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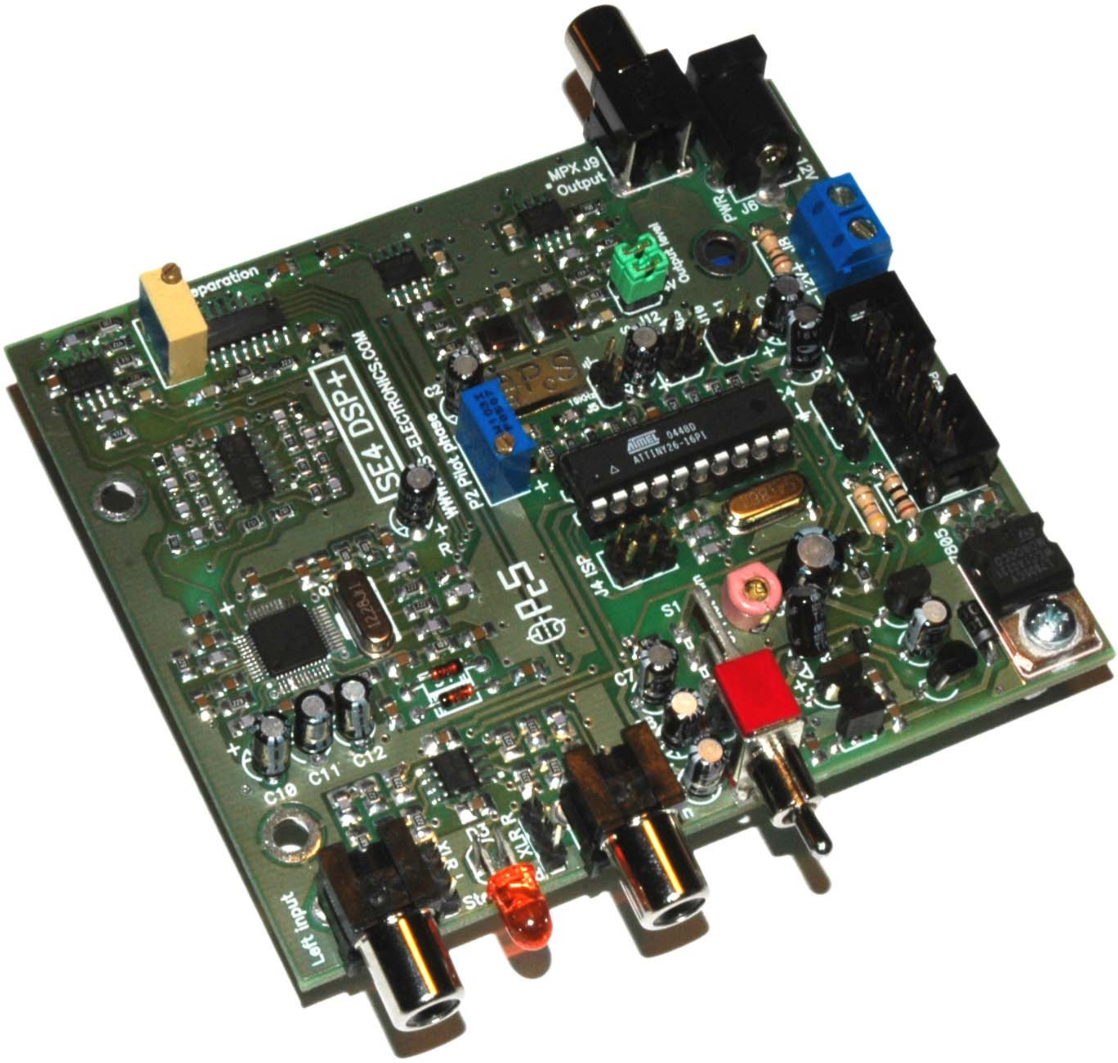
SE4 DSP +

High Performance Professional Digital Stereo Encoder With DSP Filters



SE4 DSP + without the LCD control module (connects to black IDC connector on the right)

SE4 DSP + is a high performance stereo encoder with DSP technology. DSP audio processor makes this unit stand out from the rest. It is perfect for a demanding, but cost-conscious broadcasters. Compressor, limiter, pre-emphasis circuit, very sharp low pass filter, balanced audio inputs and sharp 19KHz stereo carrier notch filter make sure your signal stays where you want it, providing high quality audio with excellent channel separation without causing interference to nearby channels. High quality components and printed circuit board assure 24/7 operation for years.

