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Senior Project Final Report	2011	Matt Steadman



# LoopKore

# **Computer Engineering Senior Project Final Report**

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## 1. Introduction

LoopKore is an all-in-one electrical instrument looping station, with several key features. The LoopKore has been inspired by three devices that up until now have only existed in two separate units; a multi-effects unit, a looping station and a digital music player/recorder. Looping stations are very popular tools for many musicians today, and there is a broad array of looping stations currently available on the market. They are usually packaged like an effect pedal, and are by far one of the most expensive pedals that most musicians will purchase. The prices range from 200\$ to 1000\$ for consumer electronics and even up to 2000\$ for custom made devices. The prices are usually proportional to the capabilities such the length of recorded tracks, number of recorder tracks, and ease of use and the extent of user controls provided. The LoopKore digital core is designed for maximum time allowed for recording and the number of tracks using a microSD card. The LoopKore is also designed to allow players use flash drives to select and play songs and be able to select periods so they can loop over and over and jam along with.

Aside from all the digital capabilities, LoopKore provides a multi-effects unit equipped with connection for direct recording. The unit offers sustain-boosting compression, soft and hard clipping distortion, active six and passive three band equalization, pre-amplifier and volume control which are all fully bypass-able.

## 2. Overall Block Diagram

The device is separated into the digital and analog subsystems.

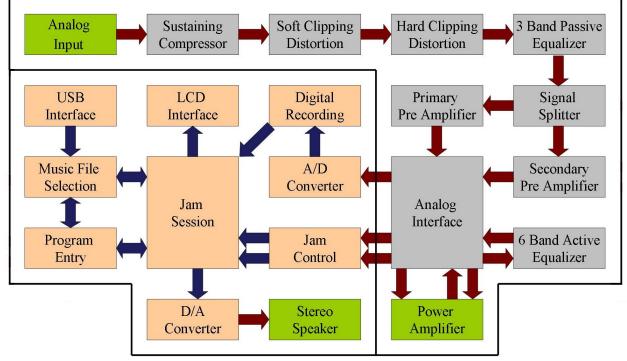


Figure 1: LoopKore Block Diagram

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## 3. Digital Audio System

At first glance when one considers the length of common songs even a 16- minute limit seems more than sufficient. Although there are very few common songs that are longer than 16-minutes when looping is involved this time frame becomes very important. Due to the large memory capacity of the LoopKore, the musician is able to fully explore sounds and music. In a one-man-band scenario loop stations are very useful because they enable the musician to play the parts of multiple instruments at different times but listen to the recordings at the same time, with the 200 minute capacity of the LoopKore the one-man-band has the opportunity to be a one-man-orchestra.

The micro-controller interfaces with the musician's USB mass storage device, reading audio files from the device, and storing recording sessions back onto it. The LoopKore has an internal 2GB memory which is facilitated by a Micro-SD card. The musician Interface consists of an LCD diplsay screen and LED's to get information to the musician, and the touch functions of the LCD screen and other on board buttons to get information from the musician. The digital audio data created by the LoopKore is sent through a DAC and this signal is available for the musician to plug in any compatible speaker that they wish.

The LoopKore has a wave file format playback and looping system at its core. This allows the user to select a song from a flash drive and hear it. The user also has the ability to select a start and stop point while the song is being played and loop the selected period for practicing and playing along to a certain period indefinitely.

#### **3.1. User Interface**

The user interfaces with the digital portion of the LoopKore through an LCD screen and several touch sensitive buttons on the AVR EVK1104 project board and a pedal button. The user can navigate through three different screens: the splash screen, select file screen, and the jam session screen. Each screen has 2 main sections information section and navigation section. The jam session screen has a third section at the top for displaying status. The look and feel of the screens is not a functionally significant part, and as such have been left to be worked on last. As further time allows improvements to the user interface will be made while ensuring performance is still maintained.

#### 3.1.1. Splash Screen

The splash screen is a simple screen which displays "LoopKore" in the information section and two options in the navigation section: select file, and play.

#### 3.1.2. Select File Screen

The select file screen displays all the files in the root directory on an attached USB drive or an inserted SD card in the information section. The User can then use the up/down touch sensitive buttons on the wheel of the board to navigate through the file list. In the navigation section the user has the option to go back to the splash screen or go to the jam session screen.

