: (a, left) The basic mixer currently in use in the completed radio.
 (b, right) The fully balanced mixer.

5 IF amplifiers:

The signal from the mixer contains many frequencies and, unfortunately, most of the energy in the signal is not in the desired intermediate frequency of 455 kHz. The functions of the two IF amplifiers are therefore to both amplify and remove excess frequency information from the signal. The amplifier thus needs to be a band pass filter with a fairly high Q value. A transformer once again helps to solve this problem. A transformer with an internal capacitance was chosen. The part was specifically designed for this application and its inductance and capacitance work to provide a center frequency near 455 kHz. In addition, the turn ratio of the transformer also allows for a large current gain in these stages.

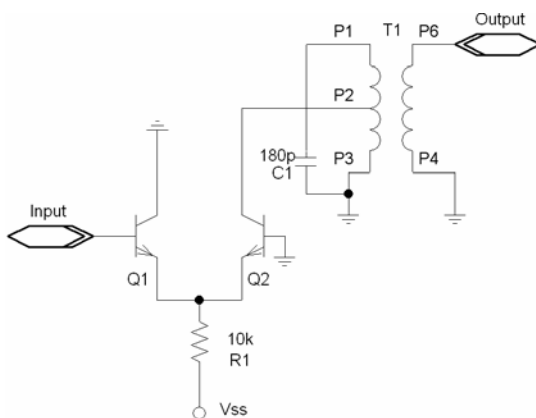


Figure 7: Schematic of the IF amplifier

The amplifier itself was composed of a differential pair of high speed BJTs. The f_t of transistors was around 300 MHz. With our operating frequency of 455 kHz, we make the approximation that we are operating at a much lower frequency and can ignore capacitances. The operating current through the transistors is approximately $9.3V/10k\Omega * \frac{1}{2}$, or .465 mA. This creates an r_π of 15.6 k Ω . The total input resistance is on the order of $2r_\pi$, or 31.2 k Ω (the β for these transistors is ~ 290).

This value is quite high; however, because of the turns ratio of the transformer much power is still transferred to the next

stage. The Q of the transform as given on its data sheet is 80, quite high and able to block out undesirable frequencies.

Measurements of the devices implemented in the radio show fairly high gains. To measure these values a small amplitude sine wave (10 mV) was applied to the input of each amplifier. The frequency of the sine wave was chosen to be the center frequency of the transformer. The gain of the 2nd IF amplifier is -79, as measured before the transformer. When the output is measured after the transformer the gain is -9, as expected with a turn ratio of 57:7 (~8:1). While this voltage gain is fairly low the current gain is quite large, benefiting from the high turn ratio of the amplifier. The two measured 3dB points occurred at 4 kHz on either side of the center frequency. This bandwidth of 8 kHz is approximately ω_0/Q . Using this equation gives a Q of this IF amplifier to be 70, well within the specifications provided by the manufacturer. The 3rd IF amplifier was tuned to a slightly different frequency (~2 kHz off from the center frequency of the last stage).

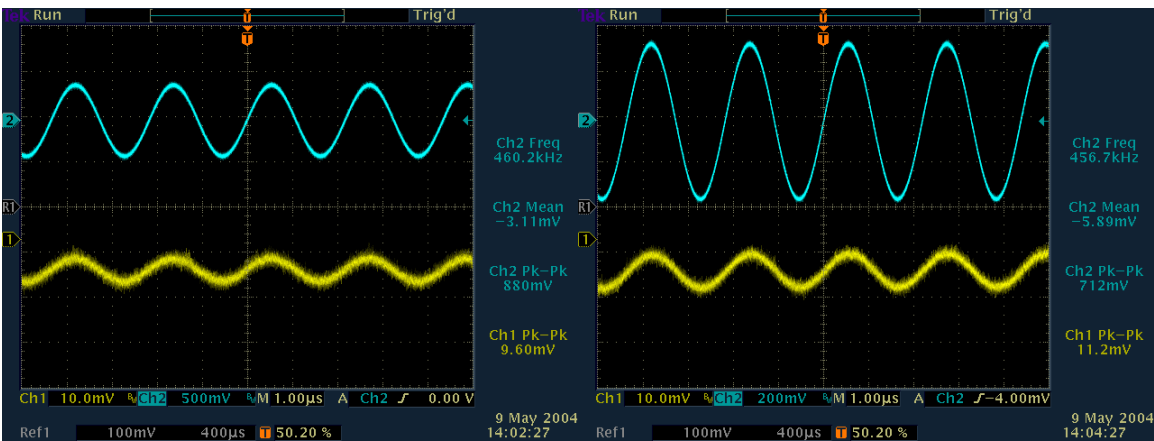


Figure 8: The measured gains of the last two IF amplifiers. Both outputs were captured before the transformer. At left is the 2nd amplifier, the 3rd amplifier is on the right. The yellow wave form is the input, while the blue represents the measured output.