Why is SE4 DSP + so great?
- Perfect for any mono FM transmitter, effective output level of 1V, 2V and 4V can be selected with jumpers
- DSP technology enables extremely sharp filters and a very deep notch at 19KHz!
- Small form factor!
- Drop-in replacement for SE3 and SE4! Same mounting holes as SE3 and SE4, only slightly longer!
- On-board DC converter.
- Balanced or unbalanced audio inputs. This effectively eliminates annoying ground loops and hum.
- LC filtered MPX output signal.
- Built-in limiter, low pass filter and compressor/limiter, all controlled via backlit LCD display and keys. Same LCD unit can control MAX PRO 3 and SE4 DSP+ at the same time, if you connect the two units with the I2C cable!

How is SE4 DSP+ better than SE4?
- DC converter is not synchronized with stereo pilot
- Additional anti-aliasing low pass filter enhances signal/noise ratio
- Improvements for better stereo separation
- Improved board layout
- Reduced heat generation (7805) by about 50%.

Technical specifications:
Audio Response: 10Hz-15KHz, DSP filtered (Standards require upper level at 15KHz)
DSP: 24-bit Digital Signal Processor
19KHz notch filter, >-70 dB typ
Precise pre-emphasis, 50uS, 75uS or none, selectable via LCD/keys
Audio Input Impedance: 600 Ohms, balanced or unbalanced (can be raised to 10K, if necessary)
Audio Input Level: 0 dB
Distortion: <0.1%
S/N ratio: -75 dBm
Separation: >60 dB typ. (more than any radio receiver you can buy on the market)
Pilot Frequency: 19 KHz, DSP generated with 32x oversampling
Output Impedance: 75 Ohms
Power Requirements: 12-15VDC / 100mA
PC Board Size: 100x85mm
Audio connectors: all RCA jacks are mounted on the board, 3-pin jumper for XLR balanced input
Power connector: 2.1mm power socket, center is positive, connection terminals (easier for boxed units)
Output level: Ueff 4V, 2V or 1V

What do the jumpers and controls do?
J1, J2: Audio input
J3: I2C, connect to MAX PRO 3 or MAX PRO 4 to control both units via same LCD control module
J4: For internal testing, not to be used
J5: Output 19KHz pilot, for RDS or other use
J6: Power supply socket, 12-15V/100mA (13.8V is perfect)
J7: LCD control module, connect here
J8: You can insert power supply wires here (perfect for boxed units with integrated transmitter)
J9: MPX audio input: Connect to transmitter
J10, J11, J12: Not used
J13: Depends on desired output voltage level
-For 1V install both jumpers
-For 2V install only 2V
-For 4V remove both jumpers
S1: Stereo/mono selection. LED diode illuminates in stereo mode and it is shown on LCD display.
Trimmer capacitor: If you want to set pilot frequency precisely to 19 KHz, use this trimmer. For advanced demanding users only, others do not touch this.