Extracting the file name turned out to be more difficult than what time allowed for this

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portion of the project. Also the functions to draw to the LCD screen are limited, so a quick function to display hexadecimal versions of integer values is used to display values. This is what is used to list the files as they are encountered on the drive starting with 0x00.

#### 3.1.3. Jam Session Screen

On the jam session screen in the status section along the top indicates to the user what the board is currently doing: playing the selected audio file, recording what is currently playing along with the user's audio input, or looping a section of the selected audio file. In the information section, as defined in section 3.1, displays useful information about the current operation of the LoopKore. The navigation section will display how the user can select which section they want to loop with a touch sensitive button on the board, go to the select screen, or go to the splash screen. To start and stop playback and recording the user will use pedals for input.

#### **3.2.** Playback

When powered on the board will set the first file on an attached USB drive or SD card as the selected file. While on the jam session screen the user will press the start playback button. This starts the selected file to play back through the audio output of the board. For this to happen several steps occur.

The selected file is stored as an index into the file list on the drive. First a handle into the selected file is extracted. Then the bytes of the file are read. Once a small buffer has been filled with data from the file, the buffer is sent to the DAC.

After the user has recorded some analog input, he can then switch to playback what is in the input buffer. He does this by pausing playback, then pressing the 'up' touch sensitive button on the board. This switches from reading data from the file to reading data from the save buffer. Then when playback is resumed it will push data from the save buffer instead of from the file.

#### **3.2.1. Supported Formats**

Initially the main file format planned to support was mp3. As we have run into significant time constraints as other functionally necessary tasks have taken more time than anticipated we have changed to only support wav file formats. The user will have to ensure files on the drive are all wav format. The program does not check for file type and will play any file as an audio file.

#### 3.3. Looping

To start looping the user will first start playback of the selected file. This defaults to loop the entire file. When playback reaches the desired start position, the user pauses playback with the foot pedal, then pushes the touch sensitive button on the board once to select the start loop position in the file. The user starts playback again. When playback reaches the desired end position he again pauses playback, then presses the same touch sensitive button to select the end position of the loop. The user then resumes playback, and the file is then traversed back to the start position. When playback reaches the end position the pointer into the file is then switched back to the start position. This continues until the user pauses playback, presses the button again for a third time which cancels looping. Then playback will continue from the current position and loop the entire file.

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## 4. Analog Audio Effects

The LoopKore has an analog signal processor with multiple audio effects units that are interfaced through several switches, knobs and jacks. The effects were chosen to give the player enough control over the tone by allowing the manipulation of the signal shape, dynamics and filtering. Each unit is powered separately to prevent excessive current draw from the power supply. Each unit also incorporates a true physical bypass switching system in its circuit. This bypass system allows the player to physically reroute the signal in a way that the respective effect's circuit could not affect the signal in anyway when the unit is turned off. Each unit also have an LED to indicate whether the input signal is bypassed thorough that unit or not. The resistors are film, mostly rated at <sup>1</sup>/<sub>2</sub> Watts, and range from 50 to 100 volts. Capacitors are ceramic for small values and film or electrolytic for larger values and all are rated at 100 volts. Due the high cost of audio potentiometers only miniature trimmers are used for all adjustable parameters of every unit as they are also smaller in size. These trimmers come with a few drawbacks such as their contribution to noise, inconsistencies in the amount of turning and the resistance ratio. Operational amplifiers, bipolar junction transistors, and diodes are also used and will be explained in their respective subsequent sections. For a detailed list of components and devices used in this system please refer to appendix A.

#### 4.1. Dynamics

The signal generated by an instrument or a microphone almost always has uneven amplitude for different frequencies, or in other words some frequencies are louder than others. This unevenness causes the sound to be dull at times and to lose its "spatial" feeling and the "fullness" in the tone. This also affects further processing of the signal such as shaping and modulation. A compressor allows for leveling all frequencies to the same amplitude.

The player may also need control over the volume of the output signal in order to preamplify the external cabinet amplifier or even overdrive if cabinet pre-amp distortion is desired. A preamp/volume circuit is well suited for this purpose.

