Juggling Audio Bits
Audio DSP for DIY applications

By Harry Baggen (Elektor Netherlands Editorial)

Audio hobbyists usually confine their hobby to the analogue domain, since the opportunities for doing your own experimenting in the digital domain are very limited. There is very little affordable equipment available that allows a multitude of digital-audio processing operations and getting started with DSPs yourself requires a considerable depth of knowledge of this subject matter. With the modules from miniDSP you can easily realise all kinds of audio processing functions without the need to become intimately familiar with digital signal processing.

In the audio world, DIY is most certainly not dead yet. There are still plenty of people who design/build amplifiers and speaker boxes or modify existing equipment. Despite the fact that the source material (the music) and the playback equipment are usually digital, there are still sufficient opportunities for tinkering to your heart's content with the analogue stages that follow. Nevertheless, you will have noticed that more and more audio processing is taking place in the digital domain, ranging from signal processors to digital power amplifiers. This is quite a complicated world for the average audio hobbyist, where there are now few or no opportunities for doing anything yourself. Still, it is becoming increasingly interesting to familiarise yourself with the possibilities that these digital technologies have to offer. A digital crossover filter for an active loudspeaker system comes to mind, which would allow easy adjustment of the crossover frequency, slope, filter type and delay between the drivers.

It is true that there are a few systems on the market which offer you the possibility of experimenting yourself (for example the amplifier module AS2.100 from Hypex, which contains two digital power stages and a DSP for filtering and equalisation, or the semi-professional DCX2496), but overall the choice is very limited.
The young company miniDSP, established in Hong Kong, comprises several engineers who came up with the idea of marketing flexible and affordable digital audio products for both hobbyists and professional audio equipment manufacturers. The objective here was to keep the programmability of these products as simple as possible. In practice this means that you buy a board with a DSP on it and then you choose the required functionality from the software applications that they offer. This software is configured using a comfortable user-interface on your PC. So you only have to type in a crossover frequency and a slope for the filter and the DSP will do exactly what you have asked it to do. There is no need for extensive knowledge of DSPs and programming, and (almost) none for digital audio technology.
The basis of the entire system is the miniDSP board, a printed circuit board measuring 7.5 by 7.5 cm, which contains a DSP and a number of connectors. To this you can add an I/O-board, which has a number of digital inputs and outputs (miniDIGI) and a board with four digital power stages (miniAMP). All boards have identical dimensions and can easily be stacked together and interconnected. Using the so-called audio plug-ins you can determine the functionality of the DSP board, that is, the software you put into the DSP. At the moment, plug-ins are available for 2- and 4-way crossover filters with equaliser (in simple and advanced implementations) and a mixer with 31-band graphic equaliser. Work is in progress for other plug-ins.
The plug-ins work in combination with Adobe Air, which you have to install on your computer beforehand. Double-clicking the downloaded plug-in is then sufficient to install it.