The gain before the transformer was -72 (expected due to the similarities in design). After the transformer the gain was -45. This higher value indicates the difference in the number of turns of this transformer. The turns for this component are 50:27 (~1.8:1).

The screen captures in Figure 8 illustrate both the gain and frequency selectivity of the amplifiers. The input signal, in yellow, was quite noisy. This was a side effect of having been put through a voltage divider in order to reduce the voltage enough so that the amplifier does not saturate. A side effect of this reduction is the introduction of noise into the signal. Although this introduction of noise was unintentional, it serves to illustrate how effective the IF amplifiers are at filtering out the undesired frequencies present in input signals. As can be seen from the plots the output signal is quite clean, with only the desired signal being amplified. This is in contrast to what would be seen with a non-frequency selective amplifier – a signal where the noise was amplified as much as the signal itself.

6 Automatic Gain Control and 4th Stage Amplifier:

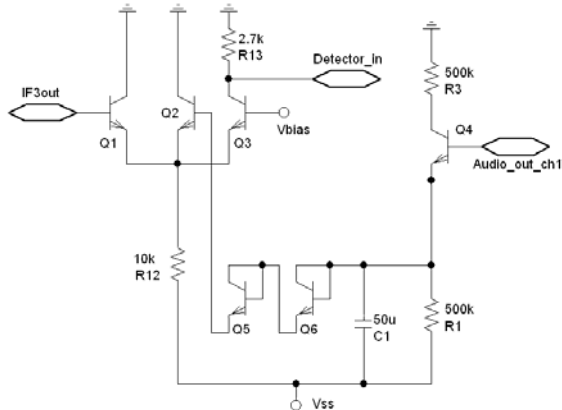


Figure 9: Fourth stage amplifier with Automatic gain control circuitry

This is also understandable because input of the demodulator is a common collector amplifier. For the gain control, the output of one of the channels is presented at the base of the transistor Q4. Transistor Q4, along with R3, R1 and C1 performs rectification and low-pass filtering of the audio output stream. After two diode drops, this rectified-filtered signal goes to the base of Q2. When the audio level is loud at the output, magnitude of voltage at the base of Q2 increases. As a result, the current flowing through Q2 increases and Q2 actually takes away some of the current that should have been flowing through Q3. This reduces the drop across R13 and consequently overall audio level is reduced (see simulation results on Table 1).

The amplifier has a tail current of about 0.8mA. This means that large-signal current flowing through Q3 is about half of the tail current. From simulation and measurement, it was found to be close to 0.39mA. The output of the 4th stage is biased at around -1.06V. The small-signal voltage gain of the amplifier is about -42.66 (assuming no current is flowing through Q2). Figure 1 shows the measured performance of the 4th stage amplifier.

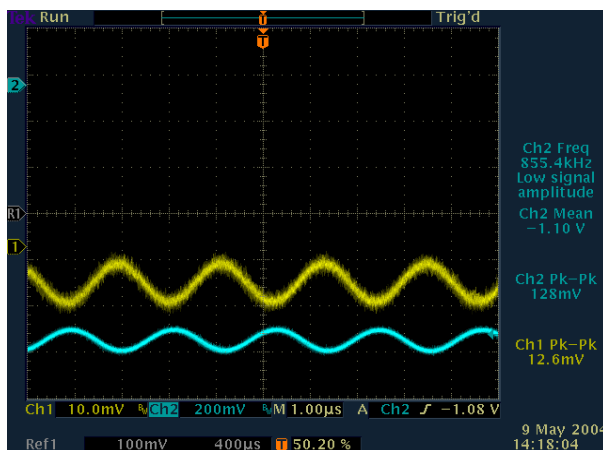


Figure 10: Fourth stage amplifier waveforms. Yellow waveform represents the input; blue waveform is the output of the amplifier. Note the biasing point of the output

Automatic gain control uses feedback to keep the gain of the overall radio to a level that is desirable for listening pleasure. In this way, if there is a very loud channel or a very weak channel, the audio level is kept the more or less constant.

