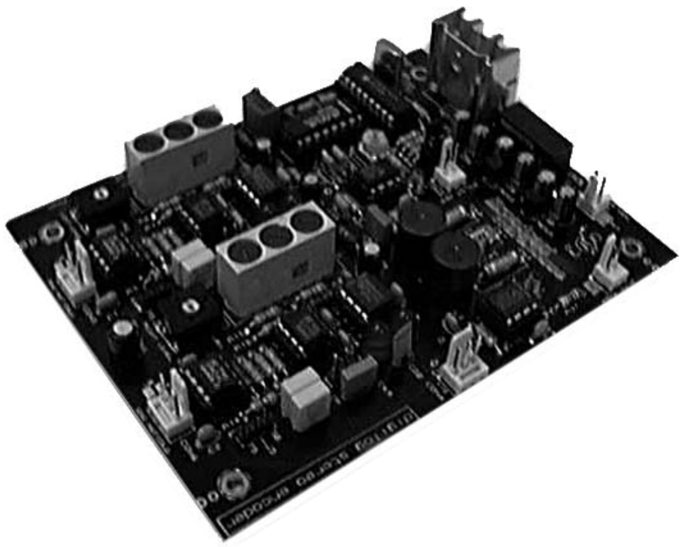
Broadcast Warehouse

DIGILOG STEREO ENCODER

The Broadcast Warehouse DIGILOG is a low cost, very high specification stereo encoder. Designed for optimum stereo seperation and excellent spectral cleanliness this encoder will provide that extra performance for radio stations who want something a little special. Audio baseband filtering combined with the latest digital techniques and post mpx filtering make this unit easily conform to even the most stringent broadcast regulations. And because the energy from the encoder is in the right places you will gain extra loudness inside your bandwidth limits.The low distortion pilot tone will allow excellent compatibility with subcarrier services such as the radio data system (RDS) High quality components and printed circuit board will ensure you have 24 hour operation for years and years. It's internal power supply allows the encoder to be run from 8 to 18 volts but still employ a split rail power supply for maximum dynamic range. Balanced input chips are onboard as standard allow balanced output professional mixing desks and audio processors to be connected to the encoder.

Features

Micro Computer controlled
More than 55db seperation
15khz brickwall filtering
Mpx filtering
Balanced audio inputs
Switchable pre emphasis
Switchable baseband clipping
Stereo/Mono switch
Audio input level controls
Socketed connecters
High grade low noise op amps
Switched mode power supply
Black oxide high grade PCB

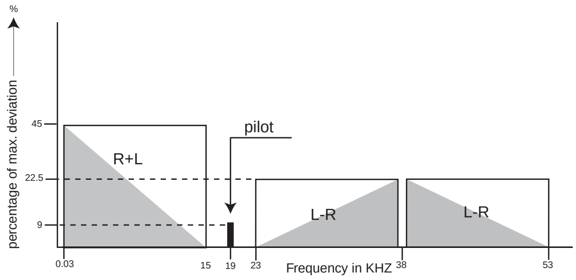


Specifications

Stereo seperation	>60 db
Pilot frequency	19 khz +/- 2 hz
Pilot THD	<0.1 %
Pilot level	9%
38 khz rejection	>65 db
IMD and beats.	>60 db (no clipping)
Input level	-10 db to +20 db
input impedance	10k Ohms
Output level	+6db
output impedance	75 Ohms
spurious> 100 khz	< -60 dbc
spurious> 200 khz	< -80 dbc
Pre emphasis	None, 50us, 75us (switchable)

DIGILOG STEREO ENCODER

What is stereo? And how does it work?



The diagram to the right shows the theoretical frequency spectrum of the stereo multiplex (mpx) signal applied to the modulation input of a VHF FM broadcast transmitter. The two stereo signals L (left) and R(right) are added as well as subtracted to give the corresponding sum (L+R) and difference (L-R)components. The L+R component occupies the lowest part of the spectrum, up to about 15khz, and affords compatibility with mono receivers. The L-R component is converted into a double sideband suprresed signal at 38khz. This type of modulation is called DSSC (double sideband suppressed carrier). It causes an upper sideband and a lower sideband mirrored against a suppressed (invisible) carrier (here, 38khz). The carrier is suprresed to keep the total deviation of the transmitter within limits.

At the receiver side, the 38khz carrier is recovered with the aid of the 19khz pilot tone contained in the MPX signal (relative level: 9%), which also serves to indicate a 'stereo' transmission. The 19khz pilot tone is doubled to give 38khz, and enables coherent demodulation of the L-R information. Next a matrix is used to distill the L and R signals from the components L+R and L-R.

Returning to the transmitter side, the levels of the components in the MPX signal are fixed to optimise the channel seperation given the available bandwidth for the FM signal, and also to ensure that the sound quality on a mono receiver is not impaired.