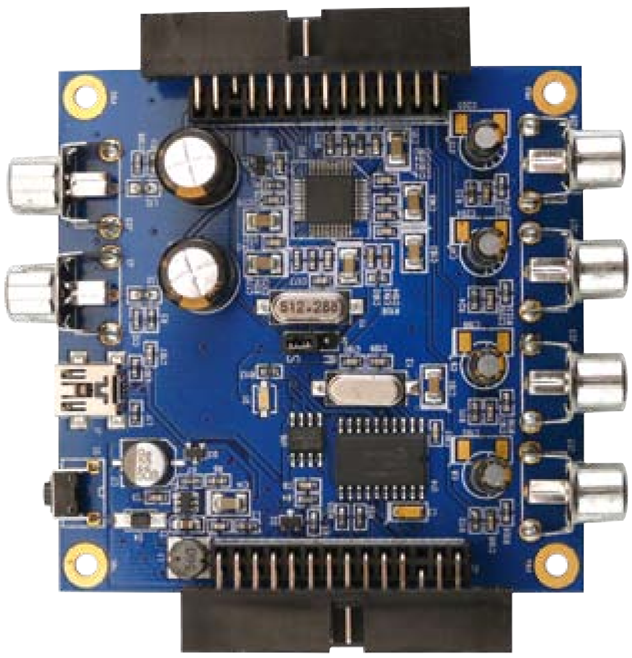
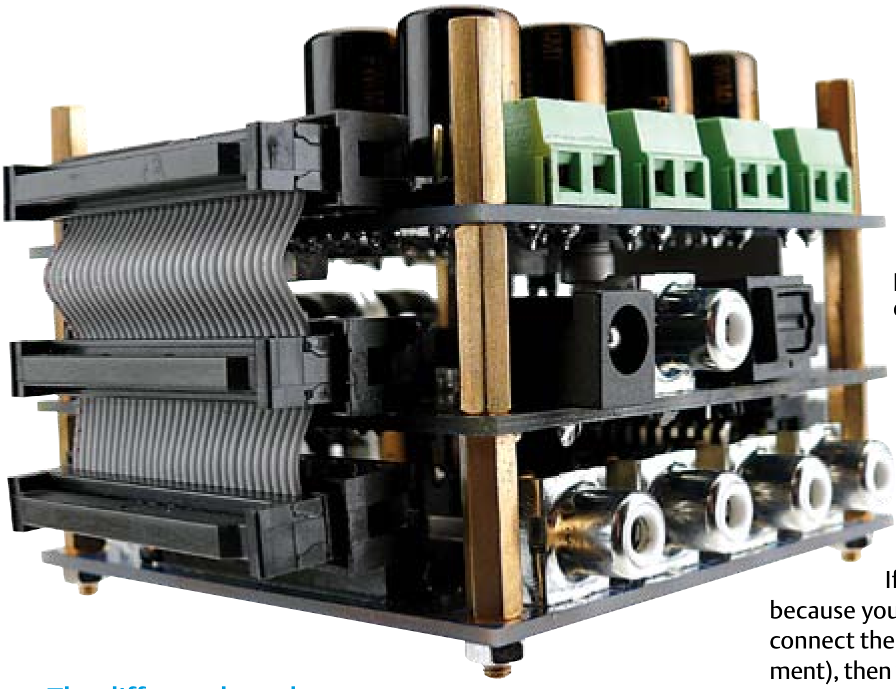


Figure 1. The miniDSP board has two analogue inputs and four outputs.



The different boards

First, let's take a look at the hardware. At the core of the miniDSP board (Figure 1) is a ADAU1701 from Analog Devices. This processor was developer specifically for audio applications. Most of the calculations are carried out in 56-bit double precision mode for an accurate result. The IC, in addition to the DSP, also contains 24-bit A/D- and D/A-converters which operate according to the sigma-delta principle, which ensures a large dynamic range. The program for the DSP is automatically read from the serial EEPROM on the board when the power supply it turned on. In addition to the DSP and EEPROM the miniDSP board also contains a PIC18F14K50, which takes care of the communications between the computer and DSP, using a USB-interface. The board can be easily connected to other audio components using the RCA connectors (2 inputs, 4 outputs). The board uses audiophile quality electrolytic capacitors from Nichicon for a minimal influence on the analogue signals. On each side of the board there is a header for interconnecting all the relevant signals to other miniDSP boards. There is also the option of connecting a potentiometer, which can be used to control the volume in the DSP.

If you require digital inputs or outputs (for example, because you want to go directly from a CD player, or you want to connect the processed signal in digital form to some other equipment), then your needs are met with the miniDIGI board (Figure 2). Here you will find two coaxial and two digital inputs and a coaxial and optical output. This board is available with and without transformers for galvanic isolation of the coax connections. The heart of this board consists of a sample-rate converter (SRC43821 from TI) which will convert all sample rates up to 216 kHz to 48 kHz, which is the operating sample rate of the DSP. Using a jumper you can select which of the 4 inputs is used and another jumper is used to select which signal is routed to the digital outputs (for example the input signal or the DSP processed signal). The board can also be used to de-jitter an S/PDIF-input signal. The last of the currently available boards is the miniAMP (Figure 3). This board contains

