California State University Northridge

ECE492-493: Senior Design

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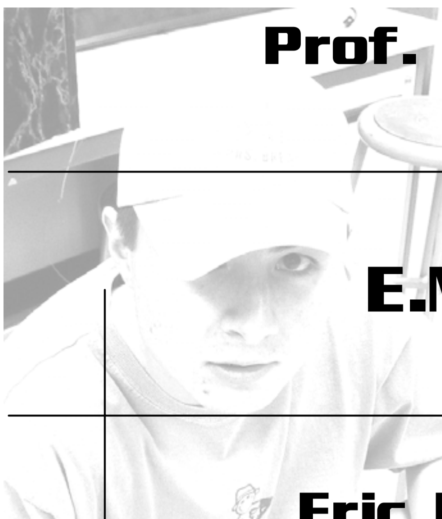
Prof. Ichiro Hashimoto



E.M.R. Stereo

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E.M.R. Stereo

Introduction

A stereo system is an electronic device used in many audio applications from basic home audio to more complicated studio applications. The function of the stereo is to provide a means of producing an audible output signal from an inaudible input signal. The external source can be a compact disc player, a record player, a videocassette recorder or a tape player.

Specifications

The specifications for the stereo are as follows:

- Preamplifier:
  All conventional inputs (CD, TV, VCR, tuner, tape and phonograph)
  EIA/FTC rules
  RIAA compensation
- Equalizer:
  Octave equalization
- Crossover:
  Electronic 3-way, Linkwitz-Riley
- Power Amplifier:
  2-channel with $150W_{RMS}$ per channel into an 8Ω load
- Speaker System:
  Designed using Thiele rules, sealed

Philosophy

The philosophy behind the project was *divide and conquer*. The National Institute of Standards and Technology gives the definition of *divide and conquer* as "an algorithmic technique". To solve a problem on an instance of size *n*, a solution is found either directly because solving that instance is easy, typically because the instance is small, or by *dividing* the instance into two or more smaller instances. Each of these smaller instances is *recursively* solved and the solutions are combined to produce a solution for the original instance.

This approach is paralleled in the E.M.R. Stereo – each stage is addressed and solved separately and then combined with other stages to deliver the end product. The stages to be implemented are preamplifiers, equalizer, crossover, power amplifier and speakers.

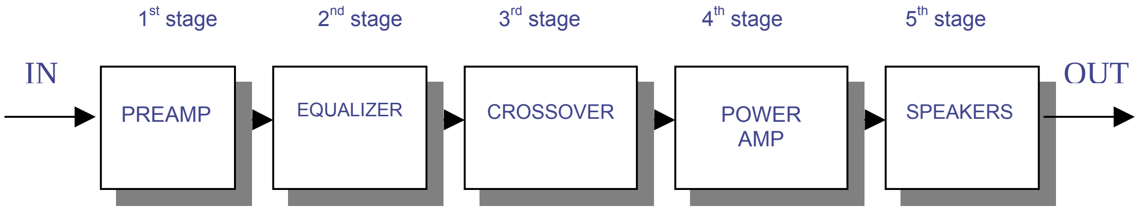


Fig. 1: Stage diagram

Modules

Using the technique of divide and conquer, the E.M.R. Stereo has been divided into five main modules. The first of them is the preamplifier stage consisting of all the inputs to the stereo. The next module is the equalizer, which allows the user to manipulate the input signal in different ranges of frequency. The crossover module then splits the input signal into three different ranges that are fed into the power amplifiers. From there the signal reaches the speaker. These modules are built separately, tested and then combined to form the final product.

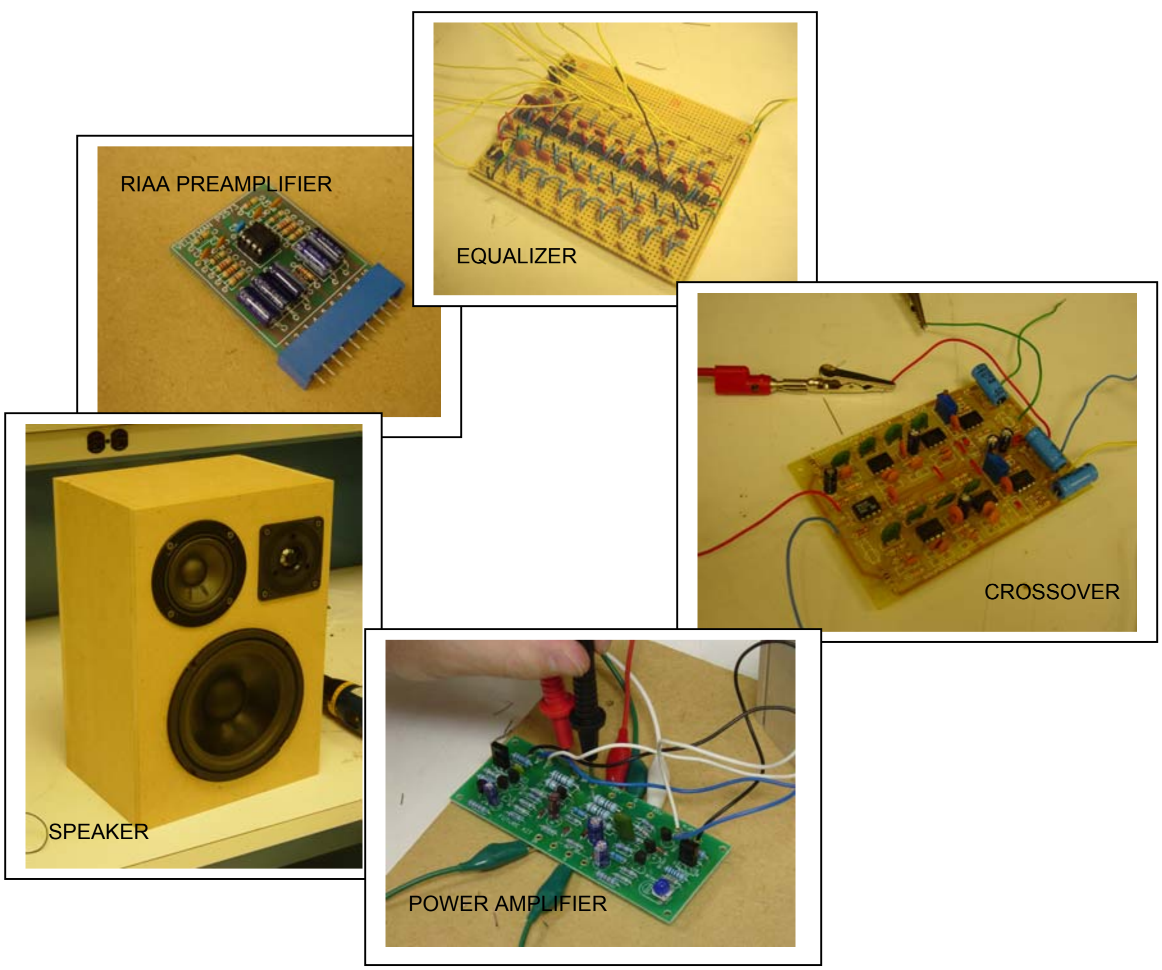


Fig. 2: Modules

Preamplifier

DESCRIPTION

A preamplifier is an audio component that adjusts the volume of an audio signal between input devices and an amplifier or group of amplifiers. The preamplifier's primary task is to control the volume.

RIAA Preamplifier

A typical record player outputs a very low level signal, typically 0.5mV. This signal needs to be taken from the millivolt range and amplified to at least 1V. The RIAA equalization is used in the playback to reduce the high-pitch noise and maximize bass dynamics in the phono playback. The audio material that is recorded on the record has been pre-equalized so that the frequency response of the whole chain from the mixing desk to a speaker will produce a flat frequency response. The RIAA compensation curve was adopted in the mid 1950s as a way of dramatically improving the fidelity of playback. This curve takes into account the limitations of the recording system mechanisms on the record surface. At low frequencies, this amplifier provides 20dB of gain. At medium frequencies it provides no gain. At high frequencies it provides 20dB of attenuation. As a result, RIAA equalization has an approximate 40dB variation over the audible range.

NAB Preamplifier

The present 15 in/s NAB equalization was developed around 1950 using 3M Type 111 tape. The standardized NAB recorded flux rises 6dB per octave below 50Hz, it's flat between 50Hz and 3.15kHz, and falls at 6dB per octave above 3.15kHz. Typical equalizations are +8dB at 20Hz and +11dB at 16kHz. This is similar to the RIAA compensation but the high frequencies also get boosted. While the public still uses records for their "nostalgic" quality, most do not use tapes today and recording devices such as compact disks have made tapes obsolete. All tape decks or players made today have some sort of equalization already built into the player itself. The available NAB compensation chips on the market are only available in reels of 2500 and are not cost effective for this project. For this reason, NAB compensation is not included in the preamplifier section.