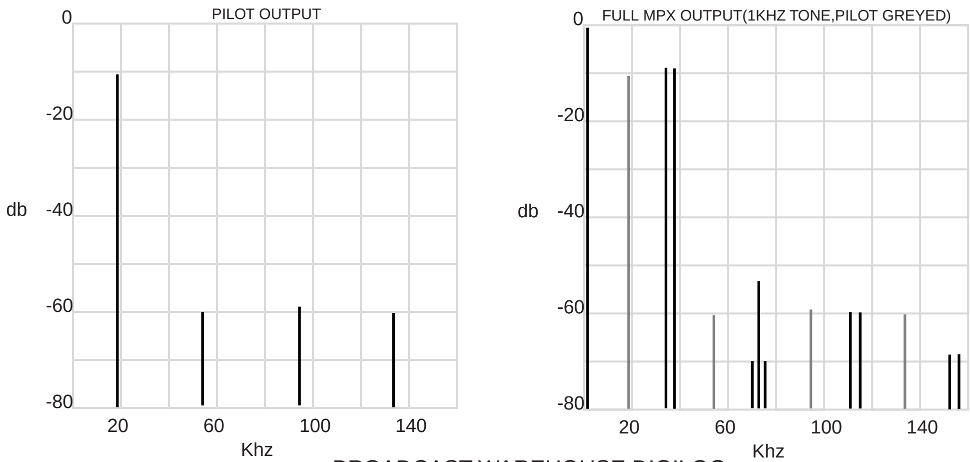
Overview of stereo encoders VS the DIGILOG.

The broadcast warehouse digilog stereo encoder employs some analog and some digital techniques to produce a very high quality adjustment and set up free stereo encoder. Why Digilog? A mixture of DIGItal and anaLOG Why the mixture? To answer these questions we need to look a little at both conventional low cost stereo encoders both analog and digital. For simplicity we will assume that the audio stages are identical and are band limited to 15khz on each channel. An analogue encoder will take the audio signals and have to form a matrix consisting of a l+r signal and a l-r signal. this takes extras components and circuitry and errors can be introduced at this stage of the process before we have even generated the multiplex. The l-r signal is then fed into a double balanced mixer which is also fed with a 38 khz signal. The output of the mixer produces a double sideband suppressed carrier. In theory great but, the chips used for the mixing process lack dynamic range(headroom) and often produce inter-modulation products when driven harder to obtain more dynamic range. The other problems are that the 38khz signal needs to be in perfect phase with the pilot signal. cheaper analog encoders will employ a variable frequency oscillator (adjustable) to generate both the 19 khz and 38khz signals. This is prone to drift and setup and periodic adjustment maybe necessary.The next step in the analog evolution chain was to digitally produce a 19khz and 38khz squarewave and then filter each one to get a squarewave back at the 19 khz and 38khz frequencys. Accurate filtering is needed to make sure the resulting 19khz and 38khz are perfectly in phase and so some coders may allow adjustment of this phase with variable inductors. Once again adjustment is needed and the inductors and capacitors of the filters are prone to drift with time and loss of seperation will occur. The bonus of analog coders is that they are pretty clean spectrally.

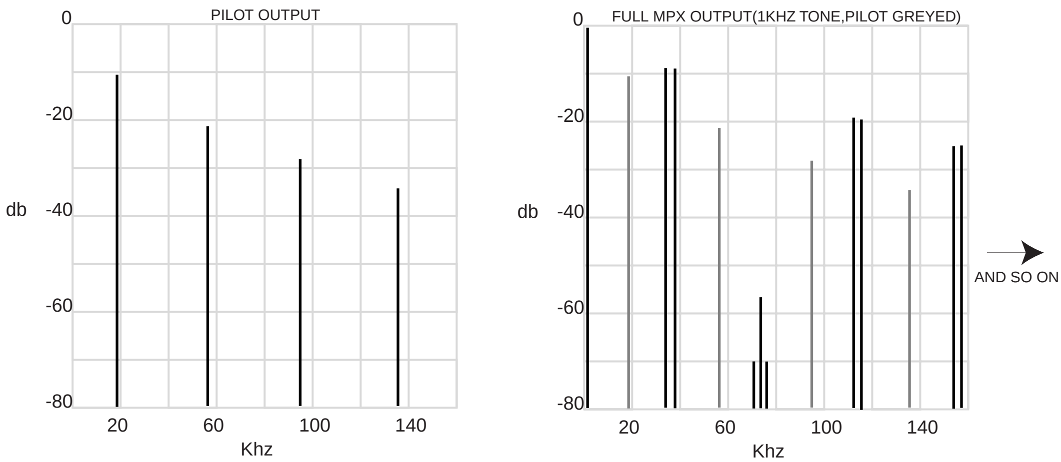
Cheap digital coders use the switching technique where the coder's switch alternates between the left and the right channels so that at any one time the DSSC signal created by switching the switch at 38khz only contains one channel which when you look at it is left minus right. This technique makes the circuit simpler by removing the need to generate your L-R signal beforehand. Cheap digital coders also just use the 19 and 38khz squarewaves as they are, unfiltered. they are Very easy to use and no setup is needed at all. The problem is the output spectrum. It is horrendous to say the least. The pilot's harmonics fall at a rate of 6db an octave and the pilots's output will extend out decaying into Mhz's away from the carrier. The same is true but even worse for the 38 khz DSSC signal whose harmonics extend further out and with more amplitude.Filtering could be used, but to remove the close in harmonics we would require a filter of such great length as not to cause phase shifts and destroy the seperation that it would not be worth it. A cheap digital coder can be implemented to sound good and have good seperation but the cleanliness of the output spectrum will be far from satisfactory. (diagram on next page)