The output of the third IF amplifier is fed as input to the fourth amplifier stage. The fourth amplifier is introduced to achieve a voltage gain, and to perform gain control. The first three IF amplifiers give current gain and not a voltage gain. From our experiments, we concluded that demodulator gives superior performance if a voltage gain

is introduced before it. This is also understandable because input of the demodulator is a common collector amplifier. For the gain control, the output of one of the channels is presented at the base of the transistor Q4. Transistor Q4, along with R3, R1 and C1 performs rectification and low-pass filtering of the audio output stream. After two diode drops, this rectified-filtered signal goes to the base of Q2. When the audio level is loud at the output, magnitude of voltage at the base of Q2 increases. As a result, the current flowing through Q2 increases and Q2 actually takes away some of the current that should have been flowing through Q3. This reduces the drop across R13 and consequently overall audio level is reduced (see simulation results on Table 1).

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Table 1: Results from simulation of gain control		
Audio_out_ch1	$I_c(Q3)$	$I_c(Q2)$
100mV pk-pk	391 μ A	2 μ A
200mV pk-pk	378 μ A	48 μ A
300mV pk-pk	327 μ A	148 μ A
400mV pk-pk	200 μ A	400 μ A
500mV pk-pk	25 μ A	750 μ A

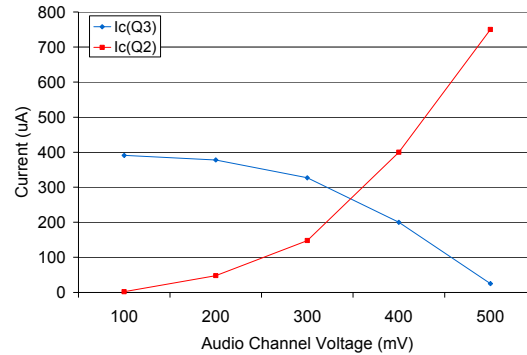


Figure 11: AGC response.

7 Demodulator:

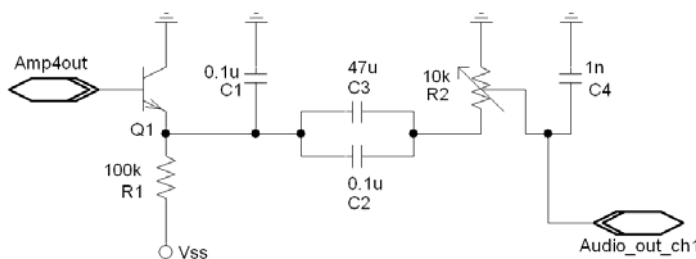


Figure 12: The schematic of the demodulator.

The demodulator functions to recover the audio signal from the AM modulated waveform. By this point in the circuit the radio signal should consist of only the intermediate frequency of 455 kHz modulated with some audio signal. The demodulator must perform

two functions in order to recover this overlying signal. The first task is to rectify the received signal. Once this is done a low pass filter is run over the resulting signal to give the final value. The low pass filter must have its cutoff frequency well clear of the intermediate frequency; however, with this at 455 kHz and the max audible sound at 20 kHz there is quite a lot of room left for the filter.

The circuit used in this implementation is fairly simple. An initial BJT serves to buffer the output and provide a rectified signal. The low pass filter is made up of the small capacitor and the output resistance of the BJT, approximately 1/gm. With a capacitor value of .1 μ F, the resulting -3_{dB} frequency works out to be 31 kHz. This fairly low value ensures that no IF frequency component slips through the filter. A 10 k Ω potentiometer is placed on the output to allow on board volume control of the audio signal.