VOLUME

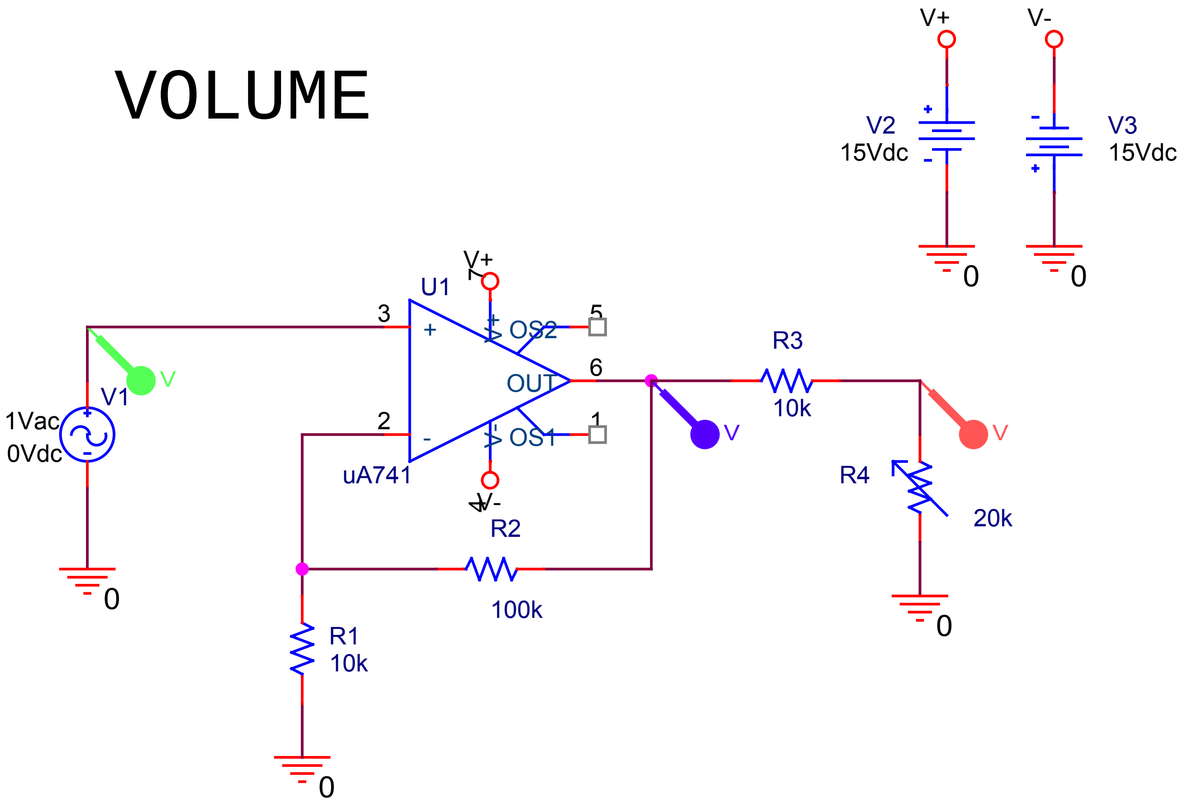


Fig. 3: Volume circuit

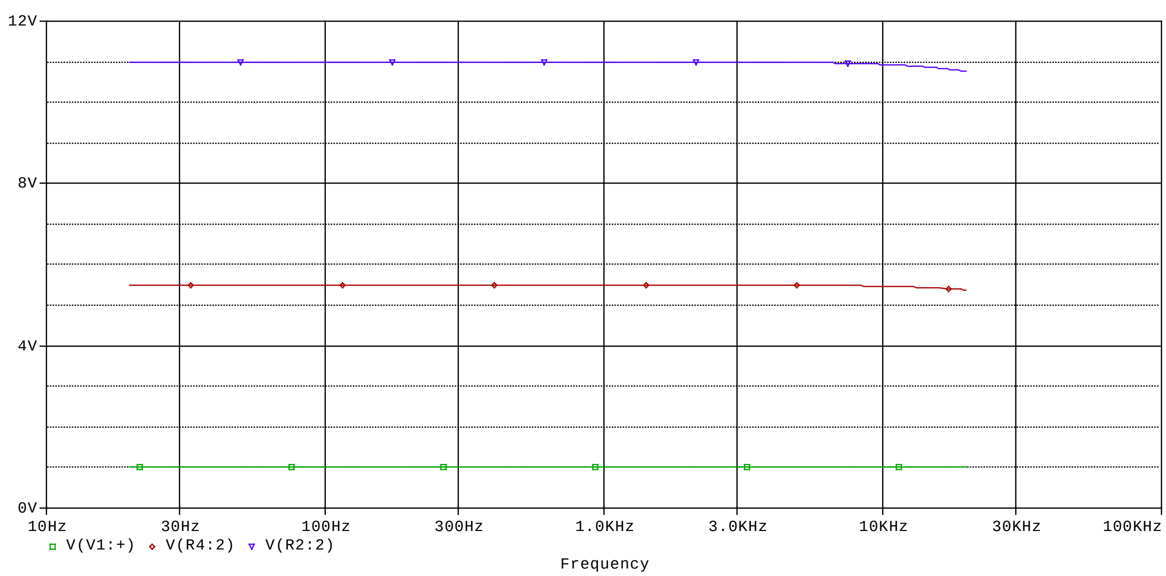


Fig. 4: AC sweep from 20Hz to 20kHz

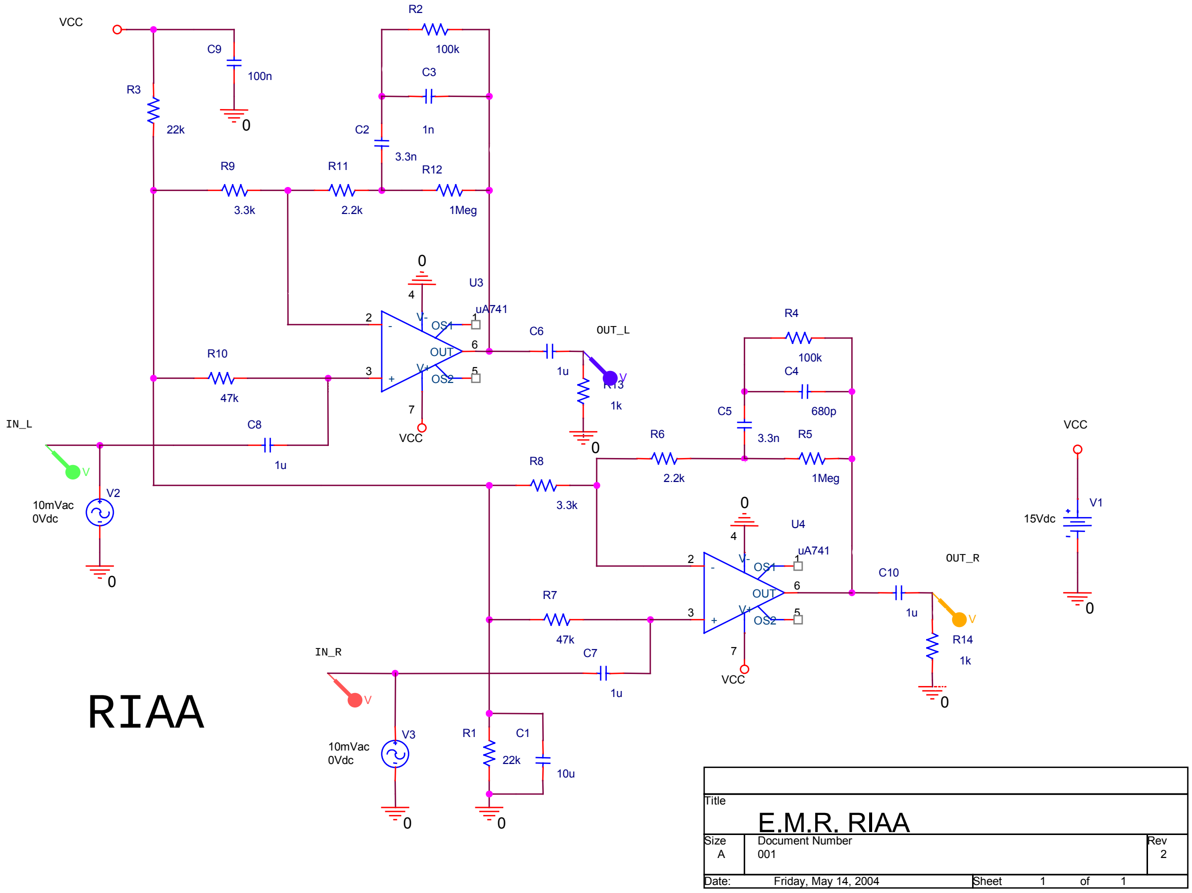


Fig. 5: E.M.R. RIAA preamplifier

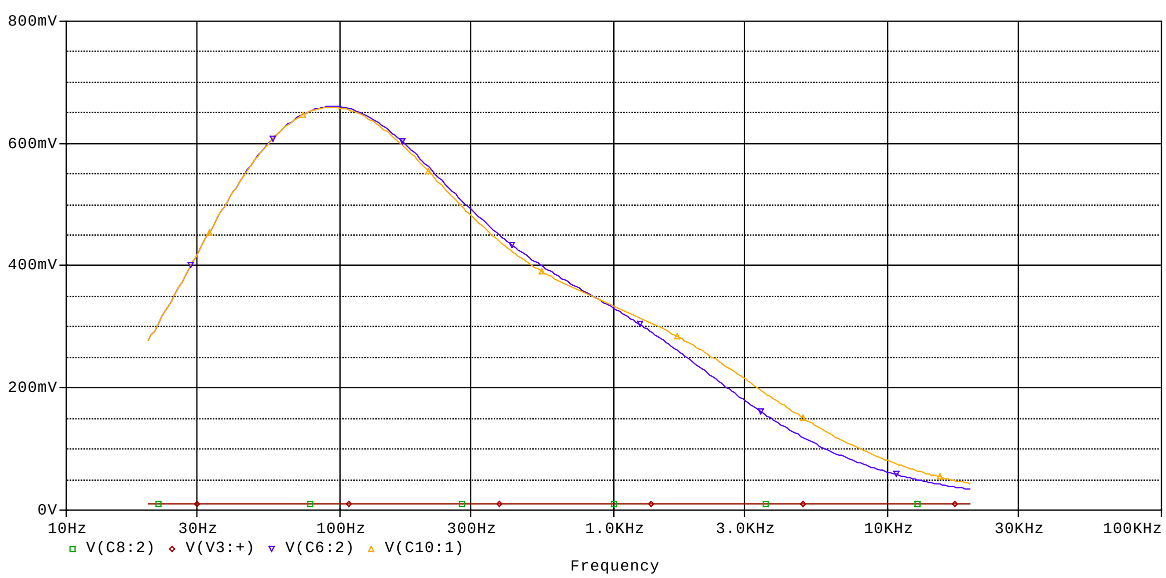


Fig. 6: AC sweep from 20Hz to 20kHz

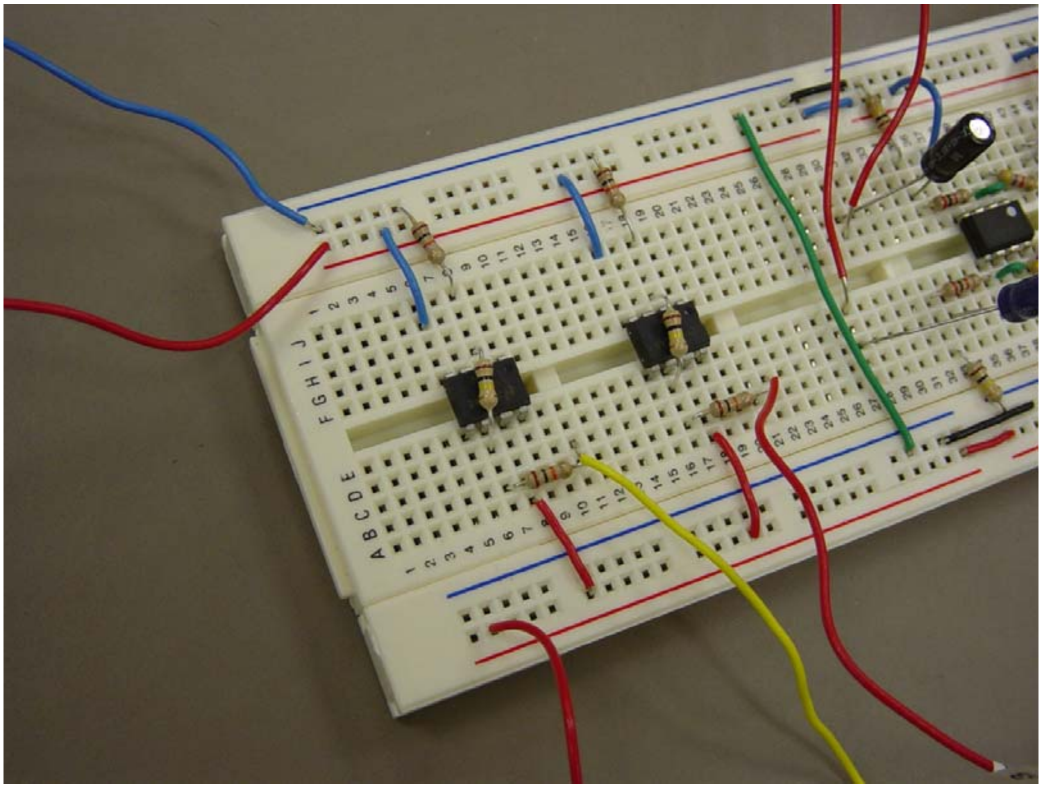


Fig. 7: E.M.R. volume circuit

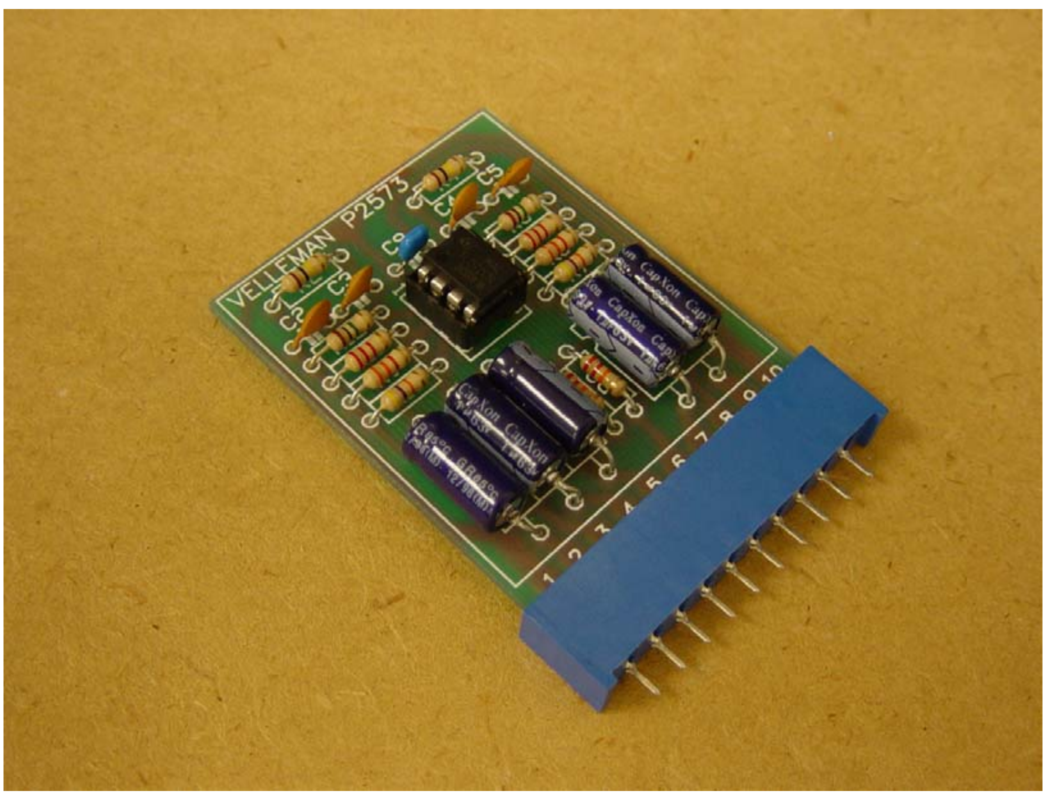


Fig. 8: E.M.R. RIAA preamplifier

Equalizer

DESCRIPTION

An equalizer is an electronic circuit capable of changing the frequency response of a given input audio signal. Its primary task is to manipulate a single frequency band within a specific set of frequencies that usually range from 20Hz to 20kHz, the audible frequency range for humans. The purpose of such a circuit is to provide a certain degree of flexibility when it comes to compensating the sound differences of an input over the audible frequency or to adapt the output signal to a specific environment.

Equalizers can be classified depending on their design and characteristics. Most commonly, equalizers are classified as parametric and graphic equalizers. Parametric equalizers are the most versatile and most precise because they can modify the characteristics of a specific frequency within the audible range (amplitude, center frequency and bandwidth can be changed). Graphic equalizers are the more popular and widely used because they can modify the characteristics of a specific frequency within the circuit (amplitude is the only changeable parameter).

Equalizers are further divided into 1/3 octave, 2/3 octave and full octave depending on the mathematical relationship between the frequencies. These terms define how the frequencies of an equalizer are spaced between each other. In a full octave equalizer, for instance, the next frequency is an octave higher than the previous one (64Hz and 32Hz – $\left(\sqrt[3]{2}\right)^3$ ratio) whereas in a 1/3 octave the next frequency is 1/3 of an octave above the previous one (80Hz and 100Hz – $\left(\sqrt[3]{2}\right)^1$ ratio).