The answer lies somewhere in between digital and analog. looking at the pilot first,we output a 19khz squarewave but at the same time add a signal to it that removes some of the edges. in between these two signals we do the same and the same in between them also. This produces a staircase type effect between a triangle wave and a sinewave rather than a squarewave. This together with the four weighting resistors used output a waveform that is more sine like than triangle. This pilot no longer contains any real signifficant energy at the 2nd to 5th harmonics. Any higher order harmonics are removed by the output lowpass filter (discussed later). The 38khz signal is generated in a similar manner. Where cheap switching encoders switch between the audio left and right alternatively the DIGILOG switches between them as well but also switches between them at different levels at different times. This has the effect of creating a digital sine like waveform that has low order harmonics suppressed. once again the output filter returns this to a sine. Because the output filter only needs to remove higher harmonics the need for such a stringent filter is no longer requried and this helps with keeping the seperation optimum. The cleanliness of the output spectrum helps in obtaining more volume as less energy is wasted in out of band harmonics. Overleaf is a figure illustrating the spectrum output of a conventional switching encoder vs the spectrum output of the DIGILOG.

DIGILOG STEREO ENCODER



BROADCAST WAREHOUSE DIGILOG



STANDARD CRYSTALLED SWITCHING CODER

The problems of Stereo.

When is good stereo not good stereo? the up's and downs of a stereo system!

The Broadcast Warehouse DIGILOG stereo encoder is capable of producing a stereo multiplex signal with a stereo seperation of over 50 db. Actually realising that amount of seperation over the whole audio frequency range not so simple. The problem is not so much the stereo encoder but everything else in the chain. The complete path from stereo encoder to listeners amplifier must have a flat frequency response across the 20 hz to 60 khz range and be without any phase shifts. We will look at each section of the chain in order starting with the transmitter the encoder is fed into and ending with the listeners radio.

The Transmitter.

The first problems can occur at the transmitters modulator. Incorrect values in the audio stages of the modulator will produce low frequency phase shifts affecting seperation. This problem is not the biggest threat and usually is not the worst culprit of phase shifts, and remember phase shifts cause bad stereo seperation. The main problem is the phase locked loop section of the transmitter.
What is a phase locked loop transmitter? A phase locked loop system of a transmitter is a circuit designed to stabilise a transmitters frequency and prevent drift of frequency. Modulation (audio) causes drift by the inherent characteristics of frequency modulation (FM), where the modulating signal (audio) causes a change in frequency of the fm transmitter. To stop the transmitter from being pulled back to frequency instantly by the PLL circuit (affectively cancelling any modulation), The frequency correcting signal is passed through a circuit called a loop filter. This loop filter dampens (smoothes and averages) the correcting pulses from the PLL circuit before passing the corrected voltage to the frequency control part of the modulator. The loop filter is usually the cause of the phase shifts due to not being able to sufficiently dampen and smooth the correcting pulses when the transmitter is fed with low frequencies. Where the loop filter can not correctly smooth and average the low frequencies phase shifts and amplitude increases occur. By lowering the frequency response of the loop filter the problem can be solved BUT by doing this we incur the penalty of increasing the lock time. This is why on professional transmitters such as the broadcast warehouse rack mount units the loop filter is built into two sections, one that has a low frequency response so low that the lock time (unaided) is over a few hours which is unacceptable so the second fast to frequency lock circuit switches in when off frequency and out again when on channel. The loop filter problem is more of an issue on a broadband no tune transmitter, because the modulator has to cover 88-108 mhz electronically (not by manual tuning). This large range means the modulator has to be more sensitive to cover the whole band and because of this sensitiveness the frequency correcting pulses (cont. on next page)................

are ten times more likely to cause phase shifts than say an exciter tuned to cover 2 mhz rather than 20 mhz.

So to summarise REAL BROADBAND transmitters will use a multi speed loop system that will ensure perfect phase down to very low frequencies. Kit broadband (no tune) transmitters will generally not and so will exhibit small phase shifts affecting the bass seperation slightly. Tuned PLL units will exhibit a bit less of a problem due to the fact that the modulator is not so sensitive. Variable frequency oscillators do not suffer from the problem at all due to no frequency correcting circuits (PLL).