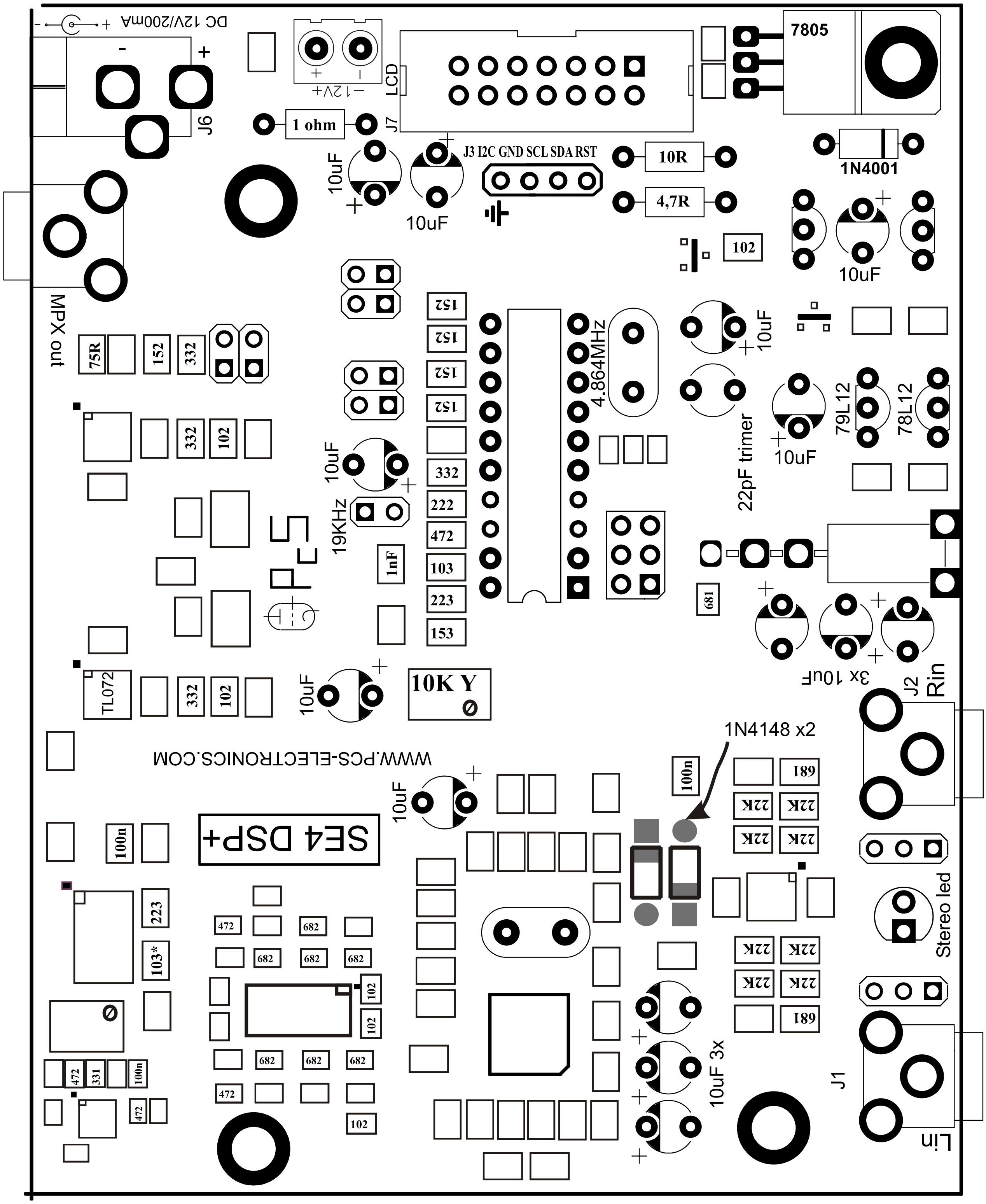


Figure 1: *SE4DSP + board layout*

INTRODUCTION - PRINCIPLES OF OPERATION

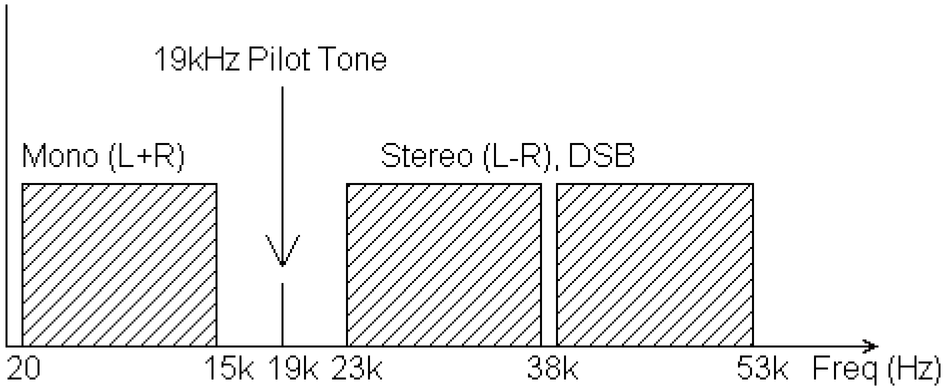


Figure 2: *Theoretical frequency spectrum of the stereo multiplexed signal*

Figure 2 above shows the theoretical frequency spectrum of the stereo multiplex signal (MPX-signal). The MONO signal on the far left goes from approx. 20Hz to 15KHz and is used to transmit the sum of both the left and right channel. This assures compatibility with older MONO receivers that only receive this part of the spectrum. Going from left to the right we stumble upon the 19 KHz pilot just above the MONO signal. This pilot has a couple of functions;

1.) It signals presence of the stereo signal; by detecting it the receiver switches to stereo
2.) It enables demodulation of the L-R signal and LEFT/RIGHT channel reconstruction

The 19 KHz signal is used to demodulate the DSB (Double Side Band Suppressed Carrier) signal stretching from 23 KHz to 53 KHz. This signal contains the L-R information (difference between the left and right audio channel). This is what the stereo encoder does to generate the Stereo Multiplex signal.

A.) Add Left and Right signals to get an L+R signal.
B.) Generate a Pilot Tone of 19 KHz.
C.) Generate a 38 KHz carrier for the Doubly Balanced Mixer (DBM)
D.) Generate the L-R (difference of the audio channels) signal for the DBM
E.) Modulate the 38 KHz carrier with the L-R signal using DBM (DBM suppresses the carrier in the process)
F.) Add up A, B and C above to get the complete MPX Signal.
G.) Use the above MPX signal to Frequency Modulate a carrier in the 87.5-108 MHz band.

SOME FACTS ABOUT STEREO