E.M.R. EQUALIZER

The E.M.R. equalizer is a 10-band full-octave graphic equalizer entirely built from discrete components (operational amplifiers, resistors and capacitors). The equalizer is composed of twelve µA741 operational amplifiers and it can be divided into three stages: the first is the input stage, the second is the equalizer stage, and the third is the output stage. The input stage lowers the input before it’s manipulated by the equalizer stage and the third stage boots the signal back to an acceptable level. The second stage can be described as a cascade of ten chips, one per band. A combination of resistors and capacitors sets a specific frequency and performs the filtering. An array of variable resistors changes the characteristics of each one of the frequencies. The lowest frequency is 32Hz and the highest is 16kHz.

The stereo is composed of a left and a right equalizer channel for a total of two identical channels. The equalizer receives input from the preamplifier circuit and sends the output to the crossover circuit.

The following values describe the behavior of the circuit:

Equalizer (left channel)			
Frequency	V_{in}	V_{band}	V_{out}
32Hz	1.00V	618mV	575mV
64Hz	1.00V	806mV	920mV
125Hz	1.00V	768mV	987mV
250Hz	1.00V	762mV	1.03V
500Hz	1.00V	718mV	900mV
1kHz	1.00V	775mV	931mV
2kHz	1.00V	762mV	950mV
4kHz	1.00V	787mV	1.01V
8kHz	1.00V	768mV	1.01V
16kHz	1.00V	656mV	980mV

Equalizer (right channel)			
Frequency	V_{in}	V_{band}	V_{out}
32Hz	1.00V	790mV	756mV
64Hz	1.00V	831mV	987mV
125Hz	1.00V	800mV	1.00V
250Hz	1.00V	770mV	1.01V
500Hz	1.00V	718mV	900mV
1kHz	1.00V	800mV	920mV
2kHz	1.00V	768mV	950mV
4kHz	1.00V	781mV	1.00V
8kHz	1.00V	775mV	1.03V
16kHz	1.00V	650mV	975mV

Table 1: E.M.R. equalizer values

Equalizer (32Hz and 64Hz)

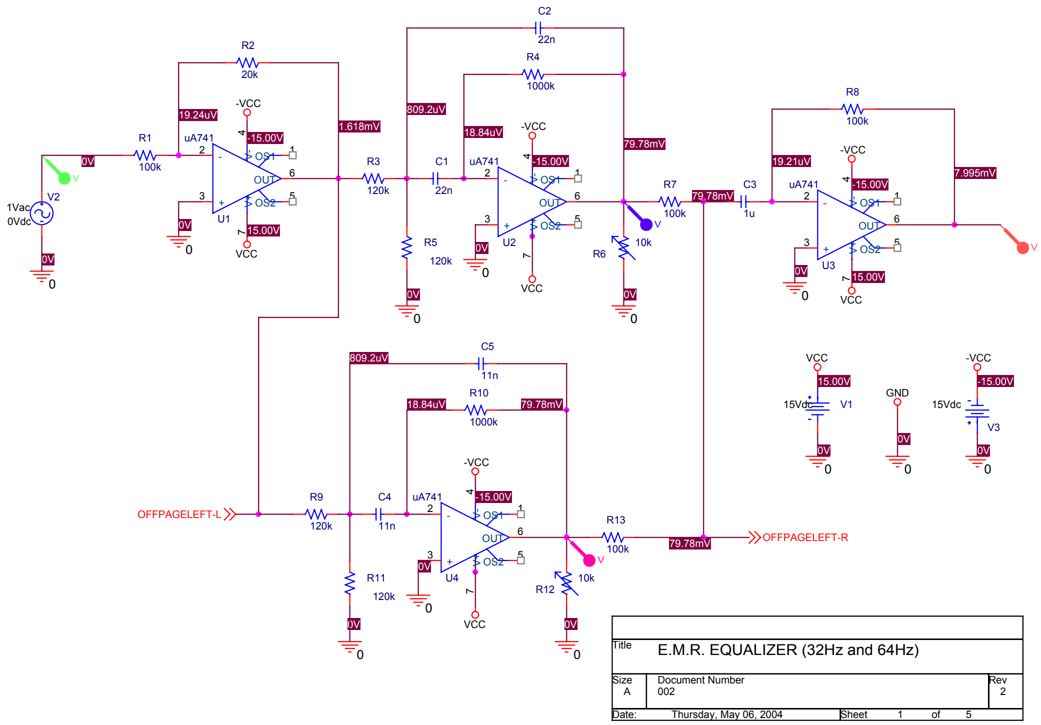


Fig. 9: E.M.R. equalizer (32Hz and 64Hz)

Equalizer (125Hz and 250Hz)

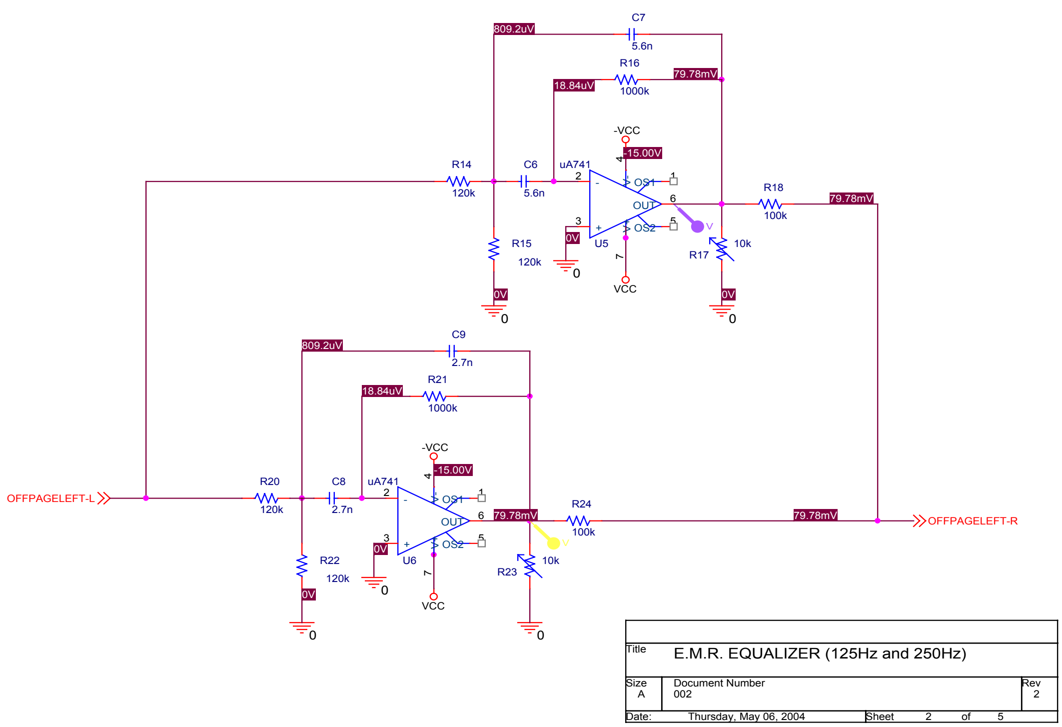


Fig. 10: E.M.R. equalizer (125Hz and 250Hz)

Equalizer (500Hz and 1KHz)

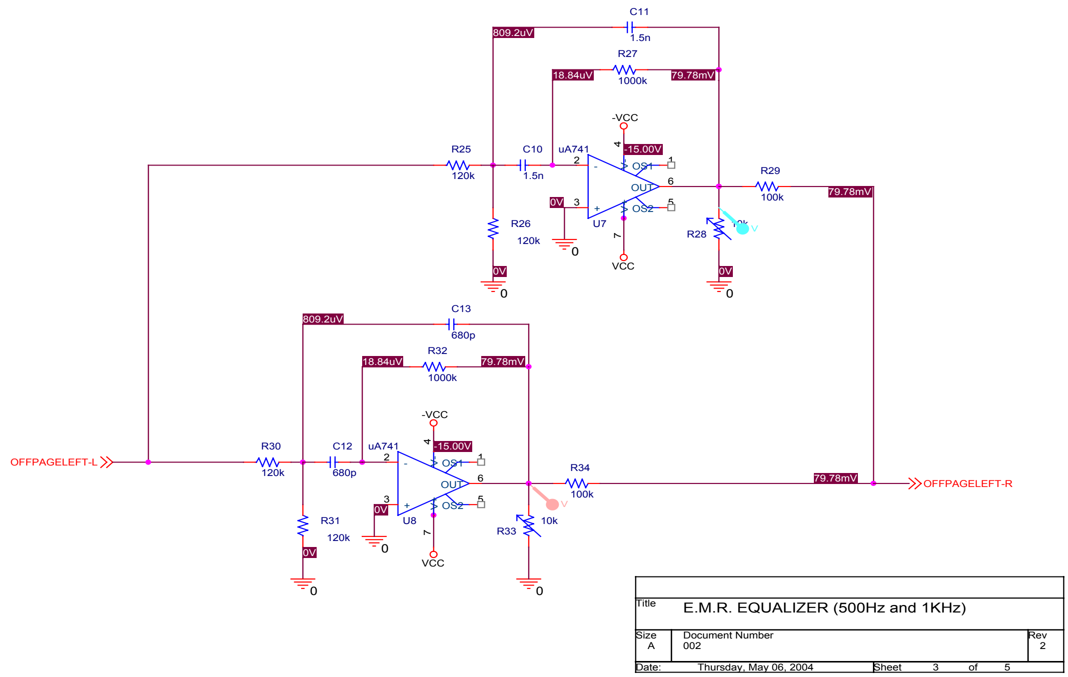


Fig. 11: E.M.R. equalizer (500Hz and 1kHz)

