ELEX 290

Camosun College
Audio Effects Processor

Prepared for

Godfried Pimlott
Joe Benge

Prepared by

Darryl Gamroth
Simon Tipler

March 10, 2001

March 16, 2001

Godfried Pimlott
Joe Benge
Camosun College

Dear Mr. Pimlott and Mr. Benge

I am enclosing our report *Camosun College Audio Effects Processor* as requested.

Sincerely

Simon Tipler
Enc

Executive Summary

The *Camosun College Audio Effects Processor* allows the User to easily add "effects" to their music or clean a noisy signal. With a simple User Interface and many effects with extreme customization features, this processor surpasses the competition.

Using a DSP processor, all mathematical calculations are very quick to render pristine sound quality. Also, the User Interface is ridiculously simple with a clear graphical display and a single Control Knob with push button. This knob controls all variables and navigation allowing the musician to make quick changes and get back to enjoying their music.

Table of Contents

1.0 CONCEPT 1
1.1 Understanding this Report 1
1.2 Background 1
2.0 DISCUSSION 2
2.1 Product Description 2
2.2 Hardware 2
2.2.1 ADSP-2181 EZ Kit Lite 2
2.2.2 PIC Microcontroller 5
2.2.3 Input / Output devices 6
2.2.3.1 LCD Display 6
2.2.3.2 Control Knob 7
2.2.3.3 Presets 8
2.2.4 Components 8
2.3 User Interface and Interaction 9
2.4 Effects 10
2.4.1 Passthru 10
2.4.2 Dirty Distortion 10
2.4.3 Cool Chorus 10
2.4.4 Rad Reverb 10
2.4.5 Phun Phasor 10
2.4.6 Funky Flange 10
2.4.7 Demon Delay 11
2.4.8 Power Pitch 11
2.4.9 Fierce Filter 12
2.5 Cost 13
3.0 CONCLUSION 13
4.0 GLOSSARY of TERMS 14
5.0 REFERENCES 16
6.0 APPENDICES 17
6.1 Schematic 17
6.2 Top Layer PCB 18
6.3 Bottom Layer PCB 19
6.4 User Interface Top Level State Machine 20
6.5 User Interface Controller Source Code 21
6.6 DSP Source Code 22

List of Figures

Figure 1: Physical Layout 2
Figure 2: Effects Building Blocks 3
Figure 3: Delay Effect 3
Figure 4: Reverb Effect 4
Figure 5: Chorus, Flange, Pitch, and Chord Effect 4
Figure 6: Modulator used for Vibrato and Distortion 4
Figure 7: Phasor using a variable frequency notch filter 5
Figure 8: Direction state machine for the Control Knob 7

List of Tables

Table 1: The scale 11

Camosun College Audio Effects Processor

1.0 CONCEPT

This report will introduce the reader to the *Camosun College Audio Effects Processor* including descriptions of the unit, input/output requirements, User Interface and controls, and the effects.

1.1 Understanding this Report

The simple menus system accesses all of the options for this system. To further understand the navigation system, read the *Camosun College Audio Effects Processor USER MANUAL*. When this report describes the use of each effect, we assume the reader is familiar with the basics of musical terminology. The Glossary of Terms includes definitions of most terms in this report.

1.2 Background

Guitar effect processors are commonplace in the music industry with nearly all guitarists using some type of guitar pedal to alter the sound from their guitar. Since the 60's, guitar pedals have improved with better sounds and more options for the musician. Leaders in this huge industry include BOSS, Digitech, Ibanez, and others. Currently, multi-effect pedals have hundreds of effects with fantastic customizability, but their price is beyond the amateur guitarist's budget.

That is not the only problem. What about people who play the flute, oboe, or saxophone? What about DJ's using turntables? There are many instruments, beyond guitar, that can use an effects processor. The User only requires only a microphone and preamp. The *Camosun College Audio Effects Processor* improves the sound of any instrument.

2.0 DISCUSSION

2.1 Product Description

This unit has stereo input and output channels, a Power Connector Port, simple controls, and many customizable audio effects enclosed in a beige case.