Even the best stereo encoder is by itself not enough to guarantee good channel separation at the receiving side over the whole audio frequency range. Many factors are involved:

<u>THE TRANSMITTER</u>
The first problems usually occur at the transmitter. Badly designed audio stages of the modulator will produce low frequency phase shifts, affecting separation. But the main problem is the phase locked loop section of the transmitter. PLL tries to correct the frequency deviations caused by the audio effectively canceling modulation. The frequency correcting signal is passed through a low pass filter (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. Variable frequency oscillators do not suffer from the problem at all due to no frequency correcting circuits (PLL). 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.

THE RECEIVER

Filter Bandwidth and Stereo Decoder of a receiver. Even if the transmitter adds no phase shifts to the multiplex signal transmitted, the receiver (radio) at the listening end can still cause trouble. The filters in the radio can cause phase shifts to the multiplex if too narrow in bandwidth. Many cheaper tuners have less filtering (less manufacturing cost) which although not great for selectivity provides for excellent separation in strong signal environments. The above is only true if the stereo decoder in the radio or tuner is ok. It is very hard to obtain any modern stereo decoder chips that give more than 45 db of separation, some give only 35 db. So even with modern day DSP (digital signal processor) stereo encoders that achieve separations 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. As you see, stereo is not just about a stereo encoder!