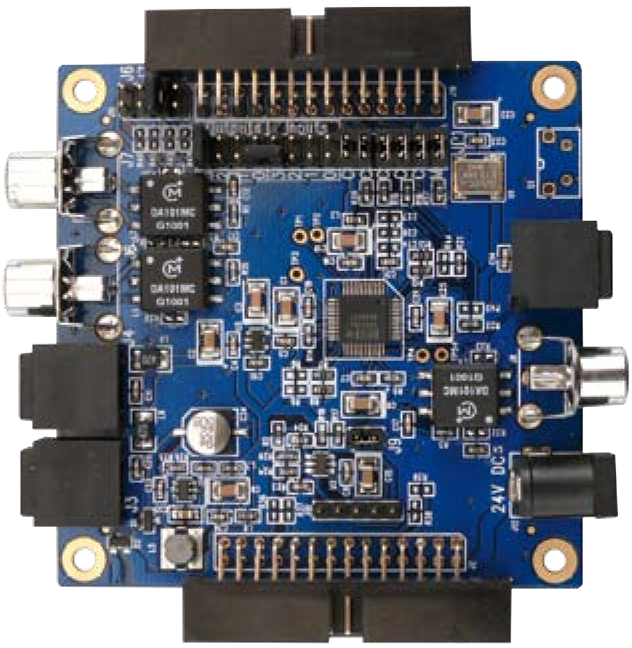


Figure 2. The miniDIGI board contains a sample rate converter and offers a substantial number of inputs and outputs.

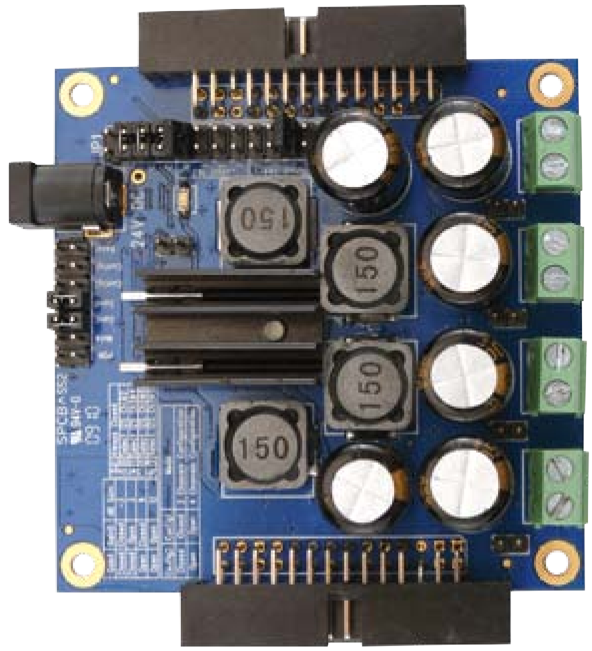


Figure 3. The third board, the miniAMP, contains a class-D amplifier-IC with four output channels, which can also be bridge connected.

Figure 4. The design of the software. From this overview you can select each function block.

Figure 5. Here we can see the part for the crossover filter. You can select the crossover frequency, type of filter and slope. At the same time you can choose a band-pass filter.

four digital amplifiers, which can be bridge-connected for a stereo version with a higher output power. The power amplifier IC on this board is a TAS5704 (also from TI). This can generate an audio power of 4 x 10 W into 4 Ω or 2 x 20 W into 8 Ω in bridge configuration. Thanks to the class-D design of this amplifier it has a high efficiency (90%) and a small heatsink mounted directly on the PCB suffices for the cooling. The electrolytic capacitors are made by Nichicon, but these capacitors are not necessary in bridge configuration and can be linked out using jumpers. The TAS5704 is presented with the audio signal in digital form, so the D/A converters of the DSP are therefore not used. Although the output power of this board is not all that high, it is ideal for experimenting because you have 4 power amplifiers at your disposal. Just connect the drivers via a few cables to the screw terminals and you can immediately evaluate the effect on the sound of any of the DSP settings that you make.
The printed circuit boards all have identical dimensions and can be assembled one on top of the other with stand-offs. Short ribbon-cable interconnects are supplied to make the connections between the boards. The power supply for the miniDSP board an be provided by the USB connector mounted on the board. The miniDIGI board and miniAMP board both contain a standard DC power supply jack for connecting a mains power supply. The power supply voltage may range from 4.5 to 24 V for the miniDSP and miniDIGI, the miniAMP board requires 12 to 24 V.

Connecting and testing

For this test we received a complete set, comprising a miniDSP, miniDIGI and miniAMP. We selected a miniDSP version with an input range of 2 V, so that you can directly connect the output from a CD player to it (there is also a version with a 0.9 V sensitivity available, to which you could easily add a voltage divider, if need be). The boards are connected together using very short interconnecting cables (see opening photo). There are plenty of possibilities with this complete set of boards, but in practice you would only buy those boards that you need for your particular application. If, for example, you want to add a digital crossover filter to a loudspeaker box with analogue inputs and use existing power amplifiers then you will only need to buy a miniDSP board. If you would like digital inputs with that then you will also have to buy the miniDIGI board. And should you want to experiment regularly with several different loudspeaker boxes, then the miniAMP board could be a very handy accessory.