Equalizer (2KHz and 4KHz)

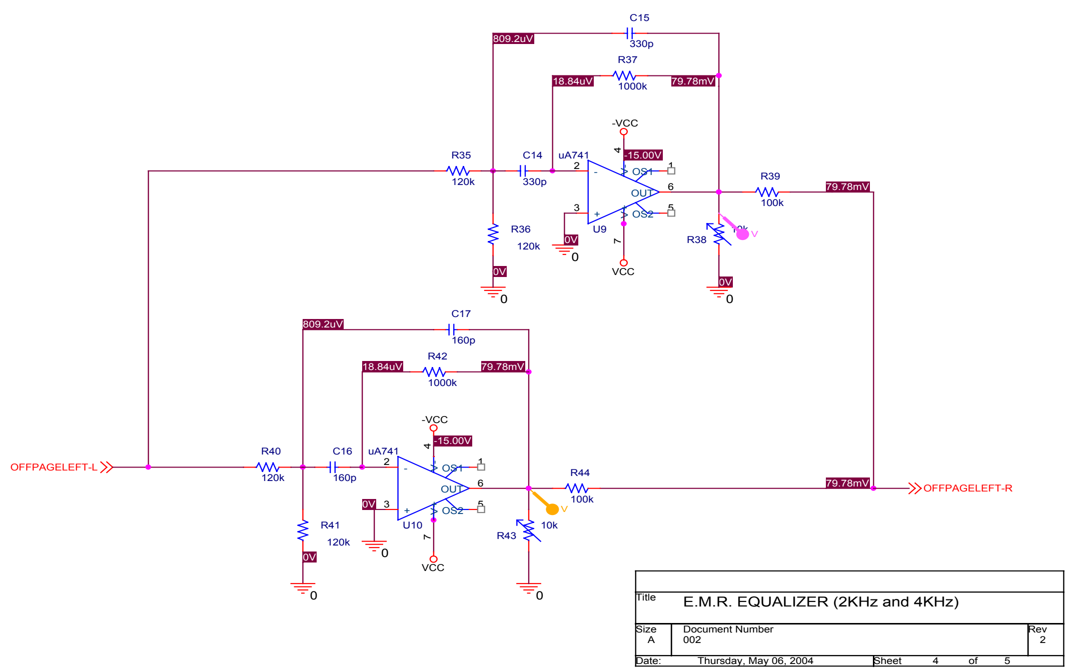


Fig. 12: E.M.R. equalizer (2kHz and 4kHz)

Equalizer (8KHz and 16KHz)

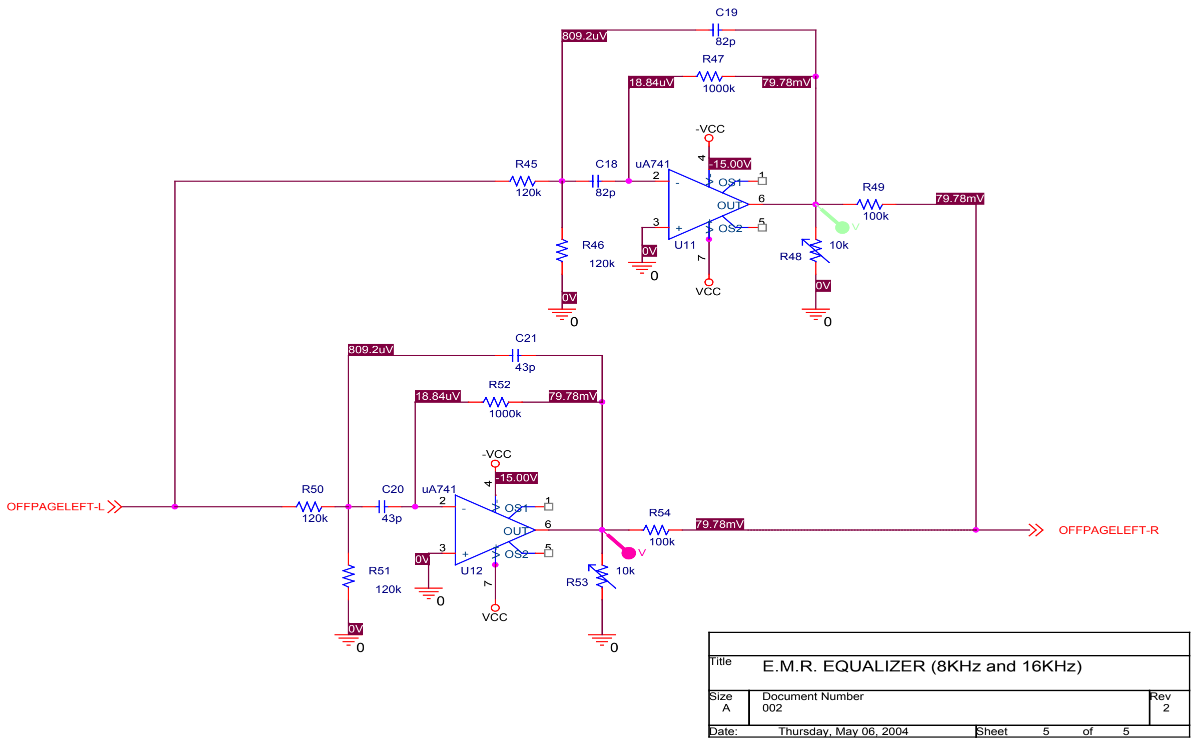


Fig. 13: E.M.R. equalizer (8kHz and 16kHz)

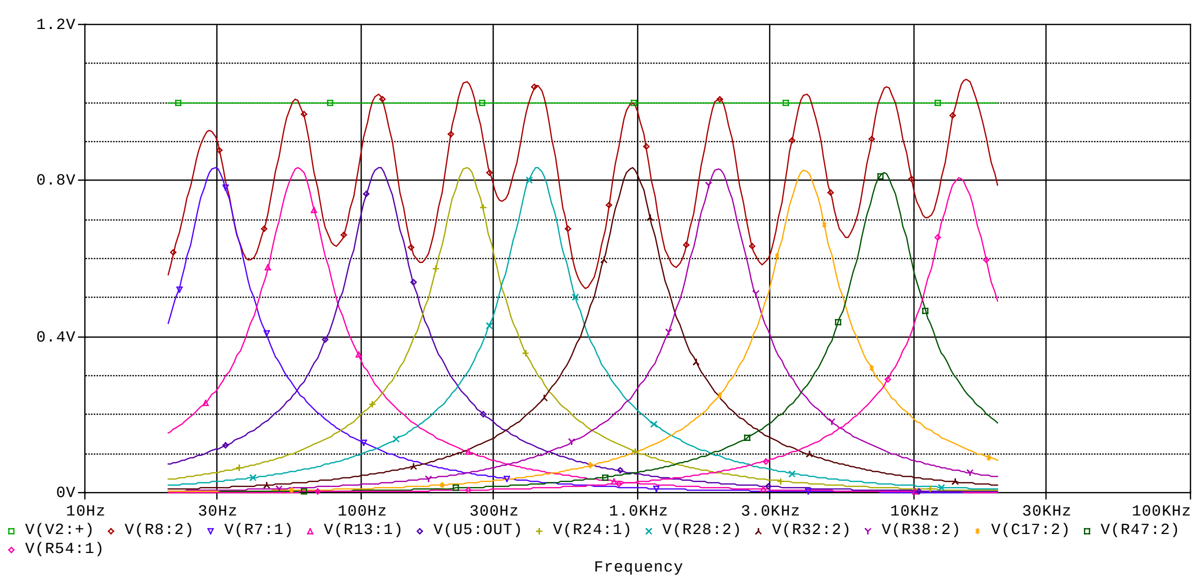


Fig. 14: AC sweep from 20Hz to 20kHz

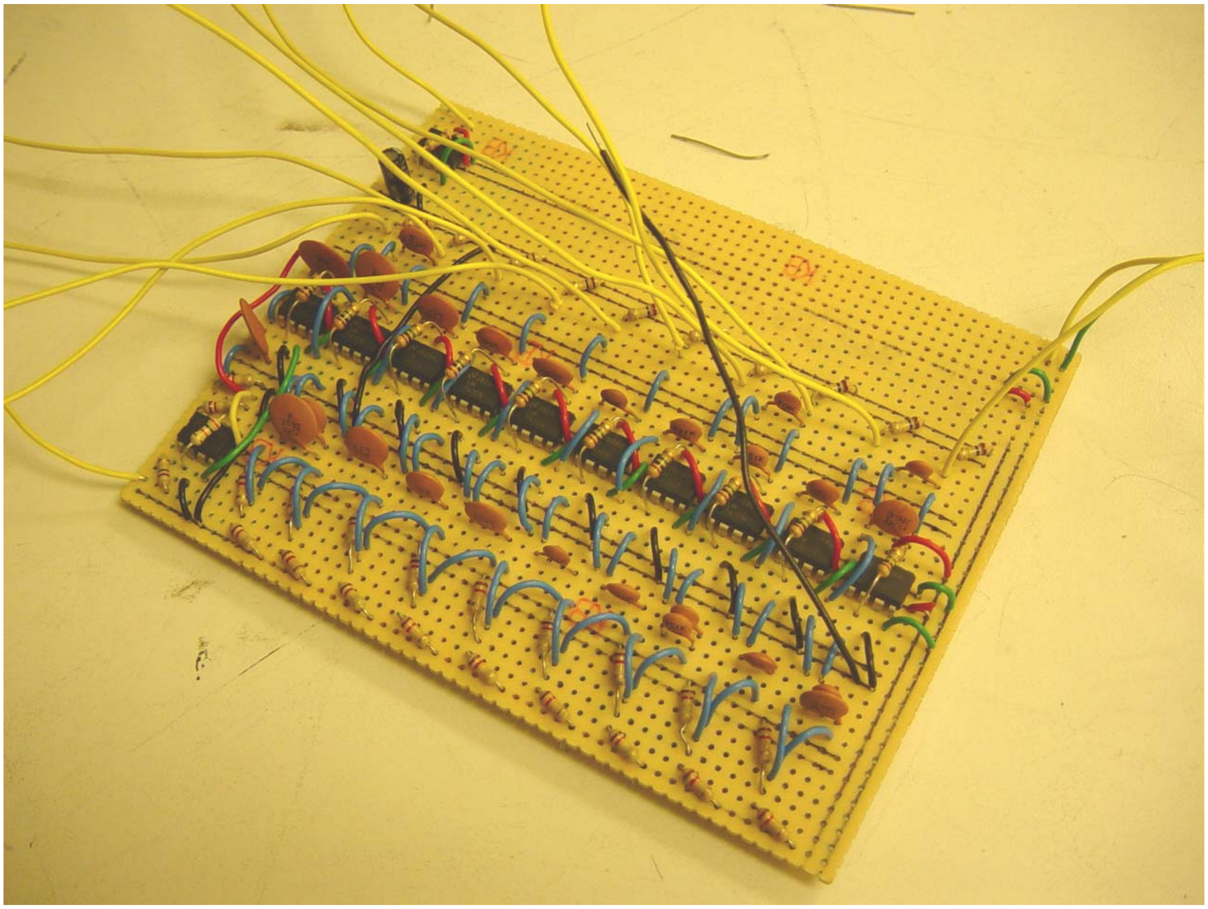


Fig. 15: E.M.R. equalizer

Crossover

DESCRIPTION

The crossover is an electronic circuit that splits the audible frequency into several sections and delivers each one of them to a power amplifier stage. Crossovers are necessary in stereo systems because speakers cannot handle the entire audible spectrum. Crossovers, in other words, distribute more efficiently the audio signal to the speakers.