CIRCUIT DESCRIPTION

Left and Right audio signals are applied to the connectors J1 and J2. Make sure not to ground the outer shield of the RCA connectors, this will help reject the noise on your audio lines. Alternatively balanced inputs can be used. Connect them to Rin and Lin 3-pin jumpers. The audio signals are fed from here into the balanced-to-unbalanced converter and passed over to DSP processor circuit. 24-bit DSP processor takes care of signal processing for us formed by op amps IC1 A and B. 50uS pre-emphasis is default, with 75uS possible with a change of two resistors (check component list). The outputs of the pre-emphasis circuits are then fed into the 15 KHz low pass filter to remove any high content above 15 KHz. This is a basic filter, better results can be obtained by using an external low pass filter and compressor/limiter. The filters are fed into switch (IC5), operated by the microcontroller (IC4). The output from the switch (mixer) is fed to the output buffer. Microcontroller also generates the pilot tone (19 KHz) via the D/A converter. All these signals (DSB and pilot) are summed up and buffered with IC3. The resulting MPX signal is then filtered and buffered (IC4).

HERE IS WHAT YOU NEED TO USE SE4 DSP+:

POWER SUPPLY

This unit is designed to work with a wall-wart that gives 12-15V at approx. 200mA, provided it has a good smoothing cap. You can connect the DC supply by inserting the power jack into provided socket.

ENCLOSURE

If you want to make your own, use aluminum or other metal, ventilation holes are recommended. The 7805 regulator needs to be bolted to the enclosure via provided spacer as it does get quite hot. Fix the PCB with all screws tightly. A shield is recommended between the exciter and the encoder, if you have them both in the same enclosure. Attractive and predrilled enclosures of exact size are available, check our site for info.
A 19” rack enclosure for SE4 DPS+ is available from our website.

SETUP AND TESTING

SE4 DSP+ is very easy to setup. What we do have to do however is match the output level of the encoder and input level of the transmitter so that the pilot tone (19 kHz) alone (no audio) gives a deviation of the exciter of 6.75 kHz (9 percent). This automatically sets the remaining audio levels. If you’re using our line of FM exciters just connect the stereo encoder to the transmitter, set encoder to Stereo, set audio level on the fm transmitter to zero and keep increasing it until the stereo led on the receiver comes on.

Let’s assume that you don’t own an expensive peak deviation meter or modulation meter/analyzers. If you have or can gain access to these pieces of equipment then you probably also know how to use them; setting of the level is as easy as adjusting the input level of the transmitter for the appropriate deviation.
To set up the encoder, disconnect audio from the input sockets on the encoder. Adjust the modulation level at the exciter so that the LED diode lights on the receiver. The next step in setting up the encoder is to optimize the stereo separation by adjusting the trimmer. The setting for this will vary from tuner to tuner slightly. To set this follow the procedure below:
1.) Disconnect one audio input source so that only one channel is connected. Apply audio to this source.
2.) Listen to the audio on a high-grade tuner and adjust the input volume pot for that audio channel so that the volume is only half that of a commercial station. The reason we want this is to be sure we are inside the +/— 75 kHz bandwidth. Over deviation will cause degradation of the stereo separation. We now should have the encoder

correctly setup with only one channel of audio that is inside the +/-75 kHz bandwidth so separation should be able to be fine tuned without problems such as over deviation affecting our measurements. Turn your amplifiers balance control so that you are listening to only the channel with no audio on.
If everything is good and well then you should have this channel a lot quieter than the other channel. Turn the amplifier up in volume so you can hear the crosstalk between the channels. Now adjust P1 and P2 until the sound in the opposite channel disappears or is at least barely noticeable. You should be able to achieve your maximum separation. You can now reconnect the other channel and apply your audio at the correct level. The encoder is now aligned and ready for operation. DO NOT FORGET TO DISABLE PREEMPHASIS AT THE TRANSMITTER WHEN YOU CONNECT IT TO THE STEREO ENCODER (failure to do so results in erratic operation, I my-self forgot to remove the jumper once and spent hours fixing the damn thing).