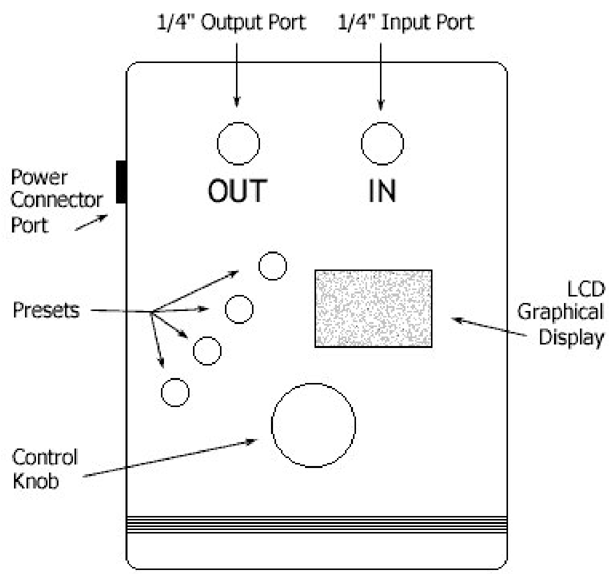


Figure 1: Physical Layout

2.2 Hardware

The entire Camosun College Audio Effects Processor consists of four parts:
1. ADSP-2181 EZ Lab demo board from Analog Devices
2. PIC16F877 Microcontroller from Microchip
3. Various User I/O devices
4. Control components

2.2.1 ADSP-2181 EZ Kit Lite

The ADSP-2181 EZ Kit Lite is a powerful inexpensive evaluation platform for the ADSP-2181 DSP. We chose this platform due to its on board AD1847 SoundPort CODEC. This is a full duplex 16 bit stereo CODEC capable of sampling at 48 kHz, ideally suited for our project. Unfortunately, this part is no longer in production; therefore, we could not produce a commercial product using this part. The ADSP-2181 is a 16 bit fixed point digital signal processor running at 33 MHz, it is adequate for our current incarnation of the effects processor but has several quirks.

All of the effects are processed in the time domain. As all effects are time based, they each share the same fundamental building blocks.   The base block for the majority of the effects is the delay line. The delay line puts an input sample into memory, which is recalled later to produce a delay. To create reverb we feed the output of the delay back into its input, which causes an echoing sound. This echo will continue forever if the gain is equal to one.

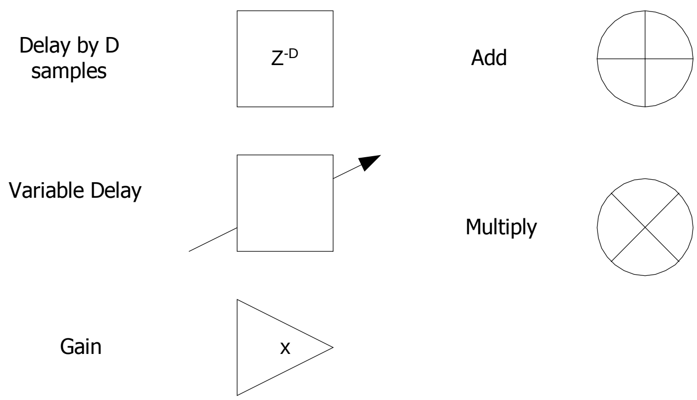


Figure 2: Effects Building Blocks

All effects are constructed using these building blocks as shown below:

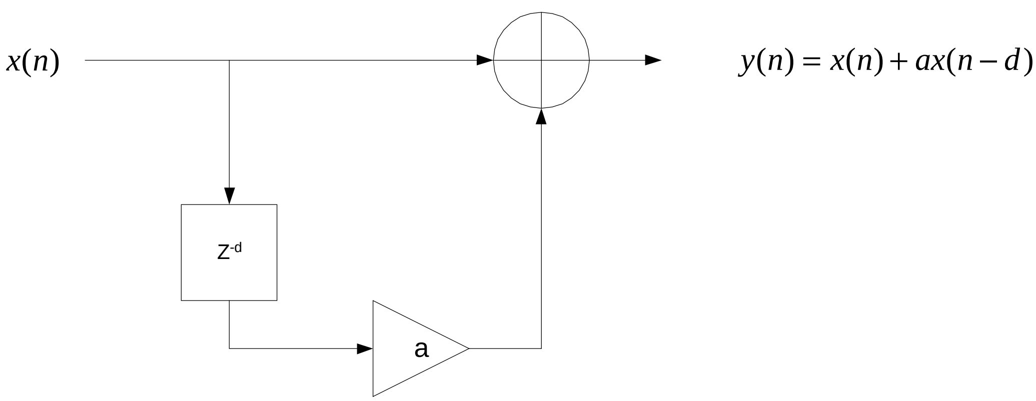
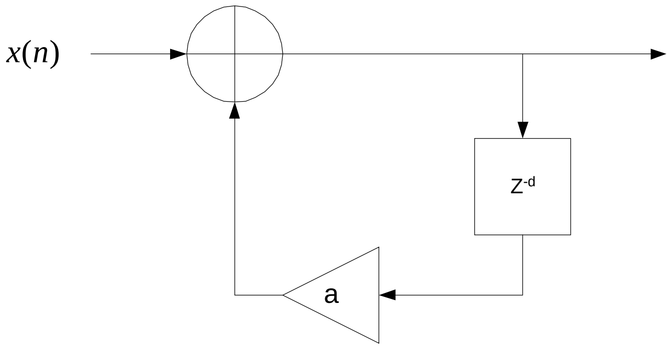
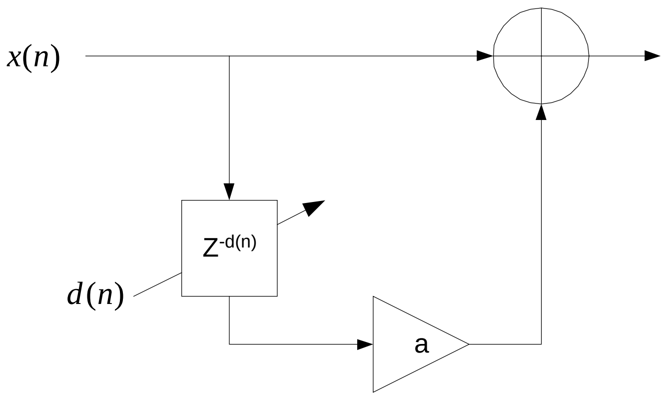