Figure 13 shows a capture of the AM modulated input into the circuit and the recovered 1 kHz sine wave output. The input signal had amplitude of 400 mV from peak to peak. It was modulated at 20% with

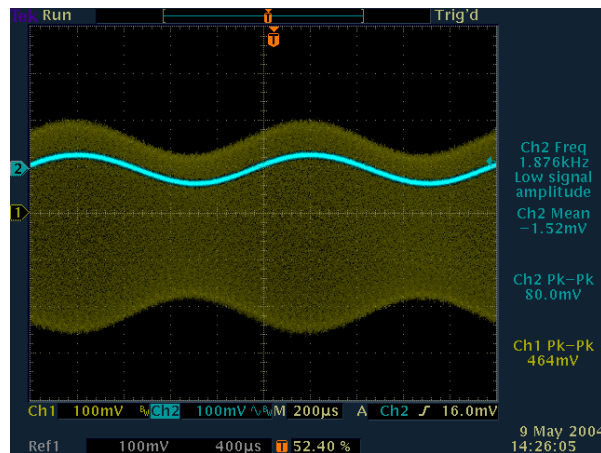


Figure 13: Demodulator input (yellow) and output (blue). The input was a 455 kHz carrier signal modulated by a 1 kHz sine wave.

a 1 kHz sine wave. The recovered sine wave had a peak to peak magnitude of 80 mV. The difference between the peak and trough magnitude on the inout wave form was only 80 mV (20% of the 400 mV input waveform). However, this magnitude was both positive and negative, the total difference being twice that. The end result was a demodulator that works quite well to produce very clean signals.

8 Pseudo-Stereo Generation:

While rare, there do exist AM radio stations that broadcast in stereo. Two or three are even within detection range of Baltimore. Such stations encode information about the two audio channels in the phase of the carrier signal, following Motorola's C-QUAM[™] stereo system definition. This system is such that standard AM detection will work on stereo stations, although there will only be 1 channel of information. In order to recover both the left and right audio information it is necessary to retrieve the phase information of the signal. Unfortunately, this method is quite involved and there a lack of exact information on the process; currently any stereo decoding is performed with prepackaged ICs and not discrete components. Instead, we have chosen to implement a system that provides stereo-like sound for all stations, not just those few broadcast in stereo. The basic principle behind our system is a small, frequency variable, phase delay. This allows the two output speakers to broadcast similar (but not identical) audio signals. In contrast to standard monaural sound, the system implemented produces sounds with a greater depth.

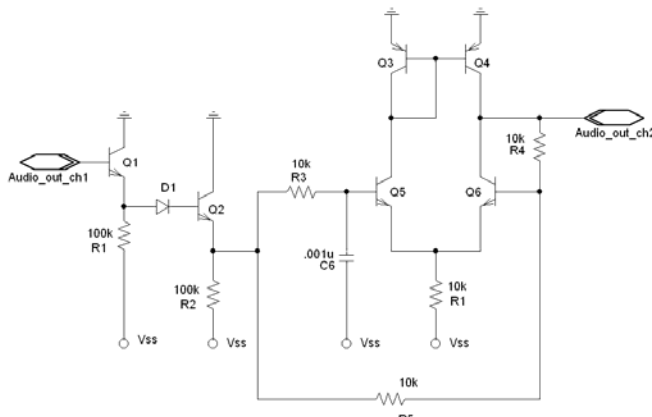


Figure 14: Schematic of the forced monaural delay audio channel generator

The all pass filter is fairly intuitive. The two inputs to the amplifier are almost identical, with the positive input having a small capacitor to Vss. This capacitor captures a small amount of energy and makes the positive and negative voltages slightly out of sync. Negative feedback acts to reduce this discrepancy and return the inputs to the same levels. This can never be quite achieved, leaving the output signal always slightly lagging behind the input. The amount of lag can be controlled by choosing the capacitor size properly.

The input to the circuit must be a consistently negative voltage. This is a condition imposed by the operating regime of the 4 BJT amplifier. When operated with a signal biased close to ground the amplifier will clip some of the positive voltage and distort the signal, something to be avoided at all costs. Instead, 2 BJTs and a diode (Q1,

The key component of the system is an all-pass filter. This device is an amplifier (Q3-Q6 in the schematic shown in Figure 14) with several resistors and capacitors that allow all frequencies in the range of interest (100 Hz to 20 kHz) to pass with unaffected amplitudes. However, depending on the frequency, the circuit will impose a small time delay on the signal. High frequency signals are affected by this delay more than lower frequency ones.

Q2 and D1 in Figure 14) are used to lower the operating voltage sufficiently for unimpeded operation. In addition, the two BJTs form emitter follower buffers, keeping the original channel 1 signal unaffected by any current draw necessary for channel 2.

