DSP Musicolour Firmware Operation.

By Mauro Grassi Copyright 2008.

We explain the operation of the firmware in more detail as well as discussing possible accessories.

Overview of the Firmware

This article describes the features of version 1.00 of the DSP Musicolour firmware. It is possible that future versions of the firmware will have enhanced or modified features. The implemented features in this version can be seen in the features panel.

Implemented Main Features in versions 1.58 and 1.60:

1. Selectable Eight Band Equalizer
2. Auto detection of mains frequency (50Hz or 60Hz operation)
3. Selectable phase controlled or Zero Voltage switched output channels (8 bit resolution)
4. For each logical output channel:
    1. Selectable Gain;
    2. Selectable Audio Passband: arbitrary Minimum and Maximum Frequencies;
    3. Selectable Acquisition Mode: Peak or Average;
    4. Selectable Chaser Mode;
    5. Selectable Quiescent Level (Filament Preheat);
    6. Selectable ZV Mode, Strobe Mode, Direct Mode or Continuous Mode.
5. Selectable Logical channel for each physical output channel;
6. Chaser Modes implemented as a virtual machine;
7. Trigger Channel with:
    1. Selectable Audio Passband;
    2. Selectable Trigger Threshold;
8. Firmware support for RC5 remote control (requires additional PC board);
9. Firmware support for high speed UART (requires additional PC board);
10. Selectable Balance;
11. Selectable Display Frequency, Brightness and Screen Saver Time Out Period;
12. Selectable Sampling Frequency from 16kHz to 50kHz simultaneous on all channels;
13. Real time 7-bit FFT using double buffering;
14. Persistent software settings and multiple non-volatile user memories;
15. Self calibration and Diagnostics;
16. Automatic and Manual Tuning of the internal fast RC oscillator for increased accuracy;
17. Adaptive Potentiometer controls;
18. Silence Detection and Triggering;
19. Input Op-amp Stage Clipping Detection;

Final Checks

We assume that you have read the first and second parts of the article in the previous two issues of SILICON CHIP. If you have followed the construction in last month's issue, you should have a fully assembled DSP Musicolour in its case and you should have tested the supply rails at test points TP0 and TP1. These should have measured close to their correct values of +5V and +10V respectively.

If they do not measure close to these values, you should refer to the troubleshooting section below for further hints, although we mentioned some extra checks to perform in that case in last month's issue, to which you should refer first.

Now before proceeding to operate the DSP Musicolour, you should check again that all the mains wiring on the back panel has been correctly performed as shown in Fig.6 of last month's issue (page. 32).

Make certain that the IEC sockets are correctly wired, using insulated spade lugs and secured with cable ties. All mains wiring should be correctly crimped. The IEC sockets usually have E, L and N markings indicating the earth, live and neutral connections respectively. You should double check these, for safety's sake, well before applying mains power.

You should also make sure that all the points on the back panel are properly earthed. To this end, you should use a multi-meter to check the continuity between the earth pin of an IEC cable connected to the male IEC socket on the Musicolour (obviously unplugged from any GPO) and the metal back panel of the DSP Musicolour.

You should also check that pins 1 and 5 of CON2 on the main board are also earthed (these pins connect to the GND plane of the low voltage side of the main board). You can check for continuity by using an Ohmmeter, in that case, continuity is indicated by a very low resistance.

The Operation of LK4 and Jumper Settings

In last month's article, we mentioned that LK4 is optional and is normally omitted. When LK4 is omitted, the DSP Musicolour has three independent audio inputs which can be used to modulate the output channels. These are the Microphone input, the Left Audio channel and the Right Audio Channel.

The Left and Right Audio channels are accepted from the speaker terminals on the back panel and are attenuated by the LEFT and RIGHT potentiometers on the front panel. The Microphone line input is a mix of the on-board electret microphone signal and any external microphone connected to the 6.5mm jack on the back panel.

When LK4 is installed, however, the Left and Right Audio channels from the back panel are mixed in hardware before being digitized by the microcontroller. In this case, either the LEFT or RIGHT potentiometer can be used to attenuate the input signal before it is digitized (and you should not install both potentiometers).

In other words, when LK4 is installed you lose one independent audio channel but you need only one potentiometer on the front panel (rather than two) to attenuate the audio input signal before it is digitized. LK4 therefore controls whether the DSP Musicolour Left and Right audio channels are mixed in hardware or software (Dual mono or joint stereo). The most common option is to omit LK4 and to disable hardware mixing.

With regards to the jumper links LK5 and LK6, we mentioned in the first part of the article that the default is to have LK5 installed and LK6 omitted. The current version of the firmware will ignore the state of links LK5 and LK6 although they may be used in future versions of the firmware. You can ignore these jumper settings for this version of the firmware.

Applying Mains Power For the First Time

If you are confident that your constructed DSP Musicolour is properly earthed, and that all mains wiring has been correctly installed, you can proceed to apply mains power for the first time.