Figure 3: Delay Effect



$$y(n) = x(n) + ay(n-d)$$

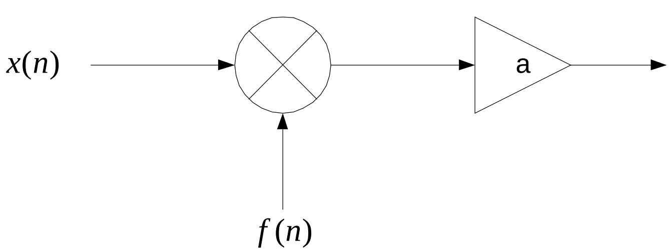
Figure 4: Reverb Effect



$$y(n) = x(n) + ax(n-d(n))$$

Figure 5: Chorus, Flange, Pitch, and Chord Effect

As shown in Figure 5: Chorus, Flange, Pitch, and Chord Effect, we can modify the function $d(n)$ as we see fit. In the chorus effect, the processor varies the delay from 20ms to 50ms at 0.25Hz. Flange is just a special case of chorus; delay varies from 0ms to a user specified amount and a selected frequency. Pitch scaling is a similar technique, except to drop and add samples to the playback buffer, we use saw tooth waveform.



$$y(n) = a \times x(n) \times f(n)$$

Figure 6: Modulator used for Vibrato and Distortion

Distortion is achieved by modulating the input signal with either a sinusoid or a saw tooth wave. The output signal can be saturated by adjusting the gain.

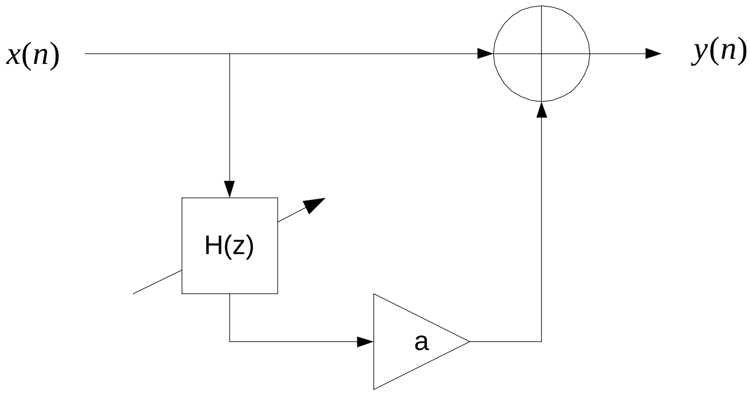


Figure 7: Phasor using a variable frequency notch filter

To implement phasor, the input signal is mixed with itself passed through a notch filter. This causes a phase shift in the output signal.

2.2.2 PIC Microcontroller

The PIC Microcontroller from Microchip controls all User input and transmits all changes in data to the ADSP EZ Lab demo board. The User Interface uses the PIC16F877 for its multiple serial communication ports, speed, many input/output ports, built in timers, and availability of optimized C compilers.

For this project, we use the Serial Peripheral Interface (SPI) to communicate with a digital potentiometer, the Analog Devices AD8400, discussed in the "Control Components" section.

The PIC also runs at the maximum speed of 20MHz. By running at this speed, the PIC can quickly compute calculations, rapidly transmit data, and control many devices. Since current consumption is not a variable in this project because of the required power adapter, this warrants running at a faster speed.

Since this PIC16F877 has 33 I/O pins, there are enough data lines for the parallel memory write, control, User input, and chip selection. Originally, there were not enough data lines for communication. To resolve this problem, the Toshiba TC74HC595 (SPI 8-bit shift register) could add eight more output lines. This device is incredibly useful because it only requires a chip select and the SPI data and clock lines, which the digital potentiometer already uses. After much rearrangement, and a change in a major component, this chip was no longer necessary.