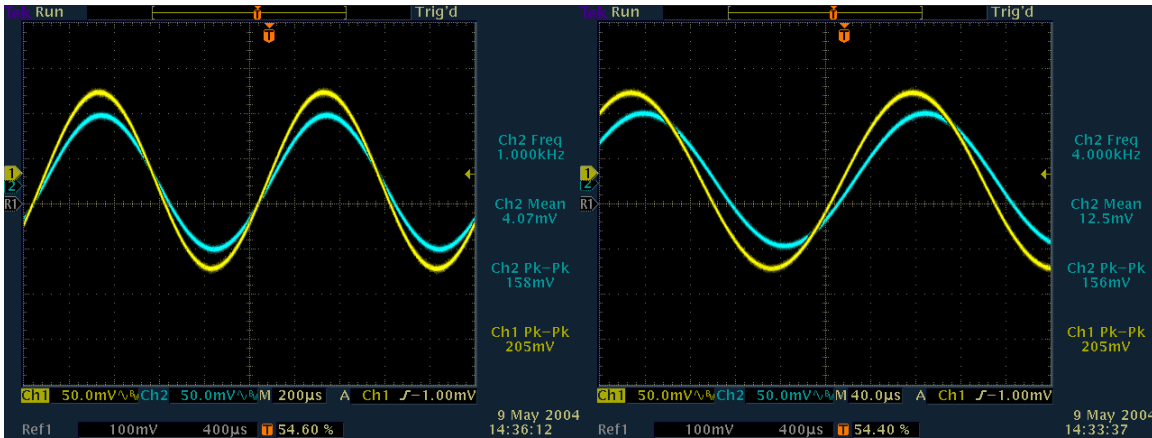


Figure 15: Measured waveforms demonstrating the phase delay of the circuit at different frequencies. Yellow represents the input and blue the output in both plots. The left plot shows the phase delay at 1 kHz while the right plot shows delay with a 4 kHz signal.

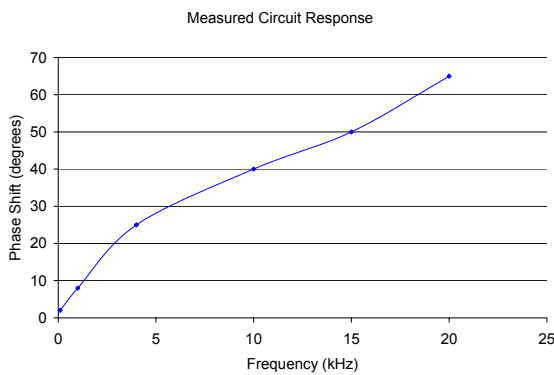


Figure 16: Plot of the measured phase delay for the Pseudo-Stereo Circuit.

The phase delay of the circuit varies with input frequency. Such phase delay is quite hard to measure on an oscilloscope, and accuracy is generally low. Phase delays were measured for various signals in the range of 100 Hz to 20 kHz. This is the frequency range of interest for audio signals. Figure 15 demonstrates two different phase delays at different frequencies. Table 2 summarizes the overall frequency delays. These values are displayed in Figure 16.

Frequency (kHz)	.1	1	4	10	15	20
Measured Shift (degrees)	2	8	25	40	50	65

Table 2: Measured phase shifts at various frequencies.

The total measured drop in voltage from the bias point of channel one to that of channel two was -1.93 Volts. This drop corresponded fairly well with the expected drop across the two BJTs and the diode. The resulting out, while negative, was still an acceptable input signal to the speakers.

9 Conclusion:

The completed AM radio appears to work quite well. In subjective testing of the entire system it was found that over 10 stations can be heard satisfactorily from the radio. Other stations vary with the time of day and placement of the antenna. It was likewise observed that the pseudo-stereo function of the radio gave a pleasing output. Tuning is somewhat problematic, due to the low precision available from the tuning capacitor. A

mechanical actuator that allowed a greater movement range would allow for better targeting of stations.



Figure 17: The completed radio with accompanying speakers.

Some noise is obvious when listening to stations that are relatively weak. It is likely that the majority of the noise is introduced by the components in the preamplifier. As mentioned in that section, resistors were used to generate bias points and currents. For a true LNA inductors would be used instead. However, this noise is still acceptable and was allowed in light of the true purpose of the preamp – to allow the reception of many stations that would not normally be receivable. On a more personal note, the project was quite enjoyable and it was very interesting to put the theory covered in the semester to work in a realistic manner. While time consuming, the experience did allow for a much greater understanding of analog bipolar circuits.