BALANCED AUDIO AND POWER CONNECTOR

SE4 DSP+ features balanced audio inputs, just connect XLR connector to Lin or Rin. Pin 1 is ground, the other two are Audio + and Audio -. Any hum problems usually magically disappear once the XLR input is used instead of the basic unbalanced RCA input. Note that you will have to purchase XLR connectors as only RCA type is included on the PCB

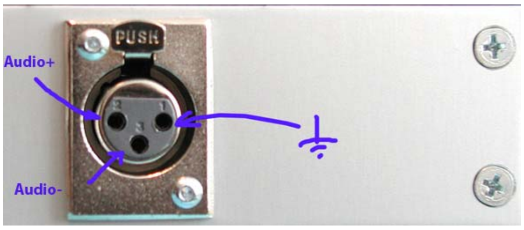


Figure 3: *Balanced audio input – XLR input*

You will also find the usual power socket (center is positive) at the side of the unit. Use either our 15V mains power supply or another power supply with appropriate ratings. See product specifications for more details. Next to the power socket is the IDC14 connector for optional LCD control unit.

Notice stereo/mono switch in the middle. Unit is in stereo mode whenever you see LED illuminated.

LCD CONTROL MODULE MENU SYSTEM

The front of the unit starts with the three keys on the left, followed by the backlit LCD display. LCD control module is equipped with our new menu system. It can be modified on request to include your call sign or any other messages you want displayed on the LCD.

The UP and DOWN keys are used to change parameter value. In normal mode the LCD is simply showing the welcome screen. Menu key can be used to enter the menu mode, repeatedly pressing this key brings up the following menus: TREBLE, BASS, COMPRESSION RATIO, COMPRESSION THRESHOLD, ATTACK, DECAY, INTEGRATION INTERVAL and PREEMPHASIS. Pressing the UP or DOWN key selects the desired parameter and allows you to modify its value. Another press on the MENU key and you're back to normal mode.

TREBLE and BASS

This setting allows you to set amount of treble and BASS in your audio. Recommended value would be about 0dB.

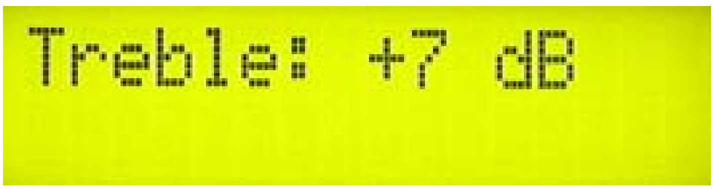


Figure 4: *Setting the treble, one of the many menu settings*

COMPRESSOR SETTINGS

A number of MENU settings control the operation of the compressor. Lets assume that the audio signal enters the transmitter at some low level. Compressor does nothing to the signal until at one point as the input signal increases the signal reaches the compression threshold. Digital signal processor starts compressing the signal beyond that point. The higher the compression ratio the higher the compression. For example, compression ratio of 1:∞ would in effect be a limiter.