Another purpose for the PIC Microcontroller is the "Interrupt on change" feature on PORT B for the Control Knob. We tested this feature; however, polling proved to be more effective. Using the "Interrupt on change" missed occasional step rotations, but polling never missed any changes.

To compose code for the PIC16F877, we used the PCM C Compiler. This compiler is extremely efficient, and it is simple to use for writing strings of data to a device. To fit large programs, this compiler spends most of the compiling process rearranging code to fill banks on the PIC. Most compilers for PIC do not rearrange segments for optimum banking. To make compiled code more efficient, the complier optimizes both delays and complicated math routines.

Another feature of the PIC processor family is the built in timers. These allow interrupts to occur determined by the programmer. The purpose for these timers in this project is to determine "timeouts" when waiting too long for a particular device to respond and to determine the rotation speed of the Control Knob.

2.2.3 Input / Output devices

The Input / Output devices consists of three groups:
1. LCD Display
2. Control Knob
3. Preset Buttons

Please refer to section 2.3, User Interface and Interaction, for more information regarding the use of the *Camosun College Audio Effects Processor*.

2.2.3.1 LCD Display

The LCD for this project is the Optrex DMF50834 with a built using the NEC upd16435 controller distributed by Apollo Displays (www.apollodisplays.com). This product is ideal for this project for its multiple options and fast speed. Some of the built in functions are:
- Reverse Line
- Magnification (Double width, double height, or both double width and double height)
- Blinking character
- Cursor
- Backlight

At the beginning of the project, the options were unknown; this made this LCD particularly advantageous. We considered using the backlight function, but abandoned it because of its extreme sensitivity to small changes in voltage. In addition, the LCD for this project was inexpensive because of a broken backlight.

To signify a selected item, we used the reverse line function. Our group felt this would unmistakably designate a selected item.

The Graphical Display shows the User the vertical menu system and all sub-menus. This system is unbelievably User-friendly to navigate and edit settings. A reversed line designates the currently selected item on the LCD.

Depending on the lighting in the environment, the User is able to digitally set the contrast in the Contrast menu item (see AD8400 in the Control Components).

2.2.3.2 Control Knob

To simplify all User-interaction, the knob with integral push button controls the entire system. We chose a 32 detent Grayhill 61C11-01-08-02. The detents provide the User feedback to acknowledge single steps. There are only three wires to communicate to the PIC controller; two wires for the direction of rotation and the other wire provides the signal for the push button.

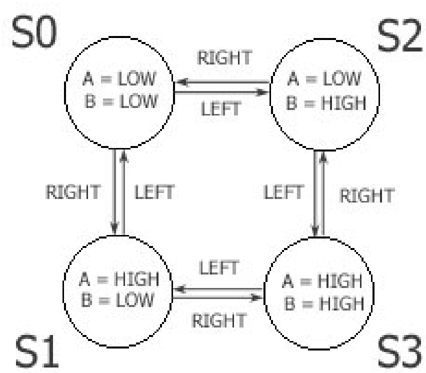


Figure 8: Direction state machine for the Control Knob

Since the PIC recognizes each step of the Control Knob, it is simple to determine the speed and direction of the rotation. Using one of the PIC's built-in timers, we determined that a change within 190ms defines "fast" rotation. Any changes that take longer than this define as "slow" or "single step" rotations. We call this process, "velocity-controlled stepping" where fast rotations result in bigger changes in setting a variable. Steps for a fast rotation are groups of ten. With a 32 step knob, a full byte change is 256/(10*32) = 0.8 of a full rotation. We chose groups of ten because this is the greatest change possible for one to easily rotate the knob.

The problem with velocity-controlled stepping is looping a variable around zero. Stepping from 255 to zero (and backwards) seems more efficient than blocking the User at the maximum value. However, if turning the knob quickly, the User may pass a desired maximum value. To stop this, we turn OFF velocity-controlled stepping when the User is 20 steps from minimum and maximum values. This way, the User can still loop around, but not make an undesired mistake.

2.2.3.3 Presets

To allocate the musician's favorite effects to particular presets we use four Preset Buttons. These presets are simple Normally Closed (N/C) contacts. The reason for these particular buttons is they are both inexpensive and esthetically pleasing (they already have red buttons).

2.2.4 Components

There are a few more parts of this processor:

- Power Connector Port
- Stereo input / output jacks
- AD8400 Digital Potentiometer
- 74HC138 Address Decoder
- K6T4008C1V 512kbyte SRAM

Power Connector Port
Since the processor consumes a substantial amount of current to provide ultimate sound clarity, it is not efficient to run this unit from batteries. Simply connect the included power adapter to the Power Connector Port, and you are ready to play!