Depending on their design, crossovers are usually divided in two categories: Butterworth and Linkwitz-Riley. The Butterworth crossover consists of low-pass filters and it's widely used in speaker design whereas the Linkwitz-Riley crossover is a combination of two second-order 12-decibel per octave Butterworth filters and it's frequently used in subwoofers.

Crossovers can also be classified as active and passive. The only distinction is whether the crossover can be adjusted (active) or not adjusted (passive). In order to change the crossover settings, it is necessary to have an internal AC power source.

Crossovers come in 2-way, 3-way and 4-way versions. This kind of distinction describes how many regions the input signal will be divided into. A 3-way system, for instance, splits the input into 3 bands: low, mid and high. The regions are separated by the so-called cut-off frequencies which set the boundaries between the regions (in a 3-way crossover there are two of them). Although 3-way systems are very popular, 2-way and 4-way systems are not uncommon. Other designs are possible so it's virtually possible to design a crossover to split the audible frequency into as many bands as needed.

E.M.R. CROSSOVER

The E.M.R. crossover is an active 3-way Linkwitz-Riley crossover from E.S.P. The circuit itself is made up of eight TL072 operational amplifiers (the schematic for the crossover shows µA741). The input stage provides the signal to the filters for the three bands. The high band is a high-pass filter (2 TL072), the low band is a low-pass filter (2 TL072) and the mid-band is a series combination of the other two bands (4 TL072). By using the E.S.P. Linkwitz-Riley Crossover Calculator and a combination of resistors and capacitors, the lower cut-off frequency was set to 341Hz and the upper cutoff frequency was set to 3.521kHz. A great deal of troubleshooting has been done on the circuit but since the mid-band for the E.S.P. crossover was never obtained, an equivalent mid-band circuit was built from discrete components on a separate board.

The crossover receives an input from the equalizer and sends the output to the power amplifiers.

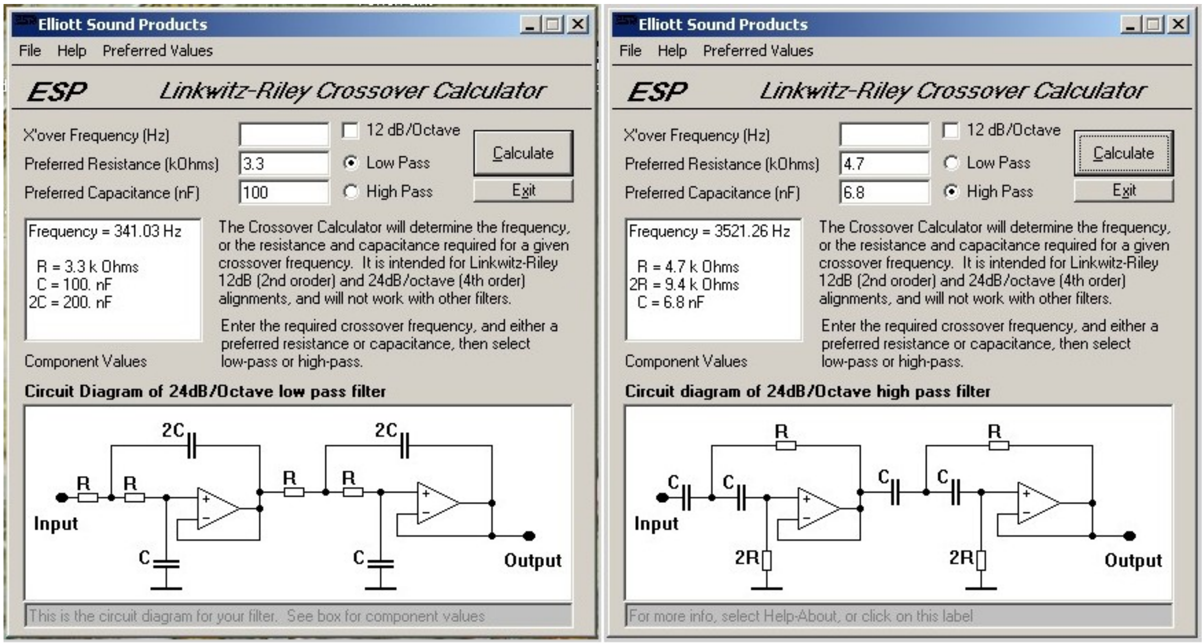


Fig. 16: E.S.P. Linkwitz-Riley Crossover Calculator screenshots for low pass and high pass

The mathematical relationships between frequency, capacitors and resistors are the followings:

$$f = \frac{1}{2\pi\sqrt{2}RC} \qquad C = \frac{1}{2\pi\sqrt{2}Rf} \qquad R = \frac{1}{2\pi\sqrt{2}Cf}$$

The following values describe the behavior of the circuit:

Crossover (left channel)			
Band	Frequency	V_{in}	V_{out}
High	16kHz	1.00V	950mV
	10kHz	1.00V	915mV
	7kHz	1.00V	830mV
Mid	3kHz	1.00V	825mV
	2kHz	1.00V	944mV
	1kHz	1.00V	981mV
Low	200Hz	1.00V	1.453V
	100Hz	1.00V	1.859V
	32Hz	1.00V	1.937V

Crossover (right channel)			
Band	Frequency	V_{in}	V_{out}
High	16kHz	1.00V	950mV
	10kHz	1.00V	920mV
	7kHz	1.00V	837mV
Mid	3kHz	1.00V	837mV
	2kHz	1.00V	950mV
	1kHz	1.00V	975mV
Low	200Hz	1.00V	1.280V
	100Hz	1.00V	1.844V
	32Hz	1.00V	1.953V

Table 2: E.M.R. crossover values

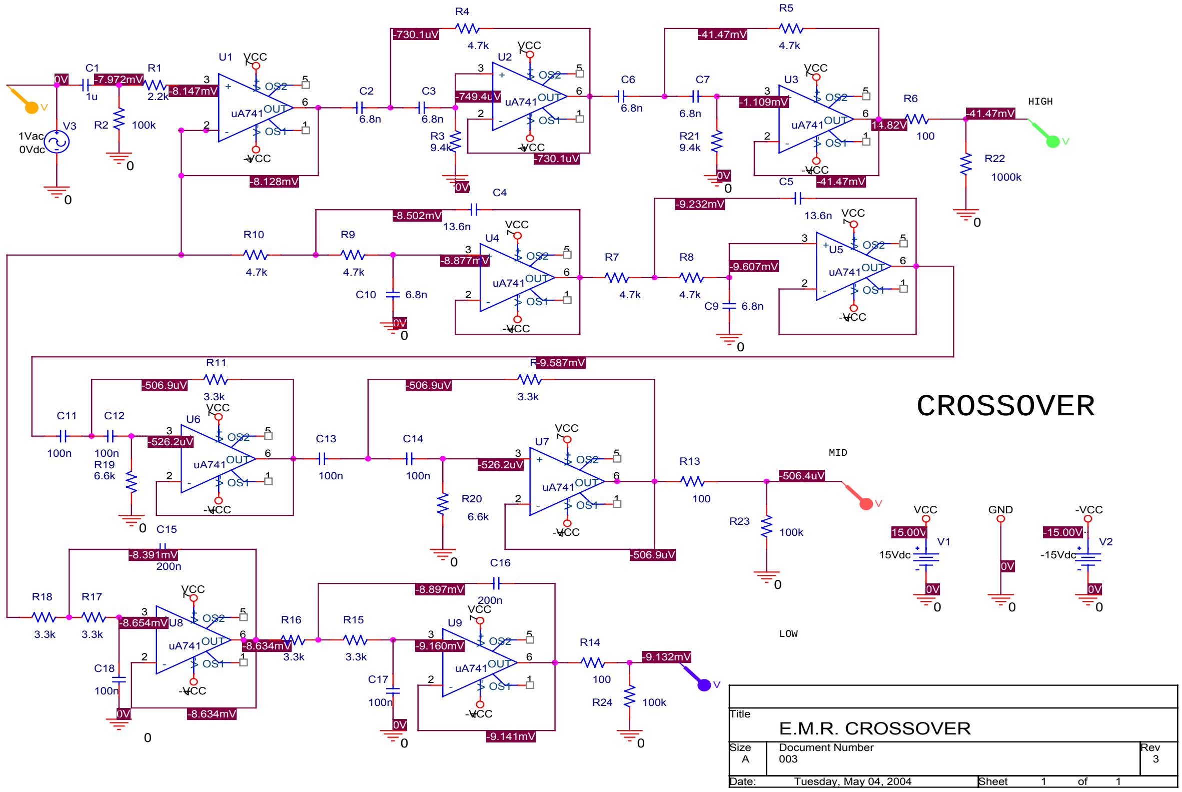


Fig. 17: E.M.R. crossover

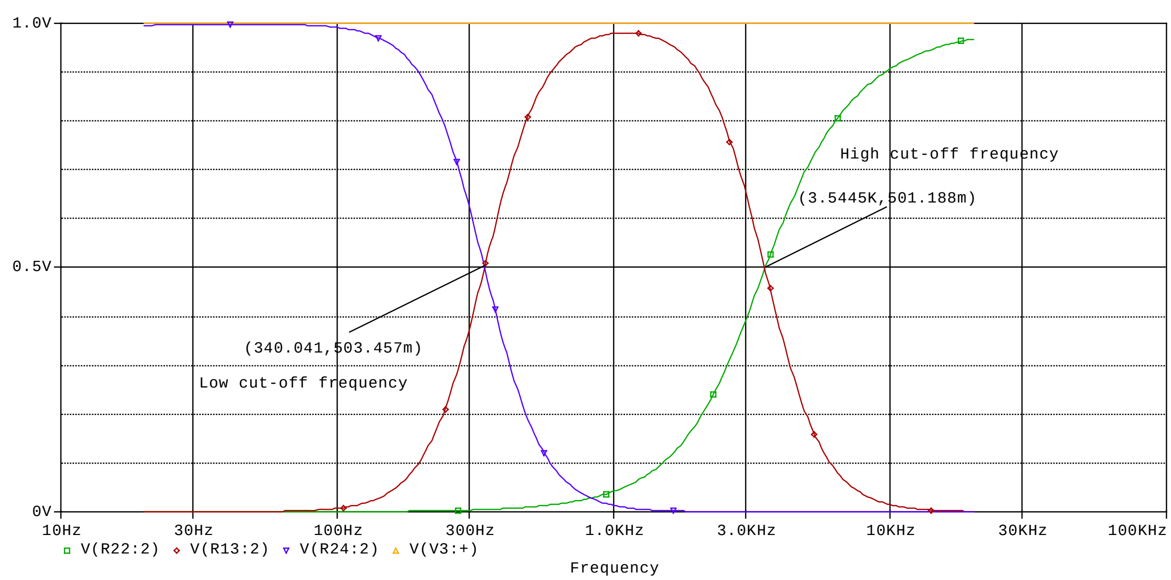


Fig. 18: AC sweep from 20Hz to 20kHz

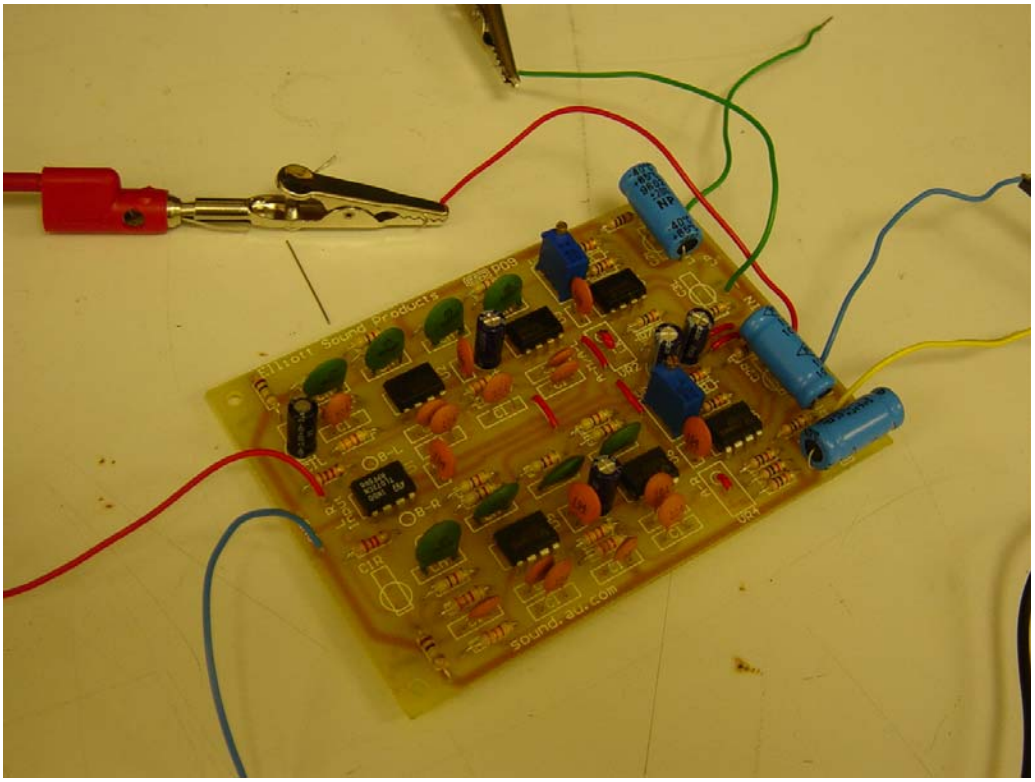


Fig. 19: E.M.R. crossover (highs and lows)