First, make sure that the plastic case has been screwed shut. You should never apply mains power while ever the case is open. Did you install links LK1, LK2 and LK3 according to your local area's mains supply voltage? You should have installed LK2 and omitted links LK1 and LK3 under the mains transformer for 240V operation. On the other hand, you should have installed links LK1 and LK3 and omitted LK2 for 120V operation. These must be correct before applying mains power.

Now turn the switch on the back panel to the OFF position. Did you remember to install the fuse in the male IEC socket? You should install a 10A fuse for 240V operation and a 15A fuse for 120V operation as explained in last month's article.

You may then connect an IEC cable from your power outlet to the DSP Musicolour's male IEC socket on the back panel and turn the power switch to the ON position.

When you first apply power, you should see the start up screen scroll past on the dot matrix LED display. The DSP Musicolour will go through a number of tests and then go to its default state.

If you do not, you should switch off mains power immediately and go to the Troubleshooting section below.

The Boot-Up Sequence

When the DSP Musicolour first boots up, it goes through a number of internal checks before commencing operation. The following occurs on boot up (in chronological order):

1. The firmware displays the start up screen and its version number (this can be disabled for a quicker boot by changing the start up settings in the SYSTEM>Startup sub-menu).
2. The firmware measures the frequency of the mains supply. If the measured value is within tolerance, the firmware accepts the measured value and assigns its internal settings for either 50Hz or 60Hz operation (the firmware chooses the value closest to the measured frequency). If the measured value is not within tolerance, the firmware will display a warning indicating that no mains was detected and will default to 60Hz operation. In normal mains powered operation, this warning should never be issued. If it is, it indicates a problem with the Musicolour's zero crossing detection system. The fall back value of 60Hz was chosen because it is the safer value for the purpose of controlling the Triacs in the output stages. If you are running from a 50Hz supply and somehow the frequency is not properly detected and defaults to 60Hz, the shorter mains period will at worst mean that less power is delivered to the output loads. Therefore even if your mains supply is 50Hz and the detection

fails and defaults to 60Hz, at least the outputs will not flicker. Flickering can occur when the Triacs are switched on beyond the next zero crossing of the mains waveform and should not occur in normal operation. Another fail-safe feature is that in the rare event that no good mains frequency is detected, the firmware will disable all output channels (the rest of the firmware will function normally, however);

3. The firmware will load any persistent settings from the last active session and initialize all internal peripherals, including enabling all interrupts in the correct sequence;
4. The firmware will detect and enable any connected accessories. It is possible to add a small infrared remote control PC board to the main board to allow the DSP Musicolour to be operated by a standard RC5 remote control. It is also possible to add an infrared IrDA compatible port or RS-232 UART port to the main board to allow the Musicolour to communicate with a PC (wired or wireless). This allows the firmware to be upgraded and for permanent tweaks to be applied to the firmware defaults. Some of the advanced features of the firmware will be explained in a future article. Using the UART, the microcontroller's flash memory is accessible and programmable. The firmware implements a RTSP (Run Time Self Programming) server in a secure part of program memory (it also switches the interrupt vectors to an alternative location). The DSP Musicolour is therefore highly customizable but for most applications you will not need to change any settings, as the preloaded defaults should be adequate. Notice that it is also possible to upgrade the firmware using the low cost dsPIC/PIC programmer published in the May 2008 issue of SILICON CHIP and CON3 on the main board. We will explain these additional features in an upcoming article;
5. The firmware will jump to the main loop (described below).

Automatic Calibration

Although the DSP Musicolour will adjust its settings according to the detected mains frequency, all its calculations assume a fixed system clock.

The system clock is derived from the microcontroller's (dsPIC30F4011) internal fast RC oscillator (nominally 7.37MHz) and a 16xPLL multiplication stage is used to achieve around 30MIPs operation (4 clocks per instruction).

Since this oscillator's frequency tolerance can be relatively high due to internal manufacturing variations, it may be necessary, if you are experiencing unusual effects like flickering lights on the output channels, to calibrate the frequency as close to 7.37MHz as possible. This is a good thing to do just in case.

The dsPIC30F4011 has an internal non-volatile calibration setting to achieve this, meaning the internal fast RC oscillator can be tuned to bring it as close as possible to its intended frequency.

The firmware assumes that the mains line frequency is very close to its theoretical value of either 50Hz (if you are in Australia, Europe and most other parts of the world) or 60Hz (if you are in America, Japan and a few other places).

Since the mains frequency can be measured by the firmware against the system clock, the firmware can then calculate the error in the internal fast RC oscillator and automatically adjust it to minimize the error.

This is what the firmware does in its automatic calibration. You need only do this once when you first run the DSP Musicolour, the setting will be preserved even when powered off.

To do this, go to the ADVANCED>Calibration sub-menu. If you wish to see how far from the ideal the microcontroller is operating, go to the INFORMATION>Error menu and the current percentage error in the measure mains frequency will be displayed.