Stereo input / output jacks
The audio input and output ports are ¼" stereo jacks. ¼" jacks are a standard in the music industry for their power handling abilities and cost of cable.

AD8400
The AD8400 is a single 256 step digital potentiometer used for the contrast of the Graphical LCD. This part communicates using the SPI port allowing extremely fast changes. This allows the User to digitally set the contrast of their screen in the "Contrast" sub-menu.

We researched other digital potentiometers; however, the Analog Devices series provides different ranges (32 step, 64 step, 128 step, and 256 step). For contrast, many steps are required to achieve optimum contrast. Also, SPI communication is extremely simple and fast for a PIC Microcontroller.

74HC138
The 74HC138 address decoder splits the DSP's 4MB addressable space into eight blocks of 512 kB. One decoder output selects the flash memory, and another selects the 512k of SRAM. We used this part rather than other address decoders simply because we used this part in studies at school.

K6T4008C1V
The K6T4008C1V from Samsung is low power, high speed SRAM. SRAM is required to provide a sufficient buffer for the digital data from the input signal. The reason for SRAM opposed to other high speed volatile storage is that our DSP requires SRAM.

2.3 User Interface and Interaction

The entire interfaces consists of three parts:
1. Graphical Display
2. Control Knob
3. Four Preset buttons

To enter a menu item, edit a number, exit number entry, etc., simply press the Control Knob into the unit. This way, speed of entry is extremely efficient, and the User does not have to search for multiple buttons.

To define a particular preset, simply select the effect in the Main Menu, and press the chosen Preset Button. Immediately this effect processes the input signal. To vary the sound, enter that particular menu item and change the variables. The effect updates the sound in "real time". This allows the musician to create very interesting sounds.

To overwrite a particular preset, simply select a new effect and press the same Preset Button. Right away, the new effect will be associated with that Preset Button, and the unit will immediately run the new effect.

For more information on User Interaction, please refer to the *Camosun College Audio Effects Processor USER MANUAL*.

2.4 Effects

2.4.1 Passthru

Simply outputs the input signal with absolutely no processing on the sound. This is called a "Dry Signal".

2.4.2 Dirty Distortion

By increasing the amplitude of the input signal, eventually the signal will "clip". Clipping is the process by which an AC signal increases past the stability point. The result is many high frequency components also known as "noise". The User has full control over the level of noise.

2.4.3 Cool Chorus

The Chorus effect allows the user to hear multiple instances of the input signal when each instance synchronizes with the others, except for small variations in their strength and timing. This means that one vocalist can sound up to three people singing the same thing. The User controls the number of "voices" heard with a maximum of three due to program space restrictions.

2.4.4 Rad Reverb

Reverb is simply a Comb Filter. This effect occurs when a sound wave bounces off walls of a listening space, but has an interesting effect when there are multiple reflections.

2.4.5 Phun Phasor

Phasing or phase shifting passes the signal through a narrow notch filter and combines a proportion of the filter's output with the direct sound. To create a weird effect, the centre frequency of the notch filter varies in a controlled manner. The User sets this variable in the *Phun Phasor* Sub-Menu.

2.4.6 Funky Flange

The process of periodically varying the delay with a low frequency (such as 1Hz) is the Flange effect. This product allows the User to set both the period and the frequency of this sound.

2.4.7 Demon Delay

Delay for our processor is really a tapped delay, which is really just a set of delays. Our system provides up to three separate delays of a signal at different gains. The User sets both delays and gains for the three "taps". This allows plenty of variables for the musician to customize their sound.

2.4.8 Power Pitch

The *Power Pitch* allows the User to "bend" the input signal in half step intervals up to one octave up or one octave down. An octave consists of 12 half step tones to create the "Equal Tempered Scale". The scale based on fifths proposed by Pythagoreas (600BC) is the basis of the Equal Tempered Scale.

Equal Tempered Scale (Chromatic Version)	Ratio	Interval Name	Just Interval
C	1.0000	Unison	1.0000
C#	1.0595	Half Step	1.0667
D	1.1225	Whole Step	1.1250
D#	1.1892	Minor Third	1.2000
E	1.2599	Major Third	1.2500
F	1.3348	Perfect Fourth	1.3333
F#	1.4142	Diminished Fifth	1.4063 or 1.4222
G	1.4983	Perfect Fifth	1.5000
G#	1.5874	Minor Sixth	1.6000
A	1.6818	Major Sixth	1.6667
A#	1.7818	Minor Seventh	1.8000
B	1.8877	Major Seventh	1.8750
C	2.0000	Octave	2.0000

Table 1: The scale

The 12-tone equal tempered scale is a natural scale for electronic music systems because of the simplicity of equal valued steps. Though nearly impossible to audibly notice any difference, the musician must acknowledge this is only an approximation of the true Just Major Scale.