#### 4.1.1. Compression

A compressor similar to the MXR DynaComp [1] is implemented with a few different modifications. This unit is placed first in the signal chain to affect the raw output of the instrument before any other processing is done on the input signal. This particular compressor uses a CA3080A operational trans-conductance amplifier or OTA [2]. This operational amplifier allows for biasing the transconductance to control the flow of current from the amplifier and therefore giving the player control over how long each note should sustain. The compressor is modified to allow adjustments to the attack of the compression. Attack is defined as how fast should the compression of signals take place. Fast attack may take away the instrument's signal's "initial burst" and therefore making it sound dull. Slow Attack takes away from proper compression and its emphasis. The unit also allows little distortion added through trimming by clipping the signal and rounding its peak edges. A control for level is also added to the circuit. For the schematic used to build this compressor please refer to figure 11 in appendix B.

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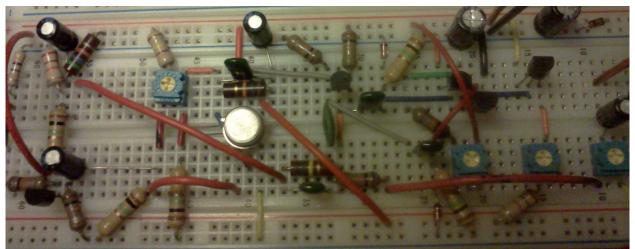


Figure 2: The Compression Circuit

#### **4.1.2. Pre-Amplification**

This unit acts mainly as a volume control for the final analog output being sent to an external cabinet amplifier. It incorporates a two stage operational amplifier circuit starting with a voltage follower and then an amplification stage. There is also a feedback loop to prevent any signal loss. In the case of low control potentiometer settings there could be little or no amplification. This usually leads to attenuation of the input signal for some frequencies. This unit also allows for further over-driving the external cabinet amplifier if some preamp distortion is desired without having to turn up every other unit's level controls. For the schematic used to implement this pre-amplifier please refer to figure 10 in appendix B.

#### 4.2. Shaping

One of the most popular signal shaping devices used by Rock and Heavy Metal players is a distortion circuit. Shaping and distorted a signal could be done through hard clipping, soft clipping and trimming. Hard clipping tends to be very edgy and sharp whereas soft clipping and trimming tend to sound fuller and warmer. The advantage of having a compressed signal before being distorted is the even clipping at all frequencies which results in a fuller and more robust sound.

#### 4.2.1 Distortion

The LoopKore uses a distorter that has two independent clipping stages. The first stage emulates soft clipping by actively filtering and clipping the signal through the negative feedback loop using a NE5532P[3], which is a low noise/high gain operational amplifier. This is done using red LEDs which happen to sound closer to a tube amplifier. Essentially the edges of the clipped signal at the peaks are rounded. The gain of this clipping amplifier is controlled via a potentiometer connected to the negative feedback loop.

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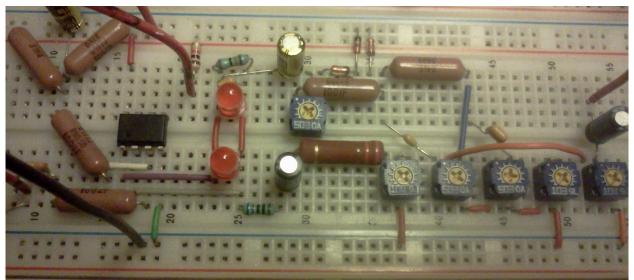


Figure 3: The Distortion Circuit

The second stage passively filters the signal and passes it through a loop of silicon diodes in both the positive and negative direction. This allows for hard clipping if an edgy and harsh sound is desired as the diodes will clip certain frequencies with sharp edges at the peaks. The amount of blending of the signal that is passed through this stage is controlled by a potentiometer. For the schematic used to build this distortion circuit please refer to figure 9 in appendix B.

#### 4.3. Filtering

Most of the time players need control over the loudness of a range of frequencies in order to compensate for the shape of the room, cabinet amplifier characteristics and more control over the effects system itself. Sometimes the player may want to sound very groovy and heavy, and sometimes rather softer and rocky. Active and passive signal filtering devices such as equalizers are used to offer such controls.

#### 4.3.1. Equalization

The LoopKore has a six band active equalizer based on MXR's M109 [4] allowing to cut or boost +/- 18db in 100Hz, 200Hz, 400Hz, 800Hz, 1.6kHz, and 3.2kHz frequencies . This unit uses LM358N operational amplifiers for buffering and biasing the input, NTE778A operational amplifiers as replacements for TL074 [5] for the first five bands and a passive high pass filter for the 3.2kHz band. Please refer to figure 13 in appendix B for the detailed schematic used to implement this equalizer.