Advanced Note: It is also useful to calibrate the internal fast RC oscillator to get a more accurate Baud Rate and minimize any errors when using the RS232 UART (this is an additional feature that will be explained in a future article. The UART will operate best when the internal fast RC oscillator is running close to 7.3728MHz).

Advanced Note: The internal calibration of the system clock is a 4 bit value. Values from 0-7 increase the nominal speed while values from 8-15 decrease the nominal speed. The current value of this calibration can be seen in ADVANCED>Tuning. This menu also allows you to manual tune the oscillator.

Advanced Note: It is possible to measure the current system frequency by monitoring the RF6 output (pin 9 of CON3). In order to do this you will need an oscilloscope and set the RF6 mode to 9. You will be able to directly measure the system clock frequency (FCY) at around 29.480 MHz.

The Main Loop

After boot up, the firmware spends most of its time in the main loop. When the main loop is running, the Musicolour is either in Automatic mode (the AUTO LED is lit) or User Mode (the USER LED is lit). In the main loop, the AUTO LED or the USER LED will flash if there is silence detected on the input signal. The SET LED will also flash if there is clipping detected in the op-amp stages of either the Left or Right audio input channels.

Clipping refers to when the input signal saturates the input op-amp stages. While an amplifier's normal response is approximately linear, clipping will distort the input signal and make the response highly non-linear. While clipping is very undesirable in an audio amplifier, greatly affecting the THD, its effect may actually be desirable in the DSP Musicolour. The firmware detects clipping of the left and right channel op-amps. This is merely an indication that you may want to increase the attenuation using the LEFT and RIGHT potentiometers.

While in the main loop, you may enter the menu system by pressing the SET button.

The display is refreshed according to the currently selected display. The selected display can be scrolled to the next available display using the AUTO button when already in Automatic Mode and the USER button when already in User Mode.

The implemented main loop displays are as follows (Table 1.). Note that they do not affect the internal operation of the DSP Musicolour, they only affect what the display shows.

Spectrum Fine: the spectrum is displayed on the display from lowest to highest frequency (left to right);
Spectrum Centered: the spectrum is displayed in centered mode;
Logical Channel Displayed Single: the output levels of each channel are displayed. The top horizontal bar indicates the first logical channel’s level. The third horizontal bar from the top indicates the second

logical channel's level. Similarly, the fifth and seventh horizontal bars from the top indicate the third and fourth logical channels' levels respectively.
Logical Channel Display Averaging: same as above except every horizontal bar in between the output channel bars is the average of the previous and next bars.
RMS display: displays the RMS level of the input signal as an analog meter.

Table 1: Showing the display modes in the main loop

In the main loop in Automatic Mode, the CH1-CH4 LEDs will light according to the logical output channel levels. The main difference between Automatic mode and User Mode is that in Automatic mode the Musicolour uses its current settings for all functions, whereas in User Mode, one of four previously stored settings profiles is used instead. Thus User Mode can be used to quickly run the Musicolour in a previously set configuration. In User Mode, the CH1-CH4 LEDs will indicate one of the four preset profiles currently active. You may press the CH1-CH4 buttons to change the preset on the fly while in User Mode.

The main loop running in Automatic or User Mode consists of the following sequence, which is diagrammatically represented in Fig.1.

Logical and Physical Channels

We should first mention that the firmware supports four logical output channels and four physical output channels. The physical output channels correspond to the outputs on the back panel. Each of these can be associated to a logical channel. In normal operation, the physical channel N is associated to the logical channel N. However, added effects can be achieved by changing the mapping from output channels to logical channels. For example, you can have all four physical channels on the back panel controlled by one logical output channel.

The Main Loop: In Detail

We now explain the operation of the main loop. As can be seen in Fig.1, the three inputs consisting of the Left and Right audio inputs and the Microphone inputs are digitized and mixed according to the balance settings under AUDIO>Balance. Any combination of these three channels can be used as the input signal. The result of this software mixing is passed to the FFT system.

The Fourier Transform

The resulting mixed audio signal is taken as the audio input and the Fourier Transform is computed on this mixed signal. The implemented FFT (Fast Fourier Transform) resolution of the current version of the firmware is 7 bits. Memory constraints as well as time constraints were a factor in deciding this. The next level, an 8 bit FFT would have required double the amount of memory and take too long to compute in real time, so that some samples would have been dropped.

The output of the FFT therefore resolves the captured slice of the input audio waveform into 128 (2^7) frequency amplitudes. These are equally spaced from 0Hz up to the sampling frequency.

The smallest frequency that the FFT can resolve is therefore F/128, where F is the sampling frequency.

For example, when F is 48kHz we can resolve down to 375Hz or +/- 137Hz. If you are not very interested in the audio sub-band above, say, 10kHz then you can lower the sampling frequency to

20kHz and the FFT will be able to resolve frequency components down to 156Hz or +/- 78Hz.