In our 1 watt LCD PLL kit product we have choosen the values to obtain the best possible performance for a single loop system and still achieve reasonable lock times (< 30 secs), even so you can still not fully realise this stereo encoders full seperation. Other broadband kit units such as the panaxsis range lock in a few seconds, from there simple single loop circuit and quick lock time we can see that their PLL circuit is even more detrimental to the stereo multiplex signal.

In short, a badly designed transmitter can be hugely detrimental to the stereo signal created by a stereo encoder. Do not jump to the conclusion that the stereo sound that you are listening to is the stereo encoder only. It is a function of the stereo coder and the transmitter, As well as.................................

STOP PRESS: Broadcast warehouse now have a dual speed PLL kit, THE PLL PLUS. Use with this encoder for best performance.

Receiver. Filters, Bandwidths and Stereo Decoders.

Even if the transmitter adds no phase shifts to the multiplex signal transmitted, the receiver (radio) at the listening end can still add problems. Firstly the filters in the radio can cause phase shifts to the mulitplex if too narrow in bandwidth. While narrow bandwidth filters help to make the signal easier to receive and sound stronger they can be too narrow to pass a stereo signal without adding phase shifts. If you have a tuner with switchable filtering and many good tuners do, you can see what we mean by applying modulation to one channel of the stereo encoder and listening just to the other channel of your tuner to the crosstalk (seperation). By pressing the IF bandwidth (filtering) button you will see that the narrower the filters are the worse the seperation. A stereo coder must always be set up with the IF bandwidth as wide as possible. Many cheaper tuners have less filtering (less manufacturing cost) which although not great for selectivity provides for excellent seperation in strong signal environments. The above is only true if the stereo decoder in the radio or tuner is ok. Do you know that it is very hard to obtain any modern stereo decoder chips that give more than 45 db of seperation, some give only 35 db. So even with modern day DSP (digital signal processor) stereo encoders that achieve seperations of more than 70 db, you will never hear it because the radio you will be listening to it on may only allow 45 db at best.

We have three different tuners in the workshop. A Sony, Kenwood and a yamaha, and guess what. They all are different. You can set up the stereo coder's balance for optimum seperation on one and the others will need the coder to be readjusted to obtain optimum seperation on them. so even if you had a perfect 100 db seperation encoder you would never ever find two tuners that would realise the seperation. some will sound better than others.

So as you can see Stereo is not all about a stereo encoder. We have produced an encoder that will give excellent results, but how good the results you get are determined by so much of the above paragraphs. Maybe now you will have a better understanding of the problems faced in broadcasting a high quality stereo signal.

Circuit description.