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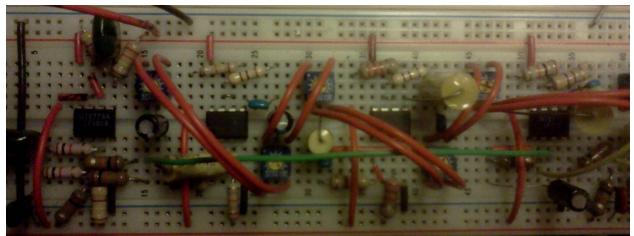


Figure 4: The Equalization Circuit

A three band passive filter for fixed ranges of low, mid, and high frequencies is also used to control the direct output of the clipping stages. The signal first goes through a low-pass filter, then a high-pass filter and their level potentiometers, and then the mid-band gain control.

#### 4.4. Routing

The input signal is received through a <sup>1</sup>/<sub>4</sub>" jack in an on/off configuration. This configuration allows for a floating ground on the power supply input plug. If no cables are inserted into the input and output jacks, then the common reference floats and the units will not power on. These units are connected together via bypass switches in a certain order. This ordering is particularly important as certain amplification or filtering stages need to be done before or after other stages. The compressor comes first in the chain followed by the distortion stages and the passive three band equalizer. The signal is to be compressed first so the clipping stages distort an even signal. At this point the signal is passed through a splitter circuit to provide the player with a compensated dual output. Each of the two split lines are then passed through separate pre-amplifiers and then to their respective <sup>1</sup>/<sub>4</sub>" jacks. If the split line is intended for recording, then a patch cable provides the connection of the split line output to the LoopKore's micro-controller board interface's <sup>1</sup>/<sub>4</sub>" jack input.

The equalizer is not routed together with the rest of the units. This is due to the fact that the equalizer is usually not used early in an effects chain and in this case would be placed after the pre-amplifier unit. But equalizers are usually put into the external cabinet amplifier's effects loop since they allow more control over the final stages of amplification rather than the pre-amp stages. Therefore the equalizer has its own dedicated input and output <sup>1</sup>/4" jacks. This allows for proper connection to an external cabinet amplifier's effects loop or it could be connected via a patch cable to the rest of the effects units.

#### 4.4.1. Splitter

In order to be able to route the signal to another device other than the external cabinet amplifier such as the LoopKore's micro-controller or just to have dual outputs for two amplifiers a splitter circuit is used. This circuit is based on ideas found on online guitar circuits and ideas

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[6]. This particular circuit uses a dual rail power supply to allow enough headroom for the splitting of the signal with less loss. After being split, the signal is passed through two voltage followers using LF353N operational amplifiers to maintain the amount of current for each line.

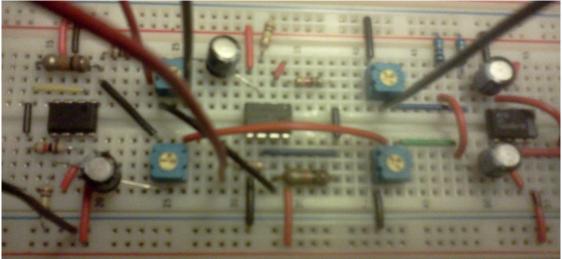


Figure 5: The main line volume (left), split line volume (center), and the splitter circuits (right)

#### 4.4.2. Volume

Both the main line and the split line are equipped with their own pre-amplifier circuits to allow full control over the volume on each line as the outputs may go to devices with different amplification capabilities. The only difference is that the main line goes through a bypass switch for its pre-amplifier but the split line is directly connected to its pre-amplifier.

#### 4.5. The Analog Interface

The connection between the digital and analog sides of the LoopKore is made via a front panel shown in figure 6. The effects connect to instrument inputs and external amplifier outputs through this panel. The effects system is also able to connect to the micro-controller board via its dedicated jacks and the ADC input control.



Figure 6: The analog interface panel

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#### 4.5.1. Effects System Interface

This effects system uses five TPDT switches, one for each of the individual units. These switches allow for a true bypass of each unit. Both the input and the output of each unit occupy one of the poles of each switch and the third pole is used for bypass indication on an LED. Each unit also has its own red LED to indicate whether the signal is passing through that unit or not. There are two inputs to this system used for an instrument, microphone, or the send line of an effects chain of an external cabinet amplifier. There are also three outputs used for any amplifying device's line-in, analog to digital converter, or the return line of an external cabinet amplifier. All inputs and outputs are connected via <sup>1</sup>/<sub>4</sub>" jacks with floating ground switch in mono orientation.