The audible spectrum for humans includes frequencies anywhere from about 20Hz to 22kHz. Although the sampling frequency of the ADC system can be set anywhere from 16kHz up to 50kHz keep in mind that according to the Nyquist sampling theorem the highest frequency that can be resolved using a sampling frequency F is F/2. Aliasing will occur for any input signals higher than half the sampling frequency. Aliasing refers to when different frequency components in the input signal are not differentiated because the sampling frequency is too low.

This means that aliasing will occur at least somewhere in the audible spectrum if the ADC system's sampling frequency is set below about 44kHz. Aliasing is usually an unwanted characteristic of a digitizing system, since different frequencies cannot be distinguished. However, since most music has very little high harmonic content, in fact, very little content above 4kHz, it may be desirable to lower the sampling frequency in order to increase the resolution of the FFT.

The only trade off is that aliasing may occur, but this is probably not a problem. In other words, if you want greater resolution, you should lower the sampling frequency.

The FFT system computes logical output channel levels in two acquisition modes. These can be set in the CHANNELS>Mode sub-menu.

The two acquisition modes are PEAK and AVERAGE. In AVERAGE mode, the average of the relevant frequency components falling within the channel's passband (set by the minimum and maximum Frequencies for the channel) will be the output level requested in the output stage. In PEAK mode, however, only the maximum level within the channel's passband will be the output level requested in the output stage.

Additionally, equalization can be enabled or disabled for each channel. The levels of the equalizer can be changed, however, by going to the AUDIO>Equalizer sub-menu.

There is an optional Equalizer module which can be enabled or disabled. The equalizer has 8 bands set to affect preset portions of the audible spectrum. The current settings of the equalizer bands can be seen under the INFORMATION>Equalizer sub-menu and cannot be changed by the user (it can be changed by reprogramming the device however).

Note that the equalizer affects the output of the FFT, not the input. After any equalization is performed, the levels of the logical output channels are set by the FFT system, if the chaser mode is set to AUTO (see the CHASER>Mode sub-menu below).

If the chaser mode is set to NORMAL then the level data produced by the FFT system is ignored and the data produced by the current chaser program (see the CHASER>Program sub-menu below) is used to set the logical output channel levels instead.

If the chaser mode is set to TRIGGERED, then the trigger pulse (produced by the FFT system) is used to step through the chaser program.

Triggering

Triggering affects the Chaser system when it is enabled. There are two sources of triggering.

Triggering can either occur directly from the Trigger channel or from the Silence detection. Silence detection triggers when there is a relative silence in the input Left or Right audio signals (the MIC input is not used for the silence detection).

The trigger can be considered a separate logical channel. It has its own selectable pass-band and threshold. When the threshold is reached the trigger occurs. The trigger is used by the chaser system to trigger the current chaser program in TRIGGERED mode.

Tips: If, for example, you want a bass response triggering a pre-set chaser program, you would set the TRIGGER minimum frequency to 0Hz and the maximum frequency to around 300Hz. Then adjust the threshold level to get an acceptable level of triggering.

The Chaser System

The Chaser system is implemented as a virtual machine. A small chaser program is executed from internal memory when a chaser program is active. The chaser program is either executed at the rate set in OUTPUT>Output Rate or each step in the program is triggered.

Setting the Output Levels

Both the Chaser system and the FFT system produce a set of output levels for the logical output channels. Depending on the chaser mode being used, the chaser levels or the FFT levels will be used to change the levels of the logical output channels.

These will then affect the physical output channels.

Channel Modes and Settings

Each of the four logical output channels can operate in one of four primary modes: DIRECT mode, CONTINUOUS Mode, ZV Mode or STROBE mode.

In the DIRECT and CONTINUOUS modes, the brightness of the logical output channel is varied in 256 levels (8 bit resolution). The output brightness is approximately linear as the firmware uses an internal dimming curve to correct the non-linearity inherent in phase control. The difference between DIRECT and CONTINUOUS mode is how the output level is set by the output system.

The four logical output channels respond to level requests from either the FFT system or the Chaser virtual machine. The main difference between the DIRECT and CONTINUOUS modes is that while in DIRECT mode the brightness is set directly, in CONTINUOUS mode, the brightness is “continuously” modified from the current brightness level. In other words, in CONTINUOUS mode, if the requested level is higher than the current level, the current level is increased by the ATTACK setting for the channel, while if the requested level is lower than the current level, the current level is decreased by the DECAY setting for the channel. Setting different ATTACK and DECAY levels for the channel can affect the level of the output logical channels in CONTINUOUS mode.

In ZV Mode, the output channel responds as in DIRECT and CONTINUOUS modes, except that the output is not a brightness level but a digital output. The output is either fully on or fully off. This mode approximates a zero voltage switching mode and can be used to reduce RF interference or can be used to achieve a digital effect.

In STROBE Mode, the output level sets the frequency of the logical output channel rather than the brightness level. The strobe frequency will be set from the maximum (equal to the mains supply frequency, either 50Hz or 60Hz, down to 1/256th of the mains supply frequency, ie. around 0.2Hz).