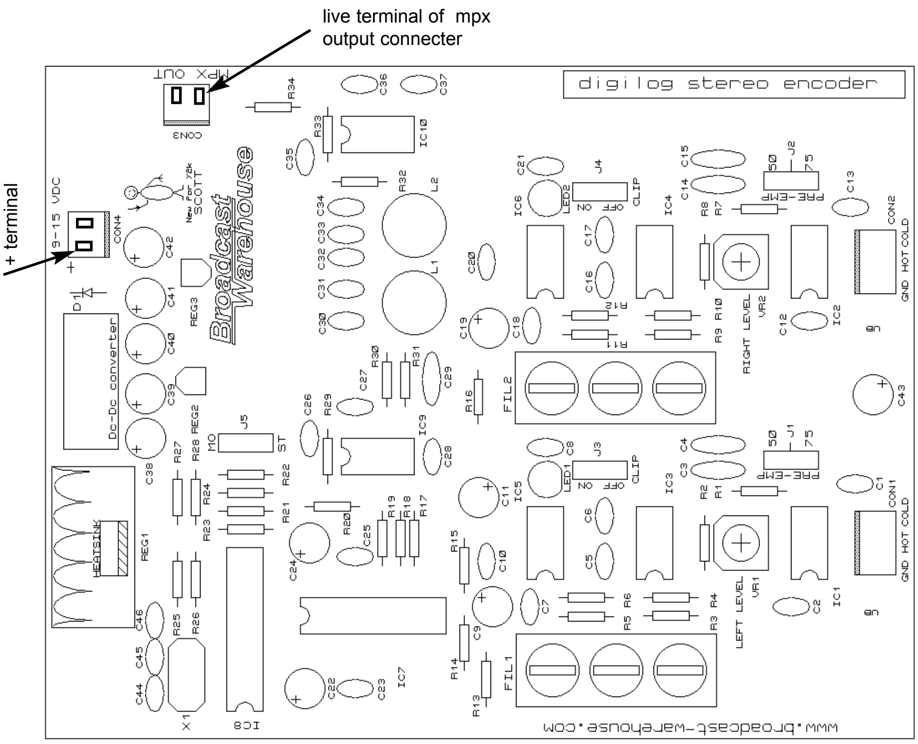
Left and Right audio signals are applied to connectors 1 and 2. These inputs are in the form of balanced inputs. unbalanced inputs can be applied also by application to the cold input pin of the connector. Input chips IC1 and IC2 convert the incoming signals to unbalanced before being passed to variable resistors VR1 and VR2 forming an input level adjustment control. The audio signals are fed from here into the active pre emphasis circuit formed by op amp's IC3 and IC4. jumpers J2 and J3 allow selection of none, 50 or 75 us pre emphasis. The outputs of the pre emphasis circuits are fed into 15 khz brickwall filter modules to remove any hf content above 15 khz and also provide a notch to protect the pilot. The filters are fed into buffer op amps IC5 and IC6 which also allow clipping via of the audio with jumpers J3 and J4. This facility can be used to remove any ringing in the preceding brickwall filters caused by squared off or clipped audio input to the coders inputs. The output from the buffer input also provides a drive into the analogue switch IC7. The control signals for the analog switch are generated by microcontroller IC8. the microcontroller also generates a 4 bit digital sine wave formed by the resistors R21 to R28. Jumper J5 allows the pilot to be shorted out performing a stereo / mono switch. the D/A 19 khz signal is fed via resistor R20 into virtual earth input op amp IC9. The outputs from the analog switch are also fed into the same op amp via R17-19 where they are combined with the pilot signal to form the composite signal. The output of the op amp is fed to a zero phase shift low pass filter that starts to cut in at about 90 khz and will remove any high order harmonics of the switching process and the pilot. Remember that due to the the D/A nature of the coder the 2nd,3rd,4th and 5th harmonics of the switching and the pilot are of an insignificant level. The filter's output is buffered by op amp IC10 which also serves as an output amplifier setting the output level and allowing a low impedance drive capability. The power supply for the encoder is formed by linear regulator REG1 which drops the single ended input supply to the coder down to 5 volts. this supplys the digital supply for the microcontroller as well as feeding the DC-DC convertor which produces a split rail plus and minus 15 volts. Each output feeds a linear 12v regulator whose outputs feed the circuits on the board.

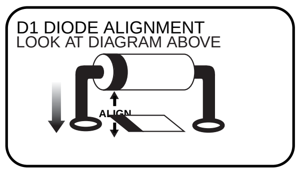
Assembly Instructions

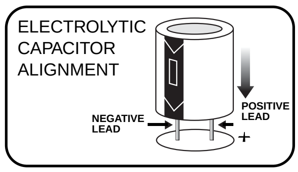
This kit is not really a first time kit builders project. If you have not soldered before we recommend you get some soldering experience from a simpler project or get this kit assembled by someone who has previous experience in electronic construction and soldering.

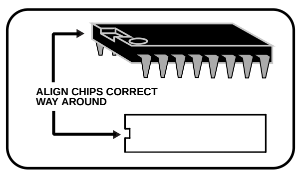
1.Empty the contents of the kit and proceed to check all of the components off against the component list, It is a good idea to tick off each component as you go through. When you have double checked all the parts proceed.
2. We always start with the lowest height components first which are resistors, Insert each resistor and solder one at a time taking care to make a good joint and not to short across any other pads/holes. Double check the component is the correct one before soldering.
3. Now insert Diode D1 observing the polarity (SEE DIAGRAM)
4. Next its time to insert the ceramic capacitors C7,18,26,31,35,44 and 45. Follow these with all of the decoupling capacitors which are C1,2,5,6,12,13,16,17,20,22,23,25,27,28,36,37 and 46. These are non polarized and can be inserted and soldered either way around.
5. Variable resistors VR1 and VR2 can be inserted and soldered next. Move on to the IC holders to fit in each of the chip positions. Make sure you line the notch on the chip holder with the notch on the ident on the printed circuit board. This wil help in making sure you insert the chip the correct way around in the socket.(SEE DIAGRAM)
6. Voltage regulators REG2 and REG3 are next and make sure again they are inserted the correct way around to match the ident on the board. Led's 1 and 2 can be inserted now making sure they are also the correct way around. They have a slant on one side, match it up with the slant on the ident.
7. Capacitors C3,4,8,14,15,21,29,30,32,33 and 34 can be next. Now insert the polarized electrolytic capacitors C19,24,38,39,40,41,42 and 43 MAKING 100% SURE they are soldered in correctly. (SEE DIAGRAM) The board has a positive symbol next to the positive hole of each polarized capacitor. Insert the negative stripe side away from the positive (+) marking.
8. Insert and solder jumpers J1,J2,J3,J4 and J5. you may if you wish put the jumper tab's on,but we recommend you wait till the end when we will configure the settings of the board. Connecters 1 to 4 can be soldered in if you wish to use them. They are industry standard locking molex types.
9. Next the DC-DC converter. Crystal X1 can be put in next followed by Inductors L1 and L2.
10. The low pass filter blocks are next. Do not hold the soldering iron on the pins for too long as you may damage the internal wires connected to the pins.
11.Lastly push the voltage regulator REG1 into the silver heatsink so that in can not go up anymore (it will be pressed up to the blip in the metal). Once the heatsink and regulator are together insert and solder them into the circuit board.
Oh! you can now insert all of the chips into there correct chip holders.!!!
It is advisable that you check your work and all the components are where they should be and that there are no solder splashes or shorts underneath the circuit board. It is better to spend five minutes double checking everything rather than risk damage at switch on due to a mistake during assembly.