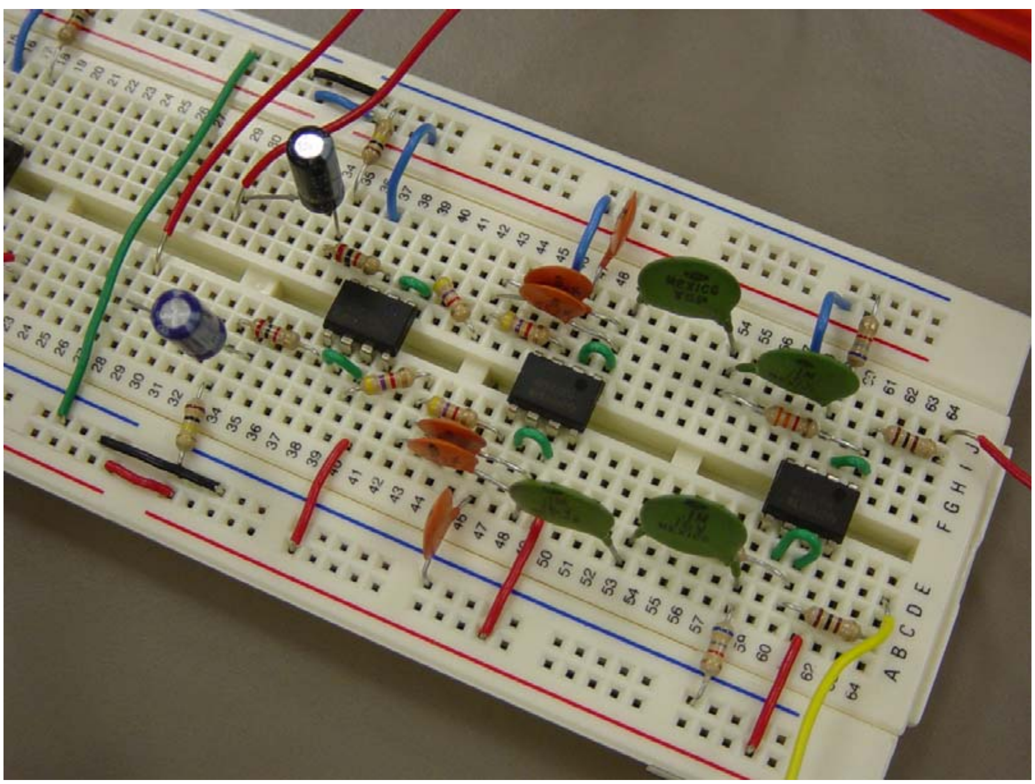


Fig. 20: E.M.R. crossover (mids)

Power amplifier

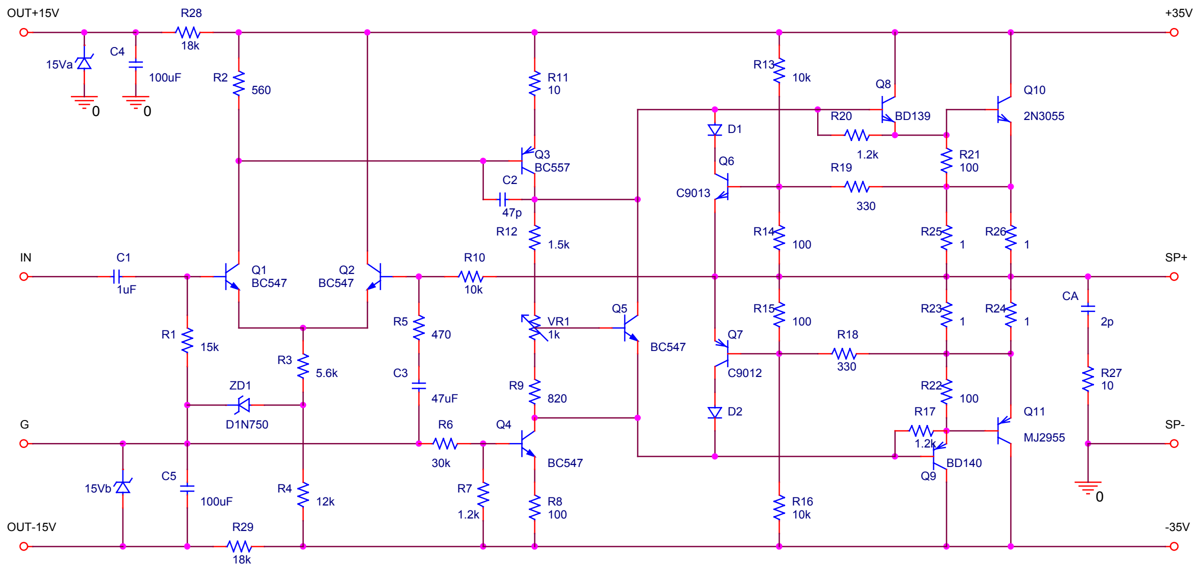
DESCRIPTION

Audio power amplifiers have been used since the 1920s. Most modern devices that transmit audio signals use some type of amplifier. The most targeted area for today's use of these devices is the audio world which has applications ranging from simple radio to complicated recording studios.

Regardless of the application, the function of a power amplifier is the same. Power amplifiers are used to convert a signal (a line-level signal in the case of audio power amplifier) to a respectively high-voltage high-current output signal. There are a variety of amplifiers, each corresponding to a particular application. Common classifications are Class A, Class B and Class AB. Class A provides the best sound quality and frequency regularity. Thanks to its always-on status, the output devices that provide power never shut down regardless of what part of the signal is being amplified. By contrast, a class B device shuts down transistors when they are not called upon to reproduce a signal (if a negative signal is presented, the positive output device will shut off until a positive signal is presented and vice versa). Class A provides smooth and continuous power while Class B provides intermittent power only as needed. Class B components function efficiently but produce poor quality results because of their start and stop nature. Conversely, Class A components produce superior quality but run inefficiently, producing large amounts of heat and necessitating massive power supplies and associated manufacturing quality. As a compromise between the two classes, Class AB design retains power at all times but decreases output when a signal is not present. In this way, Class AB devices can save power output like a Class B design without suffering from the severe start and stop distortions that plague Class B designs.

E.M.R. POWER AMPLIFIER

The E.M.R. Stereo uses a complementary class AB 50W power amplifier, which has an input speed at 1V and input impedance at 15kΩ. Six amplifiers will handle all of the amplification needed for each speaker for both left and right channels. The design of these amplifiers was outsourced through a company called Quality Kits. Empty boards and components were included in the kits. Assembly and troubleshooting was required. The amplifiers require ±35V DC. The boards include ±15V DC connections to be used to power up other components.



POWER AMPLIFIER

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Fig. 21: E.M.R. power amplifier

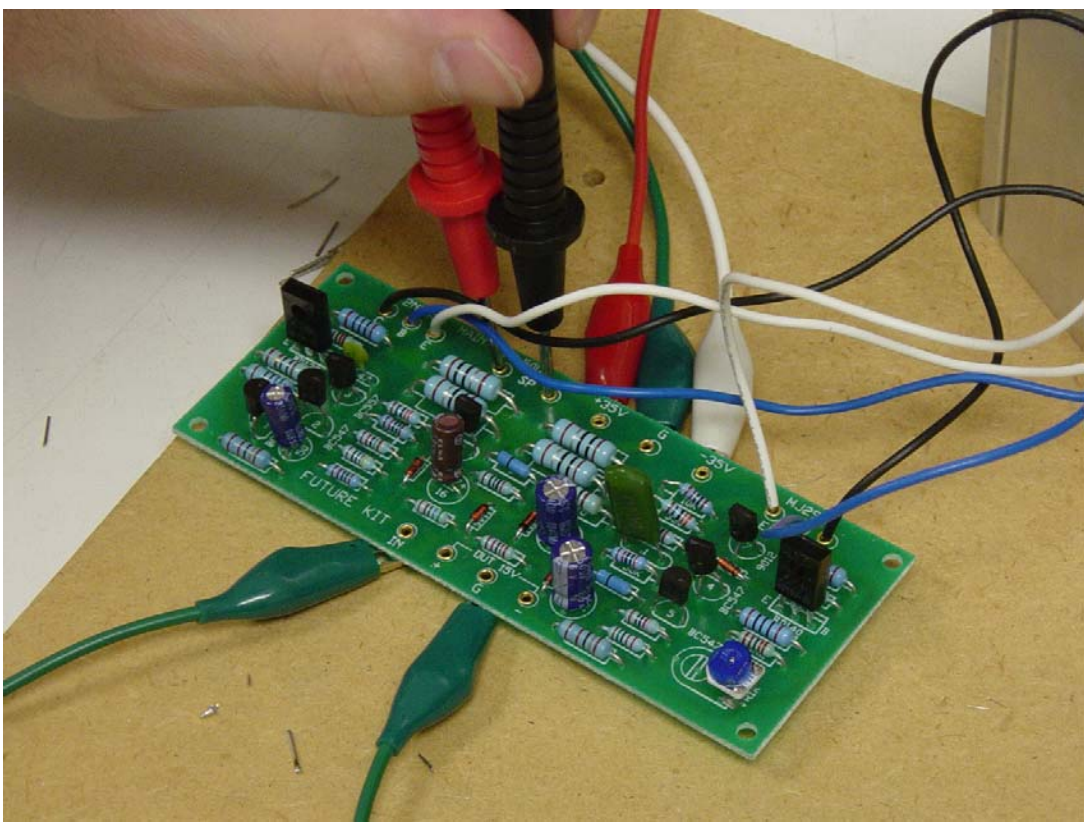

Fig. 22: E.M.R. power amplifier

Speaker

DESCRIPTION

A speaker is an electromechanical device that uses both electricity and mechanical operation to produce sound. There are three main types of speakers in the typical configuration of cone and magnet speakers: woofers, mid-range and tweeters.

Woofers are large speakers capable of reproducing low sounds and typically produce the majority of the bass from a musical program. These speakers generally range in size from 8” to 18” in diameter.

Mid-range speakers are capable of accurately reproducing the majority of the vocal frequencies in a musical program along with other accompaniment sounds. These speakers are generally smaller than woofers and range from 3” to 6.5”.

Tweeters are speakers that are designed to reproduce only the high frequencies. Designated by their small size, tweeters are commonly found in sizes ranging from ½” to 2”.