Quiescent Level or Filament Preheat

Remember that in all channel modes except STROBE, each logical output channel has a settable quiescent level. The quiescent level can be set by going to OUTPUT>Quiescent Level and is settable from 0 to 25% of the full brightness level. Use this to reduce the stress on the filaments in your incandescent lamps and to reduce surge currents through the Triacs at switch on. Note that if the channel mode is ZV (zero voltage switching) and the Quiescent Level is not 0% the channel may seem to be continuously on. In this case, you should set the Quiescent Level to 0% or disable ZV mode.

Advanced Note: A closer look at the operation of the Musicolour

The output Triacs are controlled through the optocouplers using the four output compare channels of IC1 (dsPIC30F4011). In order to maintain a constant brightness of the output lights, it is necessary for the switch on triggers to the Triacs to be synchronized to the frequency of the mains supply. To achieve this, the microcontroller uses the INT0 external interrupt pin which is supplied by one side of the transformer's secondary winding. An interrupt can be triggered on a rising or falling edge of INT0. Now a low level on INT0 is any voltage lower than about 1.5V while a high level is considered to be anything above 3.5V. We have a 5V supply but a 7.5V secondary winding. This means that the triggers to INT0 (which is the microcontroller's zero detection interrupt) are asymmetrical. The measured duty cycle is about 42% rather than the expected 50%. The firmware corrects this asymmetry, adjusting the value of a phase counter to take account of this.

Compare Fig.2. (without software correction) and Fig.3 (with software correction). In the scope grab of Fig.3. the yellow trace is the output of the Triac and the green trace is the trigger pulse.

You can see that the trigger pulse period is only 8.5ms whereas for symmetrical triggering it should be close to 10ms (this is the 100Hz rate which is twice the mains frequency in Australia) as shown in Fig.3.

Troubleshooting Tips:

We collect some common problems that may help you troubleshoot the DSP Musicolour.

Problem:

You apply power and there seems to be no activity, there is no display.

Possible Cause:

Have you installed a fuse? Is the fuse open circuit?

Problem:

You apply power and there is a sudden short of the mains supply (consequently the fuse blows or the circuit breakers/fuse in your home open). There seems to be a short of the mains supply.

Possible Cause:

This could be caused by incorrect link settings for LK1, LK2 and LK3 underneath the mains transformer.

Problem:

One or both voltages at test-points TP0 and TP1 are not at normal levels around +5V and +10V respectively.

Possible Cause:

One possible cause is that links LK1, LK2, LK3 are improperly set or omitted. Remember these have to be installed according to the mains supply voltage. Install LK2 and omit LK1 and LK3 if you are using a 220-240V mains supply and install LK1 and LK3 and omit LK2 if you are using a 110-120V mains supply. These links are found under the mains transformer, so you may have to unsolder the transformer to check them. If you have erroneously configured these links for 110-120V operation while you are actually in a 220-240V region of the world, you will get double the intended voltage at test-point TP1. This can very easily destroy REG1 and cause further damage to the main PC board. Disconnect power immediately if the voltages at TP0 or TP1 are much higher than their intended values.

Problem:

The main board seems to be operating correctly, except nothing is shown on the dot matrix LED display.

Possible Cause:

The most common cause of this problem is that the 26way ribbon cable connecting the main board and the display board is either faulty, or not all connections are good, or it is incorrectly oriented. If you can verify that the ribbon cable's 26 connectors are good, it may indicate a fault with incorrectly oriented parts. Check the transistors and ICs are correctly oriented on the display PC board. Check also that the dot matrix LED modules are in their sockets the right way around.

Problem:

At least one key does not respond to key presses or its LED does not light up.

Possible Cause:

This is most likely caused by the tactile switch being incorrectly oriented, improperly soldered, or its accompanying diode being incorrectly oriented.

Problem

When in a menu, moving the SELECT potentiometer does not affect the setting, or does so after much turning.

Possible Cause:

This is most likely not a problem but a feature. It is called adaptive control. See the text for an explanation.

Tips and Tweaks

Notice that if the display frequency is set too low you will get strange effects. Also, if the sampling frequency is lower than twice the highest frequency of the input audio, aliasing will occur.

User Operation of the DSP Musicolour

We now review the user operation of the DSP Musicolour. The Musicolour has many settings which can be changed by the user. As mentioned, the preloaded default values should be adequate for most applications.

Front Panel

There are seven push buttons on the front panel which are used to navigate through the menus and change internal settings. Some buttons have multiple functions according to context. The SELECT potentiometer is also context sensitive and is used to change settings. The incorporated LEDs in each of the buttons will light depending on the context. Usually, a lit or flashing button will mean that the button has an active function in the current menu. When the firmware is executing the main loop, the LEDs will indicate the state of the output channels and the current operating mode.