If you are sure everything is ok you can proceed to the setup and testing page.









Component list

COMPONENT	VALUE	MARKING / IDENTIFICATION
R1,7	470R	YELLOW,PURPLE,BLACK,BLACK,BROWN
R2,8	12K	BROWN,RED,BLACK,RED,BROWN
R3,5,9,11,27,30	3K3	ORANGE,ORANGE,BLACK,BROWN,BROWN
R4,10	100K	BROWN,BLACK,BLACK,ORANGE,BROWN
R6,R12,29	5K1	GREEN,BROWN,BLACK,BROWN,BROWN
R13,16	10R	BROWN,BLACK,BLACK,GOLD,BROWN
R14	330R	ORANGE,ORANGE,BLACK,BLACK,BROWN
R15	120R	BROWN,RED,BLACK,BLACK,BROWN
R17,23	10K	BROWN,BLACK,BLACK,RED,BROWN
R18,21	1M	BROWN,BLACK,BLACK,YELLOW,BROWN
R19,24	24K	RED,YELLOW,BLACK,RED,BROWN
R20	13K	BROWN,ORANGE,BLACK,RED,BROWN
R22,28	1K	BROWN,BLACK,BLACK,BROWN,BROWN
R25	47K	YELLOW,PURPLE,BLACK,RED,BROWN
R26	120K	BROWN,RED,BLACK,ORANGE,BROWN
R31	100R	BROWN,BLACK,BLACK,BROWN,BROWN
R32	3K6	ORANGE,BLUE,BLACK,BROWN,BROWN
R33	11K	BROWN,BROWN,BLACK,RED,BROWN
R34	75R	VIOLET,GREEN,BLACK,GOLD,BROWN
C1,2,5,6,10,12,13,16,17,20,23,25, 27,28,36,37,46	.1UF (100N)	104 OR 100N OR .1N (2.5 MM PITCH)
C3,14	4N7	4N7 OR 4700
C4,15	6N8	6N8 OR 6800
C7,18	27PF	27PF
C8,21,30,32,33,34	330PF	330P OR N33
C9,11,19,22,38,39,40,41,42,43	47UF	100UF
C26,35	12PF	12PF
C29	100N	100N OR .1N (5MM PITCH)
C31	150PF	150PF
C24	2.2UF	2.2UF
C44,45	39PF	39PF
VR1,VR2	10K POT	10K BLUE MINI POT
IC1,IC2	SSM2143	SSM2143
IC3,IC4,IC5,IC6,IC9,IC10	TLO71	TL071
IC7	DG201	DG201
IC8	PIC162XX	PIC16C62XX
FIL1,FIL2	15KHZ LOWPASS	5085
L1	3.9 MH INDUCTOR	392J
L2	4.7 MH INDUCTOR	472J
LED1,LED2	GREEN LEDS	GREEN LEDS
DC-DC CON	5V TO +/- 15VDC	NMH0515S
REG1	7805	7805
REG2	79L12	79L12
REG3	78L12	78L12
X1	4.864MHZ CRYSTAL	4.864
D1	1N4001	BLACK DIODE/SILVER BAND
CON1,2	3 PIN MOLEX SOCKET	3 PIN WHITE SOCKET
CON3,4	2 PIN MOLEX SOCKET	2 PIN SOCKET
J1,J2,J3,J4,J5	3 PIN JUMPER HEADER	3 PIN BLACK HEADER
HEATSINK	CLIP ON HEATSINK	SILVER CLIP ON METAL HEATSINK
8 PIN IC SOCKET X 8	8 PIN IC SOCKET	8 PIN IC SOCKET
16 PIN IC SOCKET X 1	16 PIN IC SOCKET	16 PIN IC SOCKET
18 PIN IC SOCKET X 1	18 PIN IC SOCKET	18 PIN IC SOCKET
PCB	BLACK BOARD	YOU ARE JOKING!

Setup, testing and operation of the DIGILOG.