#### 4.5.2. Micro Controller Interface

There are connections available to control the micro-controller board with two active low on/off switches. The interface is setup to connect a Marshall MG series switch board to control the selection of "play" and "record", and an input jack to the micro-controller via the circuit shown in figure 7. The circuit provides connection to the jacks and the micro-controller with green LEDs indicating the active low behavior of the switches. The "press" action of the switches is "de-bounced" using electrolytic capacitors. The circuit is also setup to receive power and ground connections from the micro-controller board.

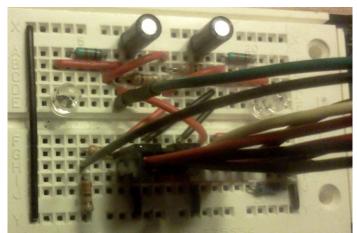


Figure 7: The micro-controller interface circuit

#### 4.6. Power Distribution

When multiple units are being powered up to amplify or filter audio signals, noise becomes a prevalent problem. In order to prevent the addition of supply noise due to high current draw, separate power supplies for each of the units are desired. LoopKore is equipped with a Gator BUS-G 9V [7] power supply. This particular provides several 9V and 18V isolated outputs that are well suited for power each of the five units. The compressor, distorter, and the equalizer each have their own dedicated lines but the splitter and both of the pre-amplifiers have to share the forth line. Since this power supply does not provide isolated grounds for each output, therefore another supply with a single output is used to provide -9V to the splitter and both of the pre-amplifiers. The power supplies are connected to the unit via five 2.1mm power jacks.

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## 5. Analog Recording System

Analog input is recorded by passing the analog signal through the internal 8-bit ADC and storing the output value in sequential locations in memory. The recordings can be stored on removable memory storage devices including USB flash drives and SD memory cards.

#### **5.1.** Conversion from Analog to Digital

In order to capture the analog audio signal into a digital format so that it can be stored and manipulated in the digital portion of the unit, it must first pass through our biasing circuit. This circuit was one that we had to design completely from scratch because a circuit of this nature depends directly on the ADC in question and the range of voltages that it is anticipating, and we couldn't find a circuit for using our ADC with audio signals.

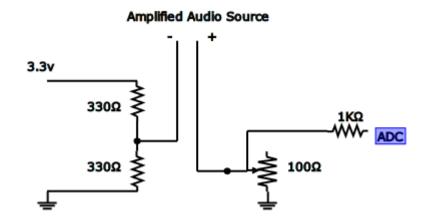


Figure 8: ADC and audio source connections

In the figure above, we see the schematic for the circuit used to force the incoming amplified audio signal from the other analog components of the Loopkore to be in the appropriate range expected by the ADC. The board is used to power the circuit and the left side of the circuit creates a virtual ground for the amplified audio signal, placing the ground of the analog audio circuitry to the midpoint between the maximum and minimum voltage level anticipated by the ADC. The right hand of the circuit includes a potentiometer that is used to control the amplification of the source. With the amplification level of the Loopkore this is achieved by setting the potentiometer near  $50\Omega$ , this needs to be set higher or lower depending on the actual source used with the circuit. The remaining  $1K\Omega$  resistor is used as a current limiting resistor, use to protect the ADC on the board.

#### 5.2. Storage

To record what is being played back the user taps a pedal button to put the system in recording mode. When the LoopKore is in recording mode, there is an interrupt enabled that happens at the standard audio frequency of 44.1 KHz. In the handler of this interrupt we place the most recent measurement made by the ADC into the next segment of an array buffer, then start the ADC conversion process over again. A few global counters and variables are updated as well that can be used in other places in the system. The ADC on the board requires 13 clock

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cycles to complete a single Analog to Digital conversion. And the conversion is done without robbing the processor of performing other tasks at the same time.

#### 5.3. Playback

Once values have been stored, they can be sent to the DAC unit which is connected to an audio amplifier on the board. The EVK1104 board has several built-in functions that are used at this point to start the audio playback. By informing the system of the frequency at which samples were recorded and the location in memory where the data is stored with these function calls, we can then start and stop the playback at will. This play functionality is connected to the second pedal button. When playback is active the contents of the currently selected file will be played through the DAC to the amplifier and out to any compatible speaker.