Adaptive Potentiometer Controls

The DSP Musicolour firmware implements adaptive potentiometer controls. This means that if a setting is to be modified using the SELECT potentiometer, the setting will begin to change only when the potentiometer position first matches the current value of the setting. This gives the potentiometer a kind of memory and is used to seamlessly change internal settings depending on the current menu.

Menu System

The settings of the DSP Musicolour are changed through a hierarchical menu system. When the DSP Musicolour is in the main loop, pressing the SET button allows you to enter MENU mode.

Keep in mind that some of the behaviour of the Musicolour is dependent on its current settings. For example, the display will be blank if the screen saver has been set to NONE and there is no key activity for the period of the screen saver timeout.

Quick Setup Checklist

The following panel shows how to setup the DSP Musicolour quickly and the relevant settings that will affect its operation:

1. Set the ADC system's sampling Frequency: go to AUDIO>Sampling Frequency.
2. Set the mixing settings for the input signal: go to AUDIO>Balance.
3. Set the minimum and maximum frequencies for each logical output channel: go to CHANNELS>Min Freq and CHANNELS>Max Freq. Alternatively go to CHANNELS>Freq to set a non-overlapping frequency mask.
4. Set the gain for each logical output channel: go to CHANNELS>Gain.
5. Set the mode for each logical output channel: go to CHANNELS>Mode.
6. Set the output connections of the logical channels: go to OUTPUT>Logical Channels.
7. Set the CHASER>Mode and CHASER>Program;
8. Exit the menu system.

User Operating Instructions: Menu System

A hierarchichal menu system is implemented. From the main loop press the SET button to enter the menu system. You will be directed to the main menu. In the main menu you may scroll up or down between sub-menus by using the UP and DOWN buttons. Use the SET button to enter a sub-menu. In any submenu, you may use the D button to go back to the previous menu (if you are in the main menu, you will be directed back to the main loop).

The following sub-menus are available in the main menu:

1. CHANNELS: this sub-menu allows you to change any settings related to the four logical channels;
2. TRIGGER: this sub-menu allows you to change the trigger passband and the trigger threshold;
3. CONSOLE: this sub-menu contains user applications, allowing the Musicolour to function as a light dimmer or communications terminal;
4. OUTPUT: this sub-menu is used to set the chaser mode, the chaser program, the output rate, the quiescent level of each physical channel and to define the logical to physical channel translation;
5. AUDIO: this sub-menu is used to change the equalizer settings, the software mixing/balance of the input signal and the sampling frequency;
6. DEFAULTS: this sub-menu is used to save and recall settings and to load default values;
7. ADVANCED: this sub-menu is used to access advanced features, including calibration, software upgrade and tuning;
8. INFORMATION: this sub-menu displays information about the Musicolour’s operation like the mains frequency, the frequency of the ADC system and the screen refresh frequency. The error in the overall accuracy of the timing system can also be seen;
9. DISPLAY: this sub-menu is used to change the display’s settings, including the screen refresh frequency, the screen brightness and the screen saver time out period;
10. SYSTEM: this sub-menu can be used to change system settings, the firmware version is displayed, the baud rate of the UART can be changed, the remote control system can be enabled and other system settings changed;

The Menu System

The following sub-menus are implemented in version 1.00 and above:

CHANNEL submenus:

CHANNELS>Min Freq: Press the channel buttons CH1-CH4 to display the current minimum frequency for that logical channel; Use the SELECT potentiometer to change the minimum frequency;

CHANNELS>Max Freq: Press the channel buttons CH1-CH4 to display the current maximum frequency for that logical channel; Use the SELECT potentiometer to change the maximum frequency;

CHANNELS>Freq: this is similar to the CHANNELS>Min Freq menu, except that after exiting, the minimum and maximum frequencies for the four channels are set in non-overlapped fashion.

CHANNELS>Gain: Press the channel buttons CH1-CH4 to display the current gain for that logical channel; Use the SELECT potentiometer to change the gain;

CHANNELS>Mode: Press the channel buttons CH1-CH4 to display the current mode for that logical channel; Use the UP and DOWN buttons to scroll through the available modes;

CHANNELS>Attack: Press the channel buttons CH1-CH4 to display the current attack rate for that logical channel; This is only relevant when the channel is operating in CONTINUOUS mode. Use the SELECT potentiometer to change the attack rate;

CHANNELS>Decay: Press the channel buttons CH1-CH4 to display the current decay rate for that logical channel; This is only relevant when the channel is operating in CONTINUOUS mode. Use the SELECT potentiometer to change the decay rate;

CHANNELS>Test Channel: Press the channel buttons CH1-CH4 to test the relevant logical channel with a range of output level requests from 0 to full level; This can be used to test the current settings for the channel.

CHANNELS>Defaults: Press SET to restore default CHANNEL sub-menu values;

TRIGGER sub-menus:

TRIGGER>Min Freq: Press the UP and DOWN buttons to set the minimum frequency defining the trigger pass-band. Exit using the SET button.

TRIGGER>Max Freq: Press the UP and DOWN buttons to set the maximum frequency defining the trigger pass-band. Exit using the SET button.