The boards are supplied completely assembled. You effectively only have to connect a power supply and you are ready to get started. Well, that is after you have set all the jumpers on all the boards appropriately. In particular the miniDIGI board contains a large number of jumper settings, among others, for the correct interconnections of the I2S signals between the boards. This can be quite confusing initially. Fortunately the miniDSP website contains documentation with various example configurations, which show exactly which jumpers have to be placed where for a certain combination of boards or application. In addition, the developers have advised us that they are continually busy with further improvements and expansion of the documentation. Even during the time we were doing our evaluation there appeared several updates to the software and documentation.

For the software-example for this test we choose a two-way filter with built-in parametric equalisers. This is an excellent test project for such a system.

In the first instance we connected the entire module to our Audio Precision System II to check the operation and perform a few measurements. The module was powered from a large regulated mains power supply (power supply voltage was 24 V).

To get a feeling for the affect that the DSP has on the analogue audio signal, all the filters in the software were set to ‘straight through’ and we supplied a digital signal via the miniDIGI board. We then measured the distortion of all the analogue outputs of the DSP. These were very low, about 0.005% at 1 kHz at nearly 100% signal level. This value changed very little when we switched to the analogue inputs of the DSP, that is, the A/D converters are now also

Figure 6. Every parametric equaliser of the two-way plug-in allows a maximum of six corrections (frequency, value, Q factor). Here a simple bass-boost has been realised.

in the signal path. The signal-to-noise ratio is at more than 90 dB (linear), which corresponds to the value that AD specifies for the ADAU1701.
The class-D amplifier on the miniAMP board operates at a frequency of about 400 kHz (changes slightly based on the applied sample frequency). The residual of this frequency at the outputs has an acceptable magnitude of about 100 mVp. The harmonic distortion in single-ended configuration amounts to about 0.07% at 100 Hz and 1 kHz/1 W/8 Ω. The output power at 1% THD and 24 V power supply voltage turns out to be 14.5 W into 4 Ω and 8 W into 8 Ω. This is very close to the specifications from TI. We didn't measure the bridge configuration, but this will certainly deliver 20 W into 8 Ω, as reported in the data sheet for the miniAMP.

In practice

After all this it was time to play with the software and have a look at how we could realise a practical circuit. After installing Adobe Air runtime (free download from the Adobe website) and downloading the two-way crossover plug-in from the miniDSP website, you only need to double-click the downloaded .air-file after which the installation is automatically carried out and shortcuts are also created. Now we only need to connect the DSP-board, using a cable with a mini-USB connector, to the PC and we are ready to start experimenting.

When the program is first started the tabbed page 'Audio Settings' appears, which shows a block-diagrammatic overview of the design of the software. The tabbed page 'System Settings' contains a number of general settings, such as the type of input signal (analogue or digital), activating an external volume control, and saving and loading of the configurations (in xml-format). In this way you can save all the settings that you have made and easily retrieve them at some later time. There is also a button here to return everything to the default settings. Finally there is a third tab that brings you to the miniDSP website. All plug-ins follow the same general design.
When clicking on one of the function blocks on the 'Audio Settings' page, the corresponding settings appear. With the two-way crossover plug-in you can select in the first block the attenuation of the input signal. This is followed by a parametric equaliser for each channel with up to 6 configuration settings. Subsequently we arrive at the crossover part where for each driver you can select the crossover frequency, type of filter and the filter slope. The available filter types are Bessel, Linkwitz-Riley (12, 24 and 48 dB/octave) and Butterworth (6 to 48 dB/octave). You can also expand each filter to a band-pass type, for example to protect a small woofer from frequencies that are too low or to combine a two-way system with an already existing active sub-woofer. This section is then followed by the 6-band parametric equaliser, which allows individual corrections to be made for each driver. In the final block you can set the attenuation and time delay for each driver. The latter is particularly important with the Linkwitz-Riley filter type because the woofer and tweeter need to have their acoustic centres perfectly above each other. The correction has quite a wide range, up to 7.5 ms, that is equivalent to more than 2.5 m.
Once all the necessary settings have been made you can click the green 'Synchronize' button and the software will make the connection with the (via the USB-cable connected) miniDSP board. The settings you made are then sent to and stored on the miniDSP board. The firmware for the DSP is also automatically updated, if that is necessary. The green button disappears after synchronising, which appears a little strange at first. Is no further synchronisation possible? There is. After synchronising, all the changes you now make are immediately transmitted to the miniDSP board, in other words, you have a 'live' connection. So you can experiment to your heart's content and change all kinds of things. You can, for example, connect a music signal and hear immediately what the effect of each individual change is on the sound image. So while experimenting with our two-way speaker box we could very easily change the level of the tweeter while listening to it and even play with the time delay between woofer and tweeter. A fantastic opportunity for experimenting with DIY loudspeaker systems! When the program on the PC is closed the last settings are automatically stored in the memory on the miniDSP board.
When you're done experimenting, you can mount all the boards in the loudspeaker enclosure, together with a suitable power supply and power amplifier. As already mentioned, the miniAMP is very handy for experiments, but for a final loudspeaker system it would be better to choose a few bigger (and better quality) power amplifiers.