### 6. Conclusion

The software portion of the LoopKore worked for selecting a file off of a USB drive or SD cards. I would then get the data out of the file. Looping through a file was also achieved. Saving input data from the buffer into a new file also worked. We were also able to get samples off of the ADC. However, due to the way the board inputs are connected and how the board processes inputs we ran into a conflict between the interrupt mechanism we set up to poll the ADC and the touch sensitive buttons on the board. We were not able to resolve this problem before demo day. We also were not able to figure out how to configure the DAC on the board to properly playback audio data.

One of the main problems with programming the EVK1104 is that we could not find good documentation on how to configure components on the board. We basically had to look through the example projects provided with the board and guess on how to set up the functionality in our project. We did find an online message board group, www.avrfreaks.com, where we could get a few tips and ideas. However specifics on how to set thing up are limited to shared projects with little documentation. What we learned from this microprocessor project board is that when choosing a project board find all the documentation provided with the board. Look into what is provided and if good examples of how to do simple things are provided with explanations on what and why each step is done. We spent way too much time trying to figure out simple functionalities simply because there was a lack of documentation on how to do them.

The hardware portion of the LoopKore was implemented with mostly positive results. It is obvious that the device needs to be enclosed in a metal box and the circuitry needs to be implemented on a PCB to minimize noise and RF interference. Adjustments for sound quality were done through trial and error with some modification to the circuits and different component values. Overall the effects unit provides clean and reliable sound for further amplification. The main problem is to find the components with desired ratings or even finding some components that are not stocked to commonly manufactured.

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# 8. Appendix A (Bill of Materials)

Materials	Interface	Volume	Splitter	Distorter	Compressor	Equalizer
R1	500Ω	100kΩ	1MΩ	10kΩ	$56k\Omega$	- 1kΩ
R2	10kΩ	1kΩ	1MΩ	220Ω	27kΩ	470kΩ
<b>R3</b>	500Ω	10kΩ	1MΩ	1MΩ	1MΩ	330kΩ
R4	10kΩ	100Ω	$50k\Omega$ (Pot)	1MΩ	10kΩ	470kΩ
R5	_	$50k\Omega$ (Pot)	$50k\Omega$ (Pot)	1MΩ	4.7MΩ	1kΩ
<b>R6</b>	-	-	-	$680\Omega$	10kΩ	10kΩ
<b>R7</b>	-	-	-	220kΩ	1MΩ	10kΩ
<b>R8</b>	-	-	-	1kΩ	1 <b>M</b> Ω	10kΩ
<b>R9</b>	-	-	-	$22k\Omega$	470kΩ	$24k\Omega$
R10	-	-	-	$100k\Omega$ (Pot)	15kΩ	$24k\Omega$
R11	-	-	-	$50k\Omega$ (Pot)	150kΩ	470Ω
R12	-	-	-	$50k\Omega$ (Pot)	$27k\Omega$	$470\Omega$
R13	-	-	-	$50k\Omega$ (Pot)	10kΩ	$22k\Omega$
R14	-	-	-	$10k\Omega$ (Pot)	10kΩ	$22k\Omega$
R15	-	-	-	$100k\Omega$ (Pot)	10kΩ	470Ω
<b>R16</b>	-	-	-	-	1 <b>M</b> Ω	$470\Omega$
R17	-	-	-	-	1 <b>M</b> Ω	$20k\Omega$
<b>R18</b>	-	-	-	-	1kΩ	1kΩ
R19	-	-	-	-	$2k\Omega$ (Pot)	10kΩ
R20	-	-	-	-	$50k\Omega$ (Pot)	10kΩ
R21	-	-	-	-	$150k\Omega$ (Pot)	$470\Omega$
R22	-	-	-	-	$500k\Omega$ (Pot)	10kΩ
R23	-	-	-	-	-	$50k\Omega$ (Pot)
R24	-	-	-	-	-	$50k\Omega$ (Pot)
R25	-	-	-	-	-	$50k\Omega$ (Pot)
R26	-	-	-	-	-	$50k\Omega$ (Pot)
R27	-	-	-	-	-	$50k\Omega$ (Pot)
R28	-	-	-	-	-	$50k\Omega$ (Pot)
R29	-	-	-	-	-	$100k\Omega$ (Pot)