TRIGGER>Threshold: Use the SELECT potentiometer to change the threshold level for the trigger. Triggering will occur when the input signal has an amplitude component within the trigger pass-band that is greater than the trigger threshold. The level is indicated as a horizontal bar. Exit using the SET button.

TRIGGER>Defaults: Press SET to restore default TRIGGER sub-menu values.

CONSOLE sub-menus:

CONSOLE>Dimmer: Press the channel buttons CH1-CH4 to select the relevant logical output channel. Use the SELECT potentiometer to change the output level of this channel. Here the Musicolour functions as a four channel dimmer;

CONSOLE>Com: The Musicolour enters an echo terminal mode. Received data from the UART is displayed on the display. The UART can be enabled using additional hardware.

OUTPUT sub-menus:

OUTPUT>Chaser Mode: the current chaser mode is displayed. Press the SET button to scroll to the next available mode.

OUTPUT>Chaser Program: the current chaser program is displayed. Press the UP and DOWN buttons to set the program. Press SET to exit.

OUTPUT>Output Rate: the current output rate is displayed. Use the SELECT potentiometer to change the rate; Press SET to exit.

OUTPUT>Quiescent Level: Press the channel buttons CH1-CH4 to select the relevant logical output channel. Use the SELECT potentiometer to change the quiescent level of this channel.

OUTPUT>Logical Channels: Press the channel buttons CH1-CH4 to select the relevant physical output channel (on the back panel). Use the UP and DOWN buttons to change the logical output channel associated to that physical channel. In Normal operation, you set CH1=1, CH2=2, CH3=3, CH4=4; if for example, you wish to have logical channel CH1 control two physical outputs on the back panel you could set CH1=1 CH2=1 CH3=3 CH4=4; If you would like to permute the channels you can also do that here.

OUTPUT>Defaults: Press SET to restore default OUTPUT sub-menu values.

AUDIO sub-menus:

AUDIO>Equalizer: The current equalizer settings are shown as vertical bars. Use the UP and DOWN buttons to scroll to the next setting, and use the SELECT potentiometer to vary the current equalizer setting.

AUDIO>Balance: the current percentages of each the three audio channels contributing to the input signal are shown. Press SET to change these. The levels are then displayed as bars. The first bar from the left is the MIC line level. The next two bars indicate the LEFT and RIGHT levels respectively. Use the SELECT potentiometer to change the LEFT/RIGHT balance. Use the UP and DOWN buttons to change the MIC contribution to the input signal.

AUDIO>Sampling Frequency: the current sampling frequency in kHz is displayed. Press SET and use the SELECT potentiometer to vary this value.

AUDIO>Defaults: Press SET to restore default AUDIO sub-menu values.

DEFAULTS sub-menus:

DEFAULTS>Load Defaults: Press SET to restore all settings to default values;

DEFAULTS>Save Settings: Press UP and DOWN buttons to change the memory number to save to. Press SET to save all current settings to non volatile memory.

DEFAULTS>Recall Settings: Press UP and DOWN buttons to change the memory number to load values from. Press SET to load all settings with previously stored values.

ADVANCED sub-menus:

ADVANCED>Calibration: Press SET to automatically calibrate the Musicolour's internal fast RC oscillator against the mains frequency. Needs only be done once.

ADVANCED>Software Upgrade: Press SET to upgrade the firmware. This mode requires a functioning UART connection, which needs additional hardware.

ADVANCED>Tune Oscillator: Press UP and DOWN to change the internal calibrating value for the system clock. This value is updated by the automatic calibration above. You can manually adjust the value here.

INFORMATION sub-menus:

There are no settings to change here. Only the values of certain system parameters are displayed. This is for operating information like the mains frequency, the screen refresh rate, the sampling frequency, the system clock, the error in the system timing from the ideal, etc.

DISPLAY sub-menus:

DISPLAY>Brightness: Press SET to change, using the SELECT potentiometer, the brightness of the display.

DISPLAY>Frequency: Press SET to change, using the SELECT potentiometer, the screen refresh frequency. Note that strange display effects can occur at low screen refresh frequencies. If this is the case, increase the frequency. Usually a level around 65Hz or higher is adequate.

DISPLAY>Timeout: Press SET and use the UP and DOWN buttons to select the timeout period for the screen saver.

DISPLAY>Screen Saver: Press SET and use the UP and DOWN buttons to select the current screen saver.

DISPLAY>Display Defaults: Press SET to restore all DISPLAY submenu defaults;

SYSTEM sub-menus:

SYSTEM>Version: displays the current firmware version.

SYSTEM>Uart: Press UP and DOWN to change the baud rate for the UART. This requires additional hardware.

SYSTEM>Remote Control: Press SET to enable or disable the remote control decoding. This requires additional hardware and can be used to control the Musicolour using an RC5 compatible remote control.