A speaker contains some variation of the following parts. The basket assembly is a casing around all of the speaker components. The magnet is attached to the basket and the basket provides support for the speaker cone which moves back and forth. The cone is attached to the basket with a soft foam or rubber surround. This allows the cone to remain attached to the basket but also provides movement. In the middle of the speaker is a dust cap. Under the dust cap is the pole piece that is part of the magnet assembly. This dust cap also covers what is called the voice coil, a coil that is made by wrapping a small wire many times around the bottom of the cone, where it forms a tube. This tube, with the wire wrapped around it, will be inserted into the gap between the pole piece and the magnet on the magnet assembly. The spider is a little piece of material that keeps the voice coil centered in the gap between the pole piece and the magnet. This prevents the voice coil from contacting the side of the gap instead of producing a vertical motion. When a signal is sent through the voice coil, a miniature electromagnet is created. When two magnets of the same pole are close, the magnets push each other away. Similarly, two poles of opposite nature attract each other. By applying a signal to the voice coil, the coil will become positive or negative depending on the signal that is currently being sent through it. This signal changes all of the time and it causes the speaker cone to move up and down. This motion creates the sound that a person can hear.

E.M.R. SPEAKER

The E.M.R. Stereo speakers use 8” woofers and are dedicated to producing the signals from the low-band in the frequency range 20Hz-341Hz. The mid-range speakers are 5” and are committed to producing the frequencies in the range of 341Hz-3.521kHz. The tweeters are 1” in diameter and produce the detail required in the high-frequency range of 3.521kHz-20kHz.

Enclosure

DESCRIPTION

Every speaker needs an enclosure of some type. Enclosures vary by speaker, but for the most part there are three types.

The most common type is the sealed enclosure. This type of enclosure can be found everywhere and is basically a completely sealed box. The speaker mounts in the enclosure and when the cone moves back and forth it pressurizes the air in the enclosure. This type of enclosure produces tight and accurate bass response.

For more bass output, a ported enclosure is common. This allows air that is behind the speaker to come out of the enclosure via a tube or slot on the same plane as the speaker itself. If done correctly, the waves from the back of the speaker will combine with the waves from the front of the speaker, thus making it louder, but reducing the accuracy of reproduction of the original sound.

The last type of enclosure is referred to as an infinite baffle. The baffle is a piece of wood or other material that the speaker is mounted to which prevents air from behind the speaker to mix with air in front of it. Speakers that are mounted in a ceiling would be a good example of an infinite baffle enclosure (where there is really no enclosure at all).

E.M.R. ENCLOSURE

The E.M.R. Stereo uses the first type of enclosure mentioned above: the sealed enclosure. This choice depends on three factors. Firstly, the sound output of the enclosure will more closely represent the audio signal being sent to the speakers. This is due to the fact the other two designs will result in peaks in the low frequency response. Secondly, the enclosure will have a standard shape and size and therefore it will be easier to build and design. Thirdly, cost is reduced. While the infinite baffle design is the most cost effective design, mount speakers in a wall for a presentation would not be practical. The sealed design is therefore the most reasonable and cost effective method.

The enclosure is designed around the woofer's specifications and the builder's preferences. The woofer's free air resonance is 53Hz, so the enclosure was designed to complement that and built so that the f_{-3dB} point would be close to 53Hz. Using an available enclosure volume calculator, the AJ Sealed Designer 2.0, the electrical and mechanical properties of the driver are input to the program and the volume of the enclosure that produced an f_{-3dB} point of 59Hz resulted to be 0.6ft^3. An f_{-3dB} point of 59Hz is quite common with sealed enclosures containing an 8" driver.

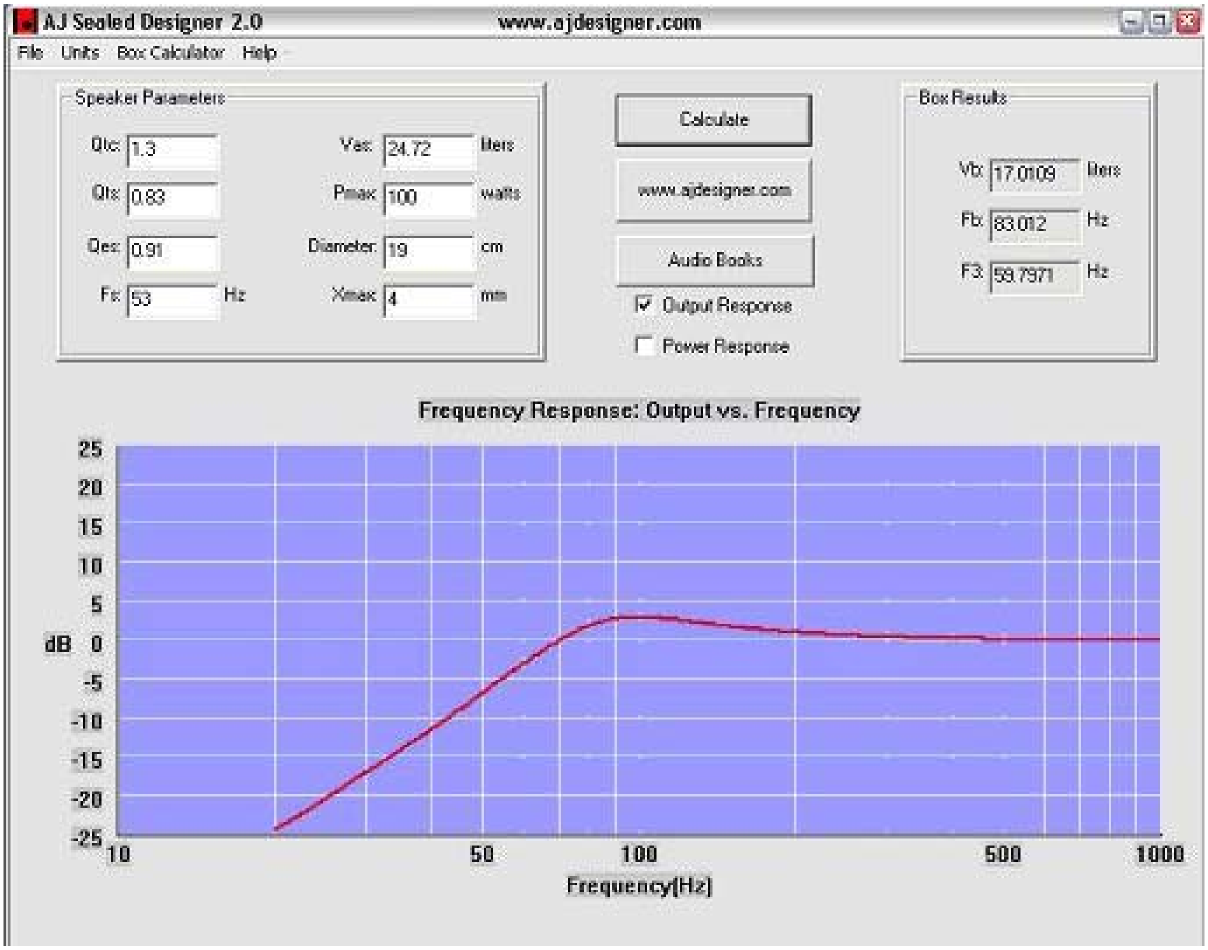

Fig. 23: AJ Sealed Designer 2.0 screenshot for output versus frequency

Fig. 24: E.M.R. speaker and enclosure

Transformer

Transformers are used to change a particular AC voltage to one that is practical for a specific need. Transformers consist of two separate coils of wire and a steel core. As voltage is fed into the primary coil, the flow of current produces flux. The flux is conducted to the secondary coil through the steel core and produces an induced voltage. The relationship between the current, the voltage and the number of turns is $V_1/V_2 = N_1/N_2 = I_2/I_1$, where 1 and 2 refer to the primary and secondary coils of the transformer. The result is a decrease in signal to provide a particular AC voltage level.

The power requirement for this application is ±35V DC and the source of power available is 120V AC at 60Hz. In order to get the proper voltage, a 70VA/90VA centered-tapped transformer will be used to reduce the 120V AC signal to a 50V AC signal. The frequency of the signal remains constant. This voltage feeds the power supply which converts the AC signal into a DC signal. The result is a ±35V DC signal to be used for component power.

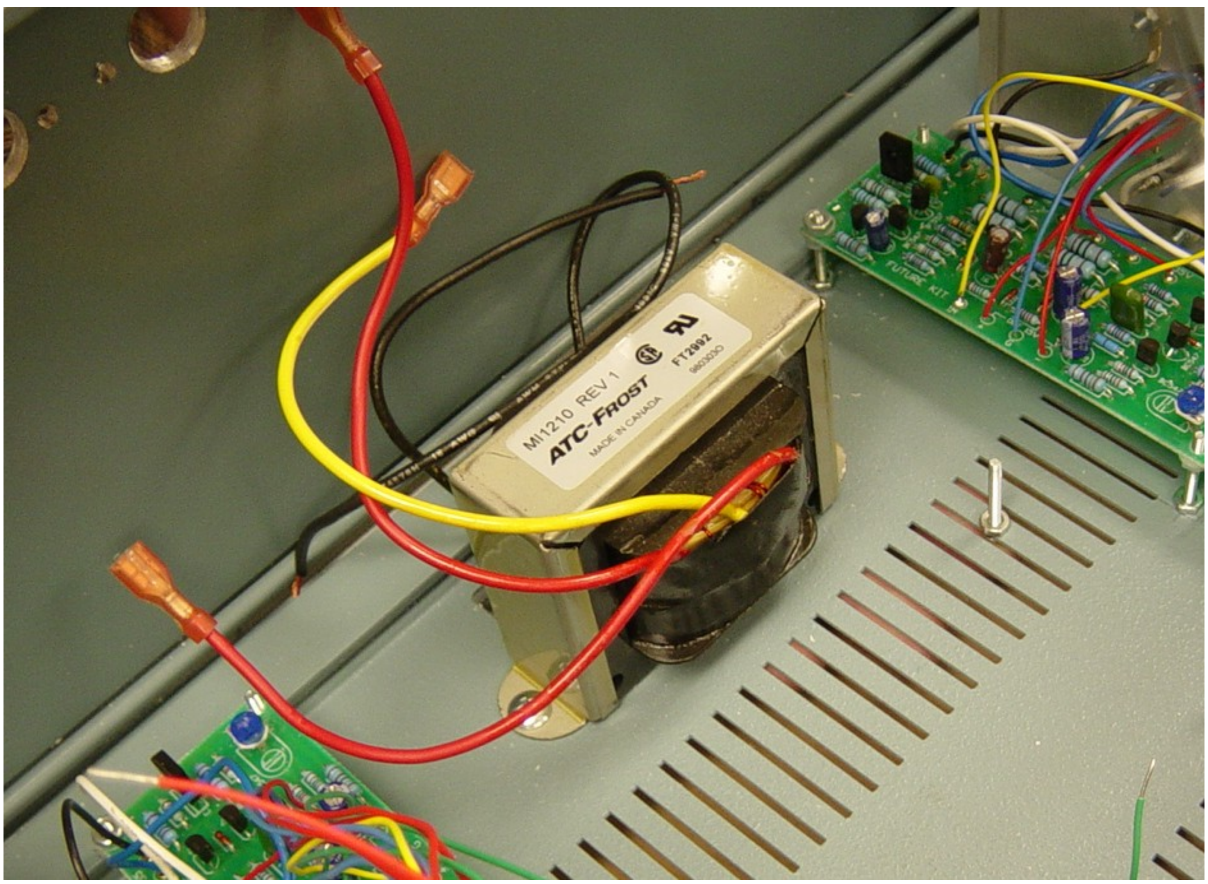

Fig. 25: E.M.R. transformer

Power supply

A power supply is used to convert voltage from an AC to a DC signal. The power supply consists of a bridge rectifier, which uses 4 diodes and 4 electrolytic capacitors. The diodes allow either the positive or the negative portions of the sine wave to pass current through. Once these sections have been separated, the capacitors provide DC ripple smoothing and the result is a DC signal. The power supply provides ±35V DC. This power supply is unregulated which means that the output voltage is dependent on the voltage that the transformer supplies when the rated current is drawn from it.