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Materials	Interface	Volume	Splitter	Distorter	Compressor	Equalizer
C1	100nF	10µF	- 47μF	1µF	ÎμF	100nF
C2	100nF	47µF	47µF	1nF	10nF	100nF
C3	-	47µF	220nF	2.2µF	1µF	10nF
C4	-	1µF	10µF	47pF	10nF	1µF
C5	-	-	10µF	1µF	1µF	220pF
C6	-	-	-	47nF	1nF	470nF
C7	-	-	-	1nF	10nF	120nF
<b>C8</b>	-	-	-	1µF	50nF	220nF
C9	-	-	-	-	10nF	68nF
C10	-	-	-	-	1µF	120nF
C11	-	-	_	-	10µF	33nF
C12	-	-	-	-	10µF	68nF
C13	-	-	-	-	-	15nF
C14	_	-	-	-	-	68nF
C15	_	-	-	_	-	10nF
C16	_	-	-	_	-	1µF
СС	-	-	-	_	-	1µF
D1	Green LED	Red LED	Red LED	Red LED	D1N914	Red LED
D2	Green LED	-	-	Red LED	D1N914	-
D3	-	-	-	D1N914	D1N914	-
D4	-	-	-	D1N914	Red LED	-
D5	-	-	-	D1N914	-	-
D6	-	-	-	D1N914	-	-
D7	-	-	-	Red LED	-	-
U1	-	TL072	LF353N	NE5532A	CA3080A	TL074
U2	-	TL072	LF353N	-	-	TL074
U3	-	-	-	-	-	MC1458P
U4	-	-	-	-	-	MC1458P
U5	-	-	-	-	-	MC1458P
U6	-	-	-	-	-	MC1458P
U7	-	-	-	-	-	TL074
U8	-	-	-	-	-	TL074
Q1	-	-	-	-	2N3904	-
Q2	-	-	-	-	2N3904	-
Q3	-	-	-	-	2N3904	-
Q4	-	-	-	-	2N3904	-
Q5	-	-	-	-	2N3904	-
SW1	SPST	TPDT	TPDT	TPDT	TPDT	TPDT
SW2	SPST	-	-	-	-	-

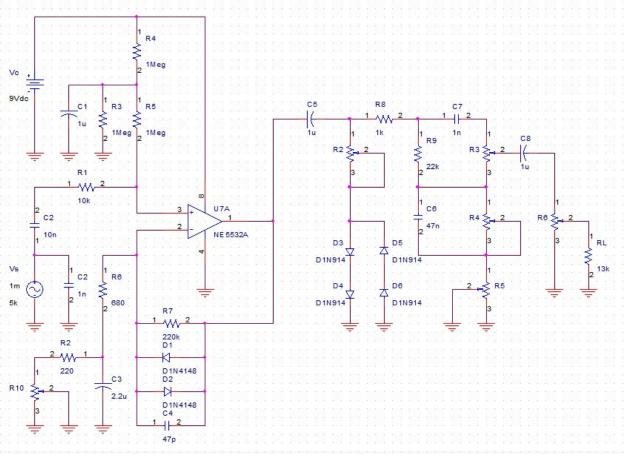
• Ateml EVK 1104 Micro-controller board

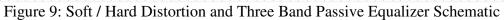
• Marshall MG Series Footswitch

• Gator BUS-G power supply

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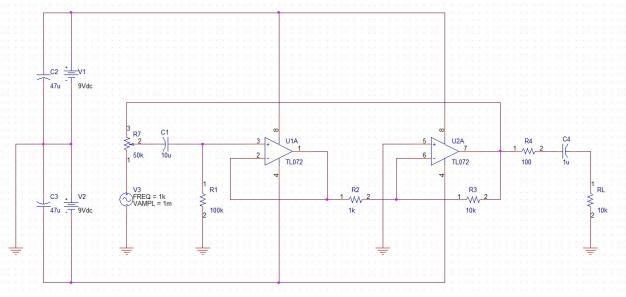