2.4.9 Fierce Filter

To reduce high frequency noise from a particular input instrument, the *Fierce Filter* allows the user to remove high frequencies. The User is also able to set the cut-off frequency.

2.5 Cost

DSP Board	$132.00	Provided
PIC	$ 15.00	Provided
SRAM	$ 32.00	
Flash	$ 10.00	Free
PCB	$ 10.00	Provided
Connectors	$ 15.00	
LCD	$ 50.00	
Knob	$ 32.00	
Buttons	$ 10.00	
Case	$ 22.00	
Misc	$ 20.00	
Sub Total	$348.00	
Provided	($167.00)	
Total	**$181.00**	

We completed this project significantly under budget. This is due to most of the expensive components supplied by the College.

3.0 CONCLUSION

The *Camosun College Audio Effects Processor* provides the musician with several advantages over many other similar products:

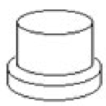
- Line input! Not only for guitar
- Extremely simple User Interface with clear LCD display and few buttons
- Future expandability

This chief disadvantage of this product is only in the number of effects. Future expansions of this product will allow for multiple inputs and more effects.

4.0 GLOSSARY of TERMS

HARDWARE

CODEC	Hardware that performs analog to digital and digital to analog conversion. Includes signal conditioning circuitry



Control Knob – large knob below the LCD. This is the main input device for the User.

Control Knob Button – asserts when the Control Knob is pushed into the unit

Control Knob Rotation – turning the Control Knob

Full Duplex	The ability to record and playback simultaneously
Input Device	Any audio source with a LINE output
LCD	The text display screen
Output Device	Any audio output device with a line input (ie Amplifier)
Power Connector Port	Requires minimum 7V at 200mA power adapter
Preset Buttons	Four red buttons to allocate particular effects to particular Presets

MENU SYSTEM

Main Menu	The vertical list of particular options and effects.
Sub-Menu	The options for a particular item in the Main Menu. Sub-Menus are only for items containing options or information.

AUDIO TERMINOLOGY

Clip, Clipping	overload, severe distortion

dB (decibel)	a unit of measurement, ratio of two voltages (dB = 20log(V1/V2))
Dry Signal	no processing of the input signal
Filter	device or program for adding or removing part of a frequency bandwidth
Line Signal	amplified signal within 100mV peak to peak produced by sound cards, turntables, etc.
Mic (Microphone) Signal	signal within 20mV peak to peak produced by guitars, microphones, etc.
Wet Signal	signal with effects added

