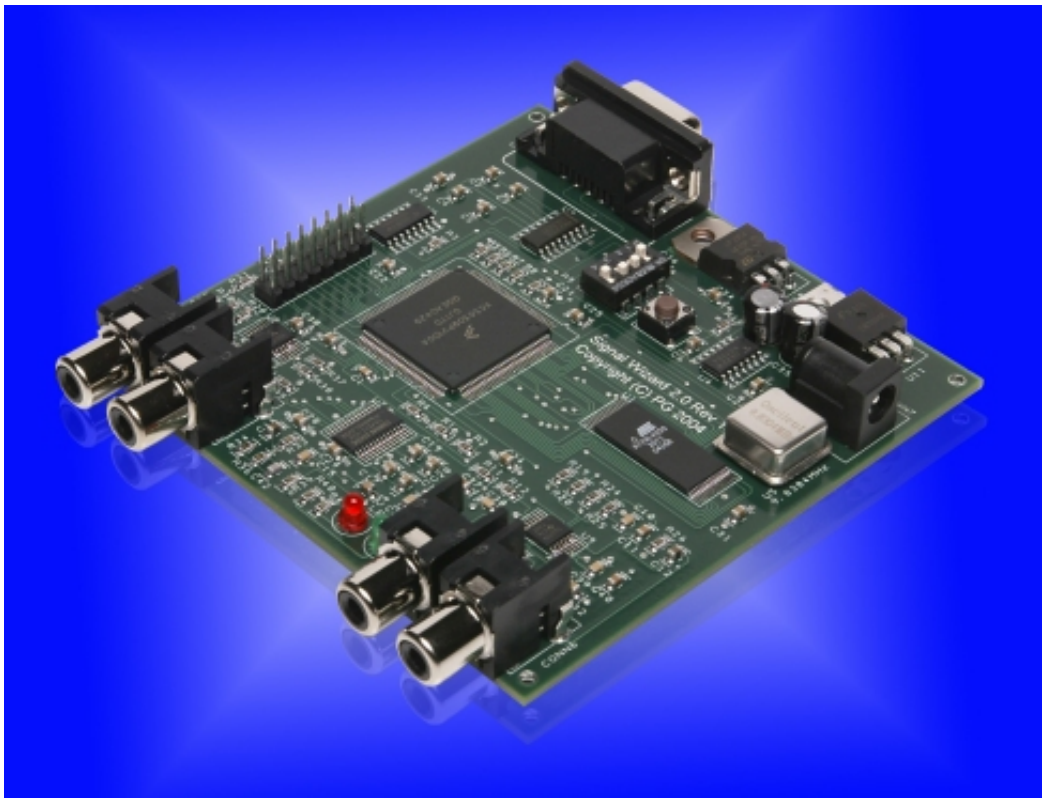
Signal Wizard