Figure 10: Pre Amplifier / Volume Schematic

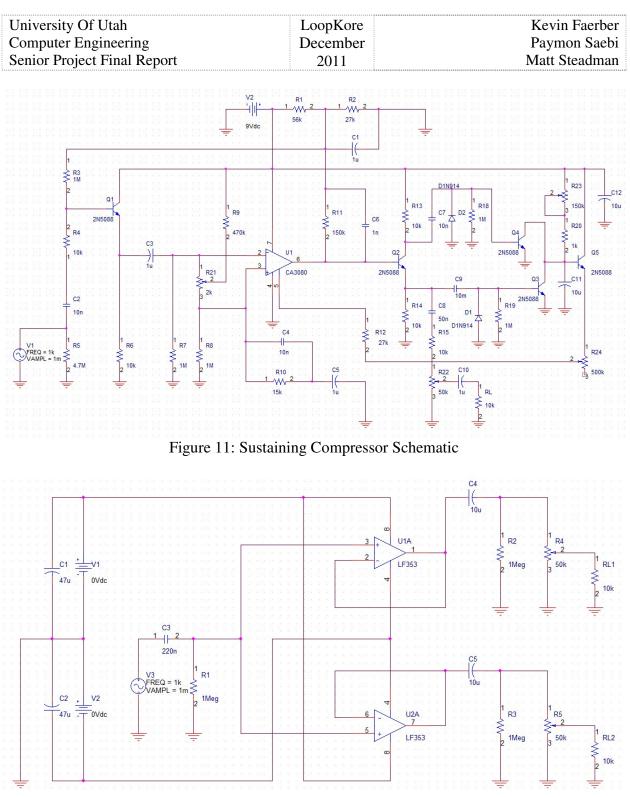


Figure 12: Signal Splitter Schematic

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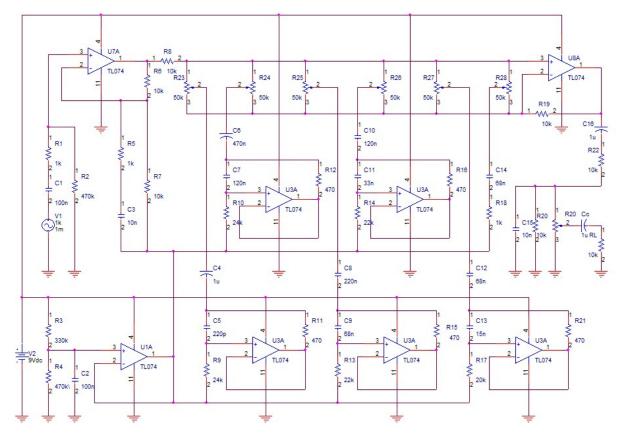


Figure 13: Six Band Active Equalizer Schematic

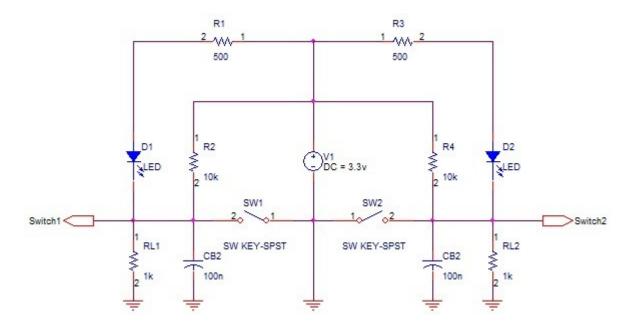


Figure 14: Micro-Controller Interface with two de-bounced active low Switches

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# Appendix C (Memorabilia)

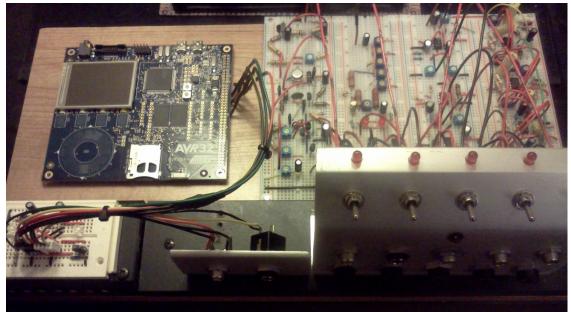
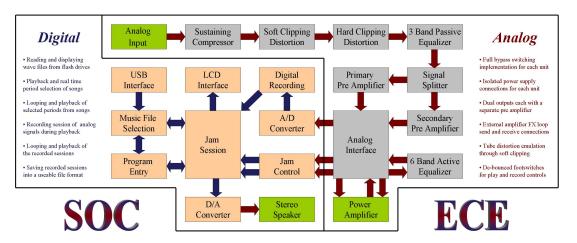


Figure 15: The Original LoopKore



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Figure 16: LoopKore Project Poster