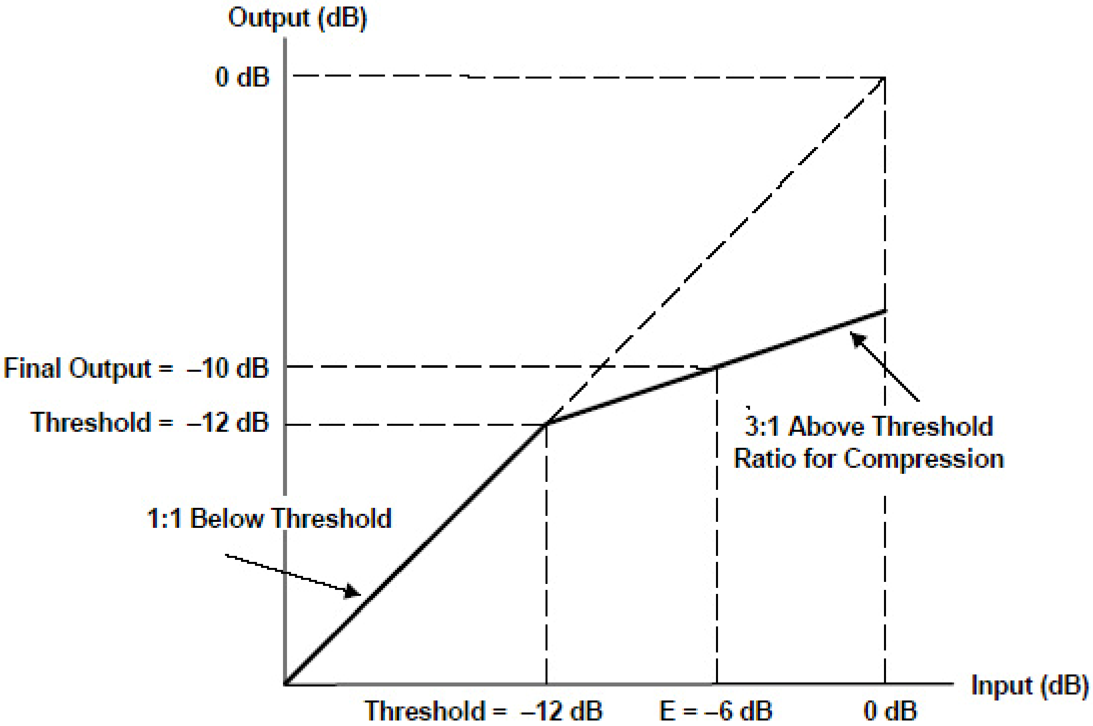


Figure 5: *Explanation of the compressor settings*

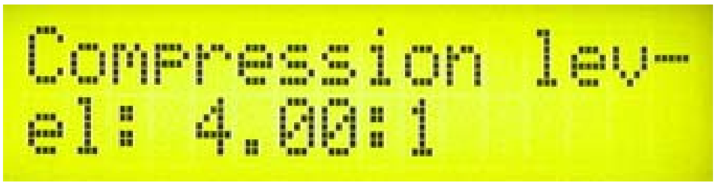


Figure 6: *Setting the compression level, levels between 1:4.00 to 1:8.00 are recommended*

Figure 7: *Setting the compression threshold, -12dB is recommended*

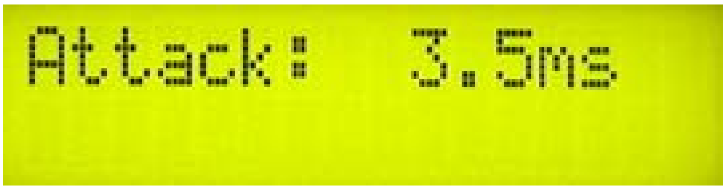


Figure 8: *Setting the attack time, this is the time between the signal rise and the actual response of the compressor*

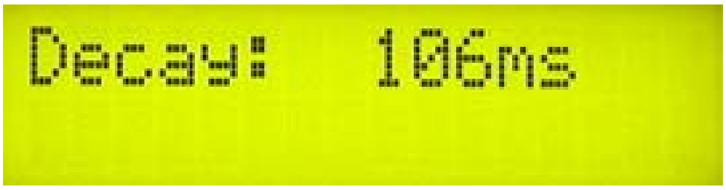


Figure 9: *Setting the decay time, this is the time the compressor needs to respond to a decrease of the signal*

Figure 10: *Setting the integration interval, this is the time the DSP extracts the samples*

PRE-EMPHASIS

It is possible to adjust the pre-emphasis of the transmitter to either 50uS (standard for EU and most of the world) or 75uS (United states and Canada).

LEGAL CONCERNS

If you have any legal questions concerning your SE4 DSP+ or any device it is your responsibility to study the regulations. It is best if you personally read the rules (and consult with a lawyer if you're in doubt). It is up to you to operate within local rules and PCS Elektronik d.o.o. cannot be held responsible for any violation thereof.

THANK YOU FOR PURCHASING SE4 DSP!

We hope you will enjoy it as much as we do and remember to tell your friends about it. Please feel free to leave your comments at our website or post your experience in our forum.

From all of us we wish you happy broadcasting!