SYSTEM>IrDA: Press SET to enable or disable the IrDA decoding. This requires additional hardware and can be used to add a wireless infrared serial port. This can be used to send and receive data from a PC.

SYSTEM>Test: Press SET to run a test on the display, the output channels and the LEDs. Can be used to check that all these are working correctly.

SYSTEM>Detected Mains: this shows the detected mains frequency and is either 50Hz or 60Hz. It should match your area's mains supply frequency.

SYSTEM>Startup: Press SET to scroll through the start up modes for the Musicolour. The initial startup can be made quicker by disabling the normal boot-up messages.

SYSTEM>RF6: Press UP and DOWN to change the RF6 pin mode. This is an advanced feature that can be useful to debug any problems with the Musicolour. The RF6 output of the microcontroller is a digital output and is available at pin 9 of CON3 on the main board. The system clock frequency can be measured at this pin, as well many other internal operating frequencies like the screen refresh frequency and the ADC system frequency. You will not need to normally use this menu.

SYSTEM>Reset: Press SET to reset the Musicolour.

SYSTEM>System Defaults: Press SET to load SYSTEM submenu defaults.

DSP Musicolour Firmware Version 1.60 Extra Menus:

In version 1.60 there is an extra sub menu in the main menu:

QUICK SETUP sub-menu:

This sub-menu can be used to load default settings for a number of different "generic" configurations. Go here to setup the DSP Musicolour with one touch.

OUTPUT>ZV Threshold: controls the threshold for all the output channels. Below this threshold, the output channel is fully OFF and above it is fully ON when in ZV (zero voltage mode switching) mode. You can vary the level with the SELECT potentiometer.

OUTPUT>Silence Threshold: controls the threshold for silence detection. When the audio level on the left and right channels are below this, the Musicolour detects it as silence, indicated on the front panel by a LED (flashing when silence between songs for example).
The silence detection also affects any output channel working in STROBE mode. As explained, in STROBE mode, the channel responds to its audio passband not in the normal way as a brightness level

but as a frequency instead. That is, normally (in DIRECT and CONTINUOUS modes) the channel is brighter when there is a greater component of audio in its passband, but in STROBE mode it flashes quicker instead of being brighter. And it flashes fully ON or fully OFF (approximating a zero voltage switching mode). Well, the silence threshold is also used for those output channels in STROBE mode. When the Musicolour detects silence (the detection is dependent on the Silence Threshold) any output channel in STROBE mode will turn OFF. This submenu makes the silence detection less or more sensitive.

QUICK SETUP>Auto: Press SET to load default settings for automatic operation. Good general-purpose defaults are loaded.

QUICK SETUP>High Contrast: Press SET to load default settings. All output channels are set to CONTINUOUS mode and other defaults are loaded so that the output channels will show better contrast and have delayed dimming with the music.

QUICK SETUP>Strobe: Press SET to load default settings that make the first output channel a STROBE mode output too.

QUICK SETUP>Chaser: Press SET to load default settings to put the DSP Musicolour into automatic chaser mode.

In other words, the four quick setup options can be used to load defaults for 4 different scenarios at the touch of a button. Remember that you can set and save your own defaults by going to the DEFAULTS submenu from the main menu.

In Depth Explanation of the Main Loop:

In more detail the main loop is as follows:

1. The firmware waits until the internal ADC system signals that the buffer has been filled with digitized and software mixed audio data (while waiting all interrupts are active, including all timers, key press detection and display refresh interrupts);
2. Once a full buffer of data has been acquired, the Fourier Transform is computed;
3. For each logical output channel, a level corresponding to the channel is computed. This may involve adjusting the output of the FFT with equalization, it will depend, for each channel on its selected acquisition mode;
4. A request is made, for each logical output channel to set its output level to the previously computed level. The implementation of this step is dependent on the channel's current setting. If a channel should be accepting data from an active chaser program, the level requested in this step is ignored;
5. If a Chaser program is active, it is serviced by the virtual machine; This may involve the triggering channel if the program is in trigger mode; Any output level requests made by the chaser program are set. Again, the implementation of this step is dependent on the channel's current settings;
6. The display is refreshed according to the currently selected display:
    1. Spectrum Fine: the spectrum is displayed on the display;
    2. Spectrum Centered: the spectrum is display in centered mode;
    3. Logical Channel Displayed Single: the output levels of each channel are displayed. The top horizontal bar indicates the first logical channel. The third horizontal bar from the top indicates the second logical channel. Similarly, the fifth and seventh horizontal bars from the top indicate the third and fourth logical channels respectively.
    4. Logical Channel Display Averaging: same as 3. above except every horizontal bar in between the output channel bars is the average of the previous and next bars.
    5. RMS display: displays the RMS level of the input signal as an analog meter.
    6. Oscilloscope Mode: display the input signal as a trace.
    7. Screen Saver: displays the currently selected screen saver;
    8. Statistics: display statistics relevant to the input signal;
7. The firmware updates any LEDs on the front panel and return to step 1. It also responds to key presses.