The first step in the setup and testing of the digilog is to set all the jumpers as follows. Stereo jumper J5 (ST ON). Clipper jumpers J3 and J4 (clip ON). J1 and J2 (no jumper connected). Even though you may not want these settings when you come to use the unit, testing and level settings is easier and more correct when setup with them as indicated.set the Pilot level trim pot half way which will correspond to 9 percent of the maximum deviation. With the audio clipping activated the total maximum deviation permitted from the transmitter will be + / - 75 khz including the pilot.
Connect the stereo encoders multiplex output to the multiplex / audio input of your exciter / transmitter.
REMOVE OR BYPASS ANY PRE EMPHASIS IN THE EXCITER / TRANSMITTER.
The thing we have to do now is to set the input level of the exciter / transmitter so that

a). The pilot tone (19khz) alone and no audio gives a deviation of the exciter of 6.75 khz (9 percent)

or b). The total deviation with both channels and the pilot gives +/- 75 khz deviation of the exciter.

how you set it correctly which is important and the method (a or b) is down to whether you have the correct test equipment. To accurately set the level you will need either a peak deviation meter or a modulation meter/analyser. If you have or can gain access to one of these pieces of equipment then setting of the level is as easy as adjusting the input level of the transmitter for the appropriate deviation reading on the equipment for the (a) or (b) method used.

If you have the equipment you can skip the next section, Follow here if you are one of the many without access to the formentioned test equipment.

We will try to guide you into setting your level for the stereo encoder as accurately as possible. We will go into method (a) and (b) from above. You can try either or both but we recommend (a) as it is more likely to give accurate results, we will explain why in a second. You can confirm your settings by seeing if the setting you have set in for example (a) is the same setting when you try the method for (b).

Manual method for (a): Stereo decoder circuits in tuners have a tendency for the pilot light to activate and the pilot to be detected at a deviation of approx 3-4 khz deviation of the transmitter by the pilot. Knowing that allows us to judge if the pilot is the correct level (roughly).
To set up the encoder, disconnect any audio from the input sockets to the encoder and make sure the jumpers are set as per the top of this page.
Now turn on the transmitter and tune in on your tuner to the transmitters frequency. When you have done this adjust the transmitters mpx/audio input level downwards slowly until the pilot light goes out. This indicates to us that the encoder and transmitters level is nearly correct and the pilot is just under 3-4 khz deviation. If we now turn the level back up slowly until the pilot light comes back on we know that the pilot is deviating the transmitter by approx +/- 3-4 khz. Continue to turn up the level very slightly which we hope will give us about +/- 6-7 khz deviation. This is now correct and the transmitter’s total level will be correct when audio is applied.
If you now apply audio to the transmitters inputs and set the volume input pots for best sound you should have a total deviation of +/- 75 khz. Please note: the sound may sound a little distorted due to the clipping still in circuit. Do not worry we will explain that in a bit.

Manual method for (b): Method b is not as easy to setup. The way we setup the level in method b is simply to adjust the level of the transmitter until we get the correct volume to our ear. This is not so easy as you will be comparing your volume against other stations that have expensive broadcast processors or compressors which make them louder but still stay inside the legal +/- 75 khz. If you have a limiter or compressor you can put before the stereo encoder then it will help you gauge your volume more accurately. A station without a limiter/compressor maybe peaking a deviation of over +/- 200 khz and still not sound as loud as a commercial station which is broadcasting inside its legal limits of +/- 75 khz. For this reason we can see that the pilot method (a) is a more accurate guess method for setting the stereo encoders level without the aid of test equipment.

Hopefully you will have now set the transmitters audio input level correctly so that the stereo encoder has 6.75 khz of pilot and a total of +/- 75khz deviation with full audio (clip jumpers on).

Pre emphasis
The DIGILOG has active pre emphasis circuits capable of pre emphasis time constants of 50 us and 75 us as well as no pre empahsis by either the removal of the jumpers or by just hanging the jumper off of one of the pins of the jumper header.
Professional broadcast stations tend to disable the pre emphasis in the stereo encoder in favour of pre emphasis in the limiter / compressor or processor. This is done to get louder volume. If we had a standard limiter / compressor in front of the stereo encoder and we left the pre emphasis on the stereo encoder active we would have to set the transmitters deviation control so that the maximum deviation was +/- 75khz with the highest level possible of audio input. because the pre emphasis provides a 6 db per octave increase from the breakpoint. We can see that audio at 15khz will have much more level than lower frequencies. The limiter will limit all audio to a fixed level, regardless of frequency, but once it is through the pre emphasis circuit your volume level will follow the pre emphasis curve. Hence more volume at high frequencies than low frequencies, the effect, not much bass volume.

By having pre emphasis at the input to the limiter the maximum level applys to all frequencys enabling much louder volume. You might not have access to a limiter / compressor and one with pre emphasis capability so all this is irrelevant. We can get around the problem on the DIGILOG by clipping which is discussed next, but this is not really the right way to do it, but it works, albeit with a very slight degradation to the stereo seperation.