Very handy

With the miniDSP-system you can in a very simple manner and a very short time assemble an audio circuit which would be much more difficult to realise using analogue components. Consider an active crossover filter with delay time correction and a few frequency corrections. With this system it is done in the blink of an eye. You do not require any knowledge of DSPs, but you will need, of course, knowledge and experience in the area of filter and loudspeaker technology. But this you will need anyway, if you are going to develop your own loudspeaker systems. The method of directly changing the settings from your computer works really well and

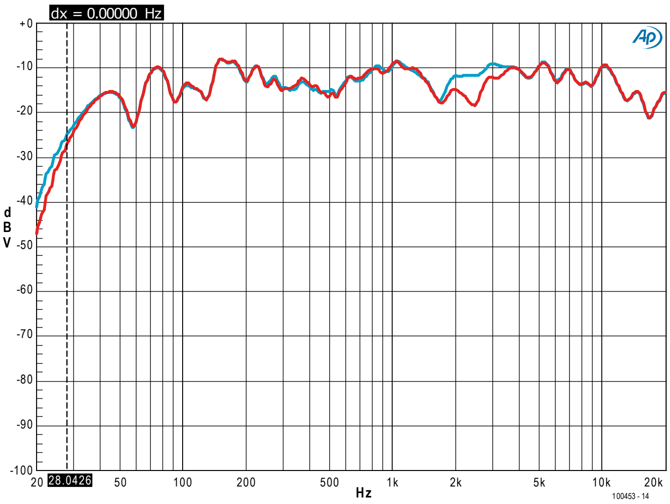


Figure 7. MLS-measurement where the time delay between woofer and tweeter was changed in small steps until the response around the crossover frequency and time delay was measured, Red = no time delay, blue = 0.2 ms time delay, about 7 cm.

even encourages you try out many more things than you were perhaps originally planning to do. After all the experimenting you can simply build a final version by building the miniDSP board with power supply and suitable power amplifiers in a loudspeaker box. And, if after a while you decide that these settings are not quite ideal, then it is very quick to connect a PC and make the necessary corrections.

Despite the extensive capabilities of the miniDSP system it is very affordable. For most applications you will only need the miniDSP board and you're done for $99 plus $10 for the plug-in. One board on its own is capable of driving two two-way speaker boxes. However in most cases it will be more convenient to build a miniDSP board into each speaker box, in addition, the board has sufficient computing power to implement a four-way filter.
The audio quality of the miniDSP board is quite good, but we can imagine that some audio enthusiasts are not satisfied with this and would prefer to use other A/D- and D/A-converters than those that are built into the DSP. This is also a possibility with this board, by using the I2S-in- and outputs on the miniDSP board. And there are many more adjustments and options... The most important thing is that with these boards you can very easily make a start with digital audio processing. And all that for a very attractive price!

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Internet Link

www.minidsp.com

miniDSP, rev A (0,9 V) of rev B (2,0 V): $99.00

minDIGI rev A (less S/PDIF-trabsformers): $55.00

minDIGI rev A (met S/PDIF-trafo's): $60.00

miniAMP: $60.00

Audio plug-ins: $10.00 each

Kits of various combinations of boards are available as well.