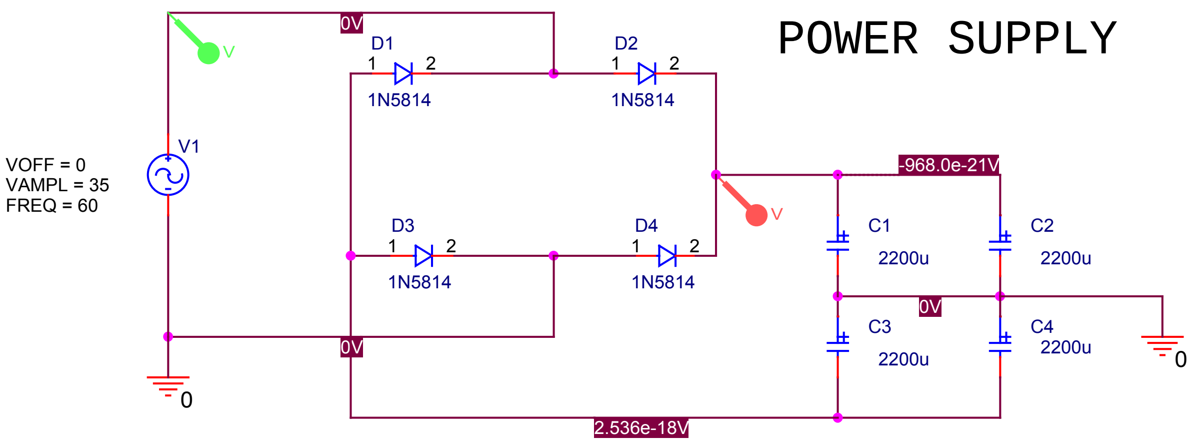

Fig. 26: E.M.R. power supply

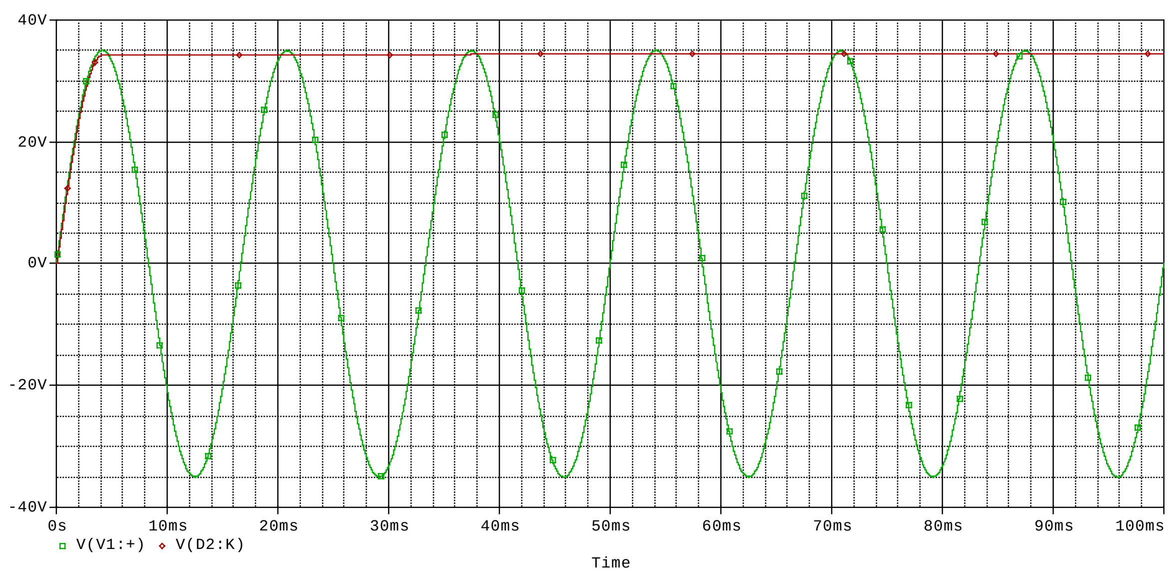

Fig. 27: AC to DC conversion

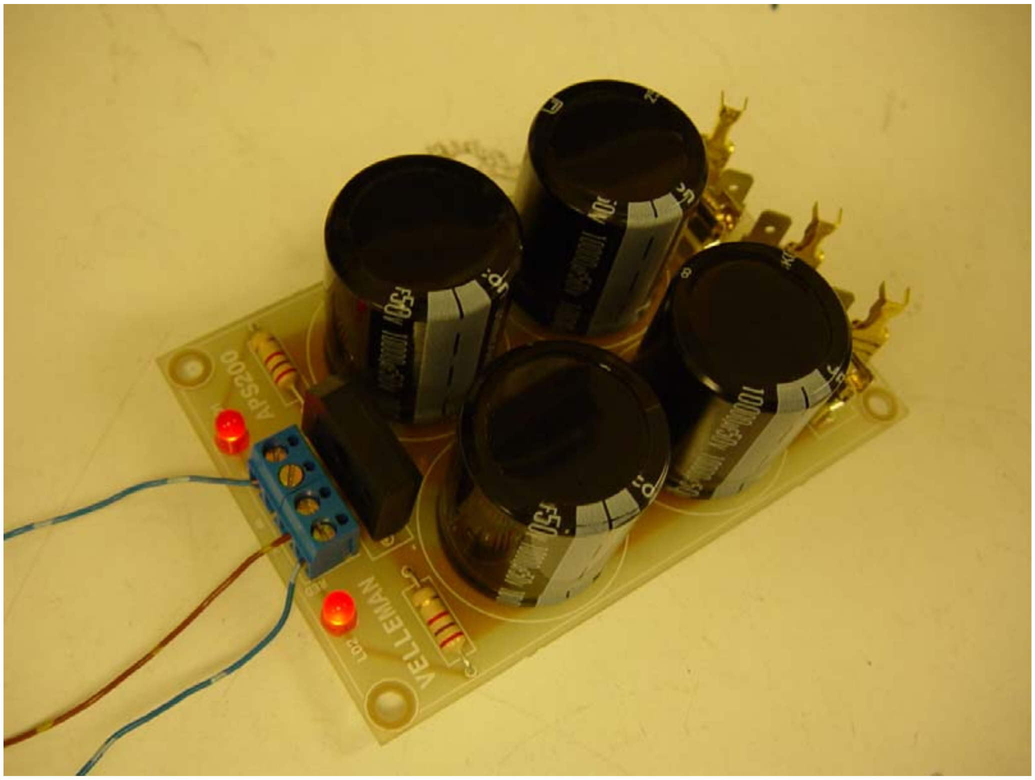

Fig. 28: E.M.R. power supply (lows)

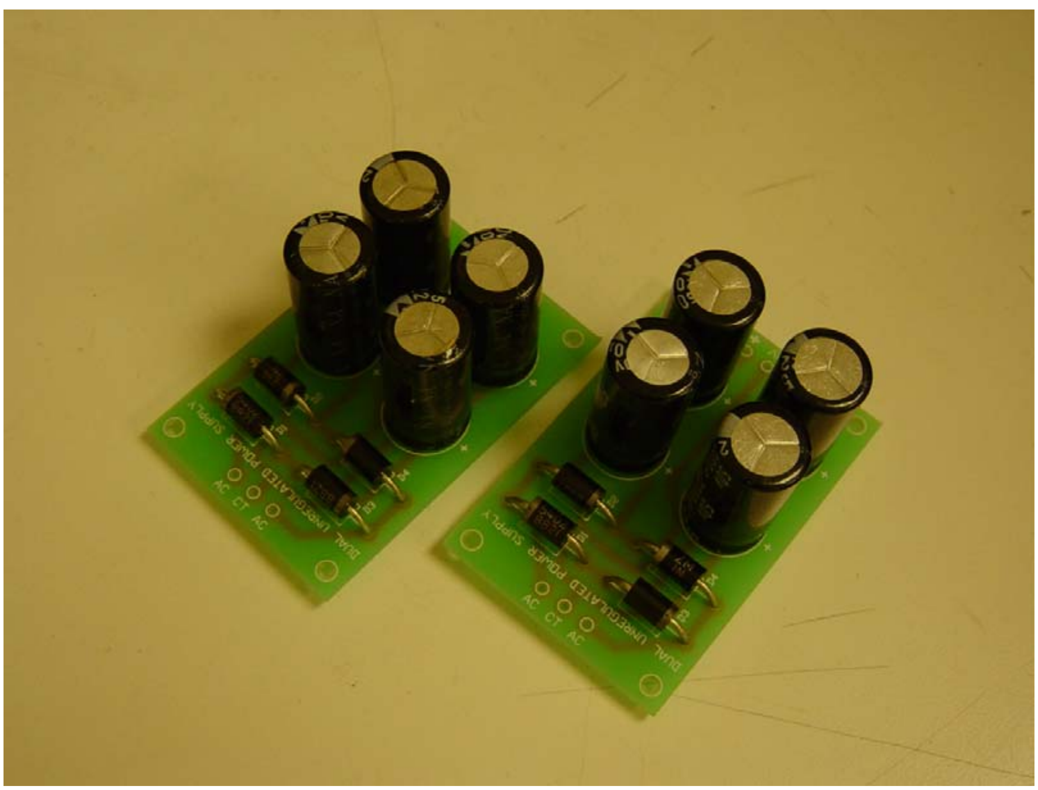

Fig. 29: E.M.R. power supply (mids and highs)

Debugging

The E.M.R. Stereo was built in stages by assembling each module separately. The preamplifiers posed no serious problems. The only technical difficultly that arose was finding a dual potentiometer to control the volume on both channels simultaneously. The equalizer was prepared on a generic PCB. Due to the design on the back of the board (the contact side), a few of the solder joints were shorted to other joints which caused the equalizer to function improperly. These solder joints were easily identified and fixed in a matter of minutes. The crossover was outsourced so it did not present any problems. The PCB for the 3-way crossover was made so allow multiple configurations. Using each of the two boards as a mono 3-way crossover, the appropriate connections were made. Unfortunately, the mid-band produced no output on either board. The signal was traced back from the output to the input but no anomalies were noticed so a new mid-band was made on a separate board. The outsourcing of the power amplifiers produced error free soldering on the provided PCBs. The quality of the product was reasonably good and no errors were made while soldering components onto the boards. The only problem encountered with one of the power amplifiers was when a power supply voltage of ±47V was provided when the maximum allowed was ±35V. This caused a failure in the power BJTs. The speakers and enclosures presented no problems.

Expenses

The following table gives a rough estimate for the amount of money the group spent to build the E.M.R. Stereo. The costs are divided into sections to reflect the cost for each one of the stages. The total cost is the sum of printed circuit boards, breadboards, resistors, capacitors, transistors, heat sinks, speakers, transformers, power supplies, power cord, wood and other miscellaneous components.

STEREO SECTION	***COST***
PREAMPLIFIERS	$ 22
EQUALIZERS	$117
CROSSOVERS	$100
POWER AMPLIFIERS	$142
SPEAKERS	$109
TRANSFORMERS	$ 26
POWER SUPPLIES	$ 67
MISCELLANEOUS	$ 47
TOTAL	***$630***

Costs

Considering the amount of material necessary to build one E.M.R. Stereo, the group could sell 1 unit to the public at a price of $500 (2004) or 100000 units for a price of $400 (2004). The cost of 500000 units 5 years from now will be reduced to $365 per unit (2009).

Contacts

The E.M.R. Group is composed of Eric Etnyre (BSEE), Bryan McCord (BSEE) and Enrico Reineri (BSCE). The E.M.R. Stereo was designed, assembled and tested entirely by the components of the group during the Fall 2003 and Spring 2004 semesters at California State University Northridge (CSUN). The stereo was a senior design project for ECE492 and ECE493, both taught by Ichiro Hashimoto.
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Fig. 30: Eric Etnyre (BSEE)

Fig. 31: Bryan McCord (BSEE)

Fig. 32: Enrico Reineri (BSCE)