Audio clipping.
The DIGILOG has the facility for clipping of the audio prior to stereo generation.
Why clip! The main reason we included clipping on the DIGILOG was to get around the problems of ringing in the filters caused by the removal of certain frequnecies in the filters. Many compressor limiters have a peak limiter or clipper at their output to set the maximum output level from the unit. When this happens harmonics are introduced to the audio signal. When a complex waveform (audio is very complex) enters a filter, the filter removes the harmonics that are in the filters stopband. The removal of these harmonics will cause an increase in amplitide at the fundamental frequency concerned. Fourier analysis shows us that a squarewave (worst case clip of a sine) is made up of many sinewaves, all odd harmonics of the fundamental. The worst case situation with our 15khz brickwall filters is just above 5 khz, where a 5khz clipped waveform will have its 3rd harmonic and so on content removed by the filter. This causes an increase in amplitude of up to 3-4 db of the fundamental. From this we can see that in any country with strict broadcast regulations, the transmitter should not exceed +/- 75 khz deviation regardless of whether a sine or complex waveform is applied to the transmitter. You can see that a stereo encoder and transmitter set up for the correct deviation with a tone (sinewave) will have less deviation than the same coder and transmitter fed with squarewaves of the same level. The transmitter may need to be turned back to accomodate the increase in amplitude generated by the 15khz brickwall filtering converting the squarewaves and clipped audio to 15 khz band limited audio (no harmonics above 15khz). Where you could set up your system with your limiter to only allow +/- 75 khz of deviation, you try and now apply squarewaves to the input of the system and sweep tthrough the frequencies. Your 75khz can easily double. To get around the problems of audio filtering the overshoot compensated filter was designed. This technique turned overshoots back on themselves correcting the overshoots and maintaining a constant 15 khz band limited output level regardless of how complex the input waveform was. This is outside the scope of this unit and so won't be disscussd further. This is where clipping comes in. If we accept that clipped audio is quite random and peaks and hence filter overshoots are quite irregular we can accept clipping as an alternative to control deviation. In our tests we have found that 3 db of clipping will not noticably affect the stereo seperation,audio clarity or the cleanliness of the output spectrum of the DIGILOG. This is only the case if the clipping is employed lighlty so that no more than 3 db or so of clipping is active. Any more than this by overdriving the clippers will cause excessive distortrion, slight loss of seperation and the output spectrum to be compromised. If you keep this in mind and use the clippers as a way of getting around overshoots you will be ok.
The clipping function also allows within reason the ability to obatin more loudness. Measure up loudness, against distortion and out of band harmonics. Earlier we referred to clipping aliasing. Aliasing makes a signal appear to be there that is not. If we were to clip a 8 khz signal and produce a harmonic of it at 24 khz, the 24 khz signal would be in the L-R band. because the way the DSSC L-R signal is composed the 24khz signal looks like an 14 khz signal. Remember a 14 khz signal double sideband supressed carriered around 38 khz would have sidebands at 38khz +/- 14 khz which = 24khz and 52 khz. So a clipped 8 khz signal that has a harmonic at 24khz will come out as a 14 khz alias when the stereo is de multiplexed. We have just generated a mystyrious 14 khz signal from no where. That is Aliasing........
Do you know that expensive top processors such as the cutting edge omnia employ clipping to the stereo multiplex in basically the same way and have exactly the same problems and benifits. They have the usual X band limiter compressor stuff but clip at the stereo muliplex generation to significantly increase the volume. They still suffer from loss of seperation, distrortion and output spectrum cleainliness when too much clipping is active.
If you have an excellent compressor limiter that has brickwall filtering internal then there will be no benifits of clipping on the stereo coder as there will be no overshoots in the stereo encoders brickwall filtering. The question you are asking is should i activate clipping or not! there is no easy answer. Try to digest the last page or so and make your own mind up, if in doubt, leave it out.

Volume pot's and the volume input.
the DIGILOG has balanced input stages, this is prefered in professional applications due to the large common mode rejection (rejection of hum and noise pick up in the cables). To take advantage of the balanced system you will need a source (mixer, limiter etc) with balanced outputs, nearly all audio equipment that is even semi professional has a balanced output (XLR or 1/4 inch jack).
The balanced input level are fed to volume pots which can be adjusted to allow a large range of input signals. If you have an unbalanced output from your equipment (phono RCA) then you simply connect to the cold pin of the connecter with your live of your unbalanced feed. the earth connects to the gnd of the input connecters as per normal. Due to the fact that a balanced line has twice the level of an unbalanced line (due to an out of phase signal on the opposite leg) you will experience half the gain with an unbalanced line. This can be overcome by shorting together two pads on the board marked UB with solder. This has the effect of forcing the balanced input chip into unbalanced mode keeping the circuit gain's the same as a balanced feed.

DIGILOG STEREO ENCODER