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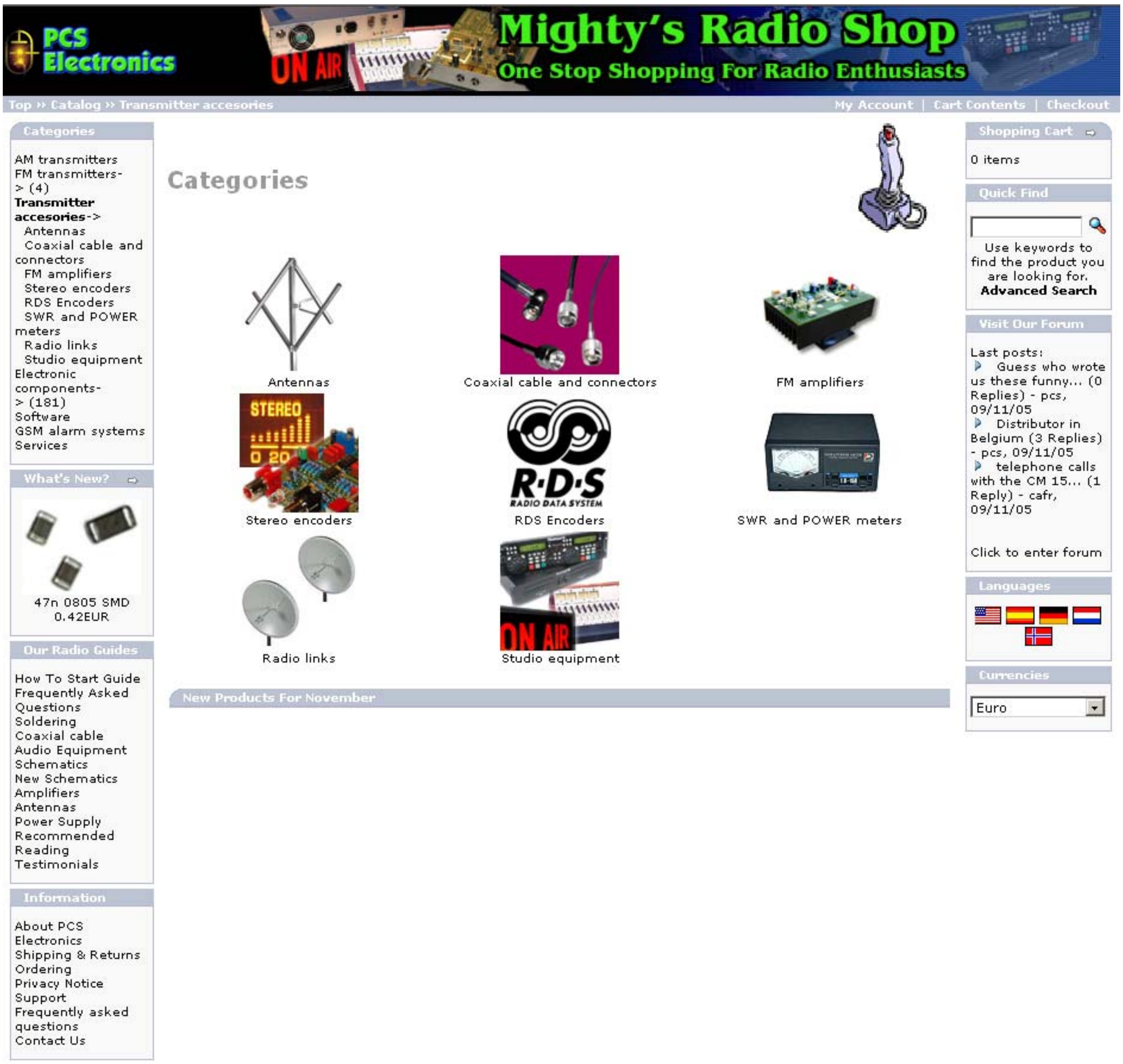

Coming soon (End of November 2005 to start of December 2005)