5.0 REFERENCES

1. Analog Devices. *ADSP-2100 Family – EZ-KIT Lite Reference Manual*. Norwood, MA, 1995.

2. Analog Devices. *ADSP-2100 Family – EZ-KIT Lite Evaluation Platform Data Sheet*. Norwood, MA, 1998.

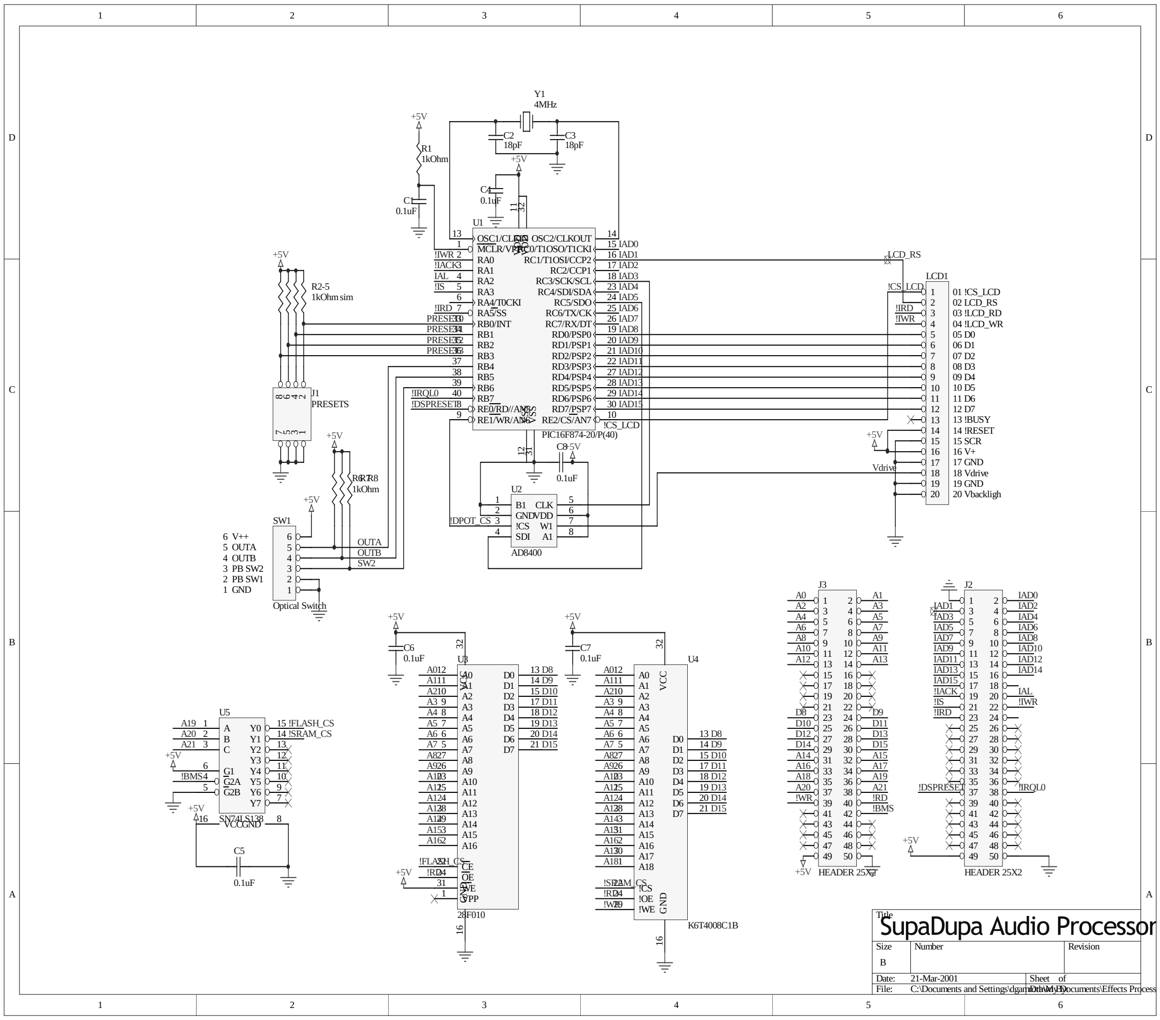
3. Behringer International. *DX500 Pro Mixer USER'S MANUAL*. Hanns-Martin-Schleyer-Straße. D-47877 Willich, Műnchnëide ||, September 1998.

4. Hutchins, Bernie. *Music for Electronic Engineers*. Electronotes. Ithaca, New York, July 1975.

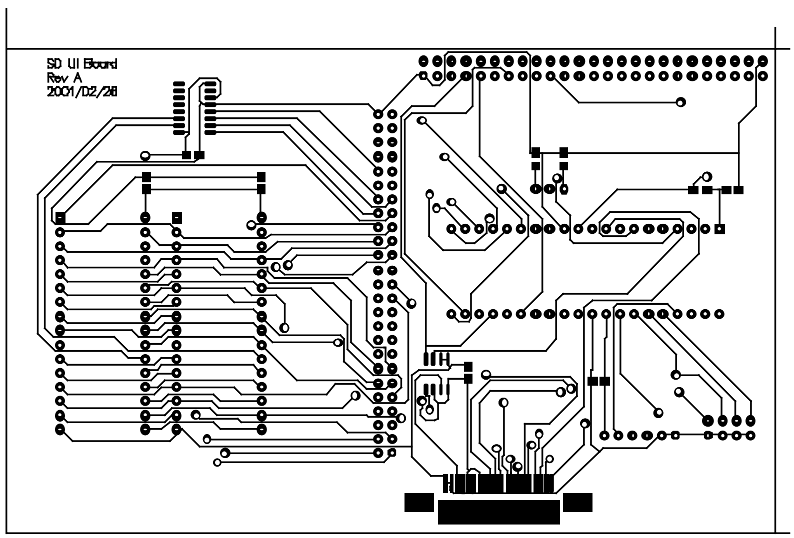
5. Orfanidis, Sophocles J. *Introduction to Signal Processing*. Prentice-Hall Signal Processing Series. Upper Saddle River, New Jersey, 1996

6.0 APPENDICES

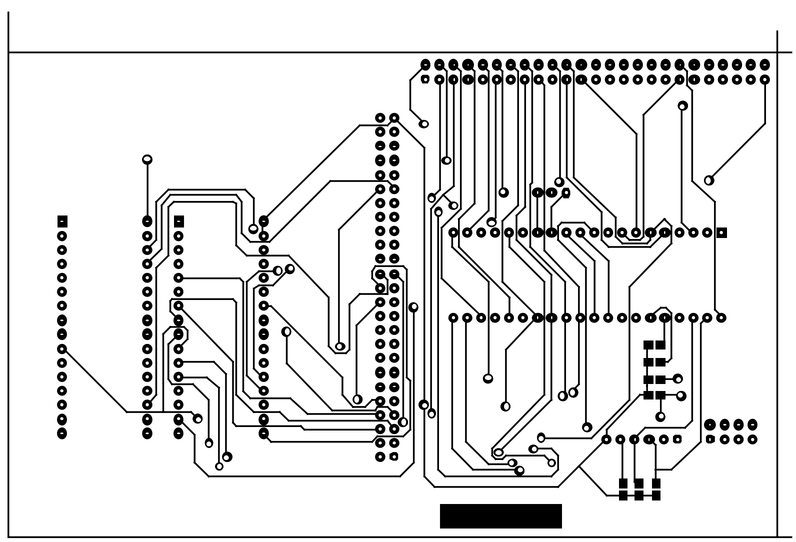
6.1 Schematic



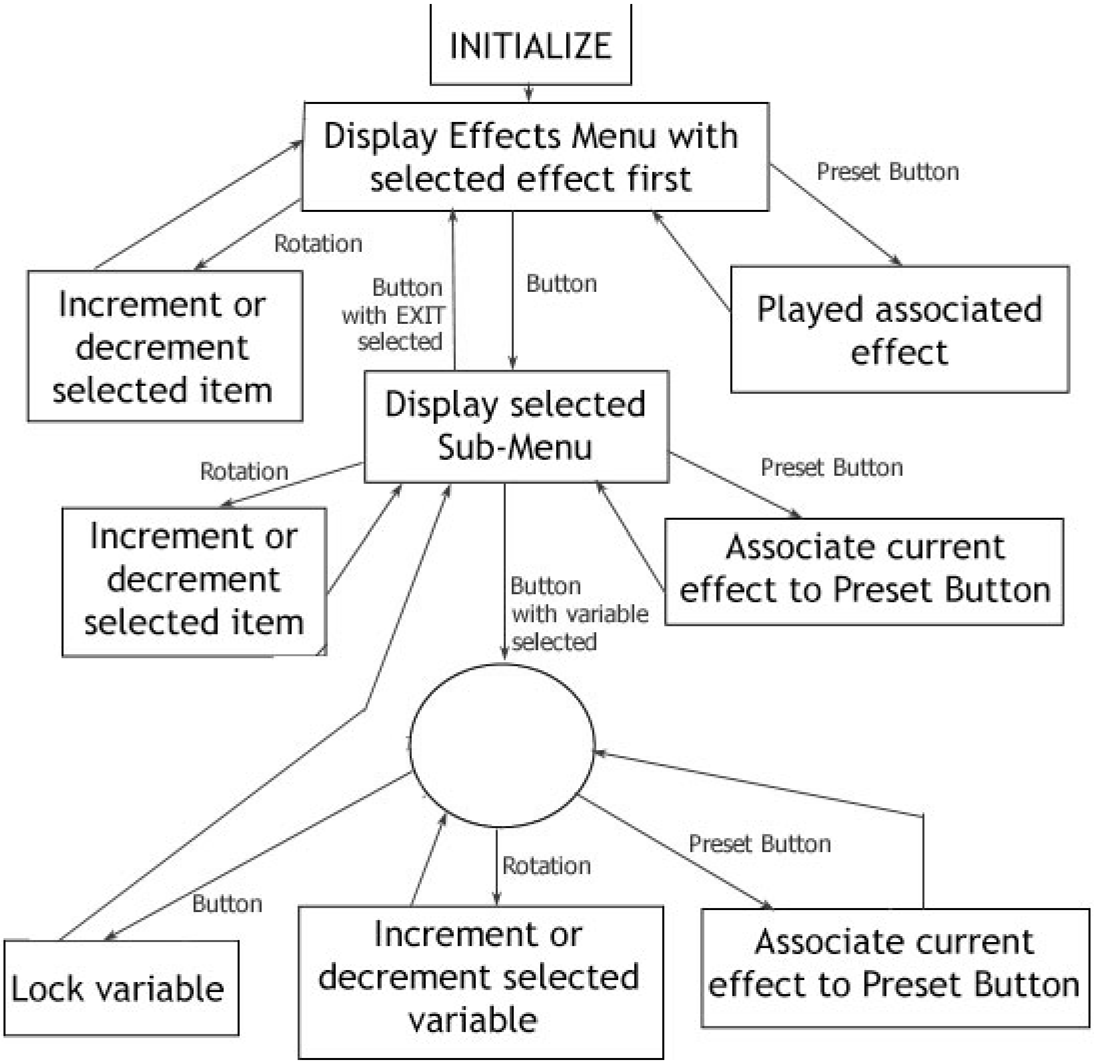
6.2 Top Layer PCB



6.3 Bottom Layer PCB



6.4 User Interface Top Level State Machine



6.5 User Interface Controller Source Code

See hard copy.

6.6 DSP Source Code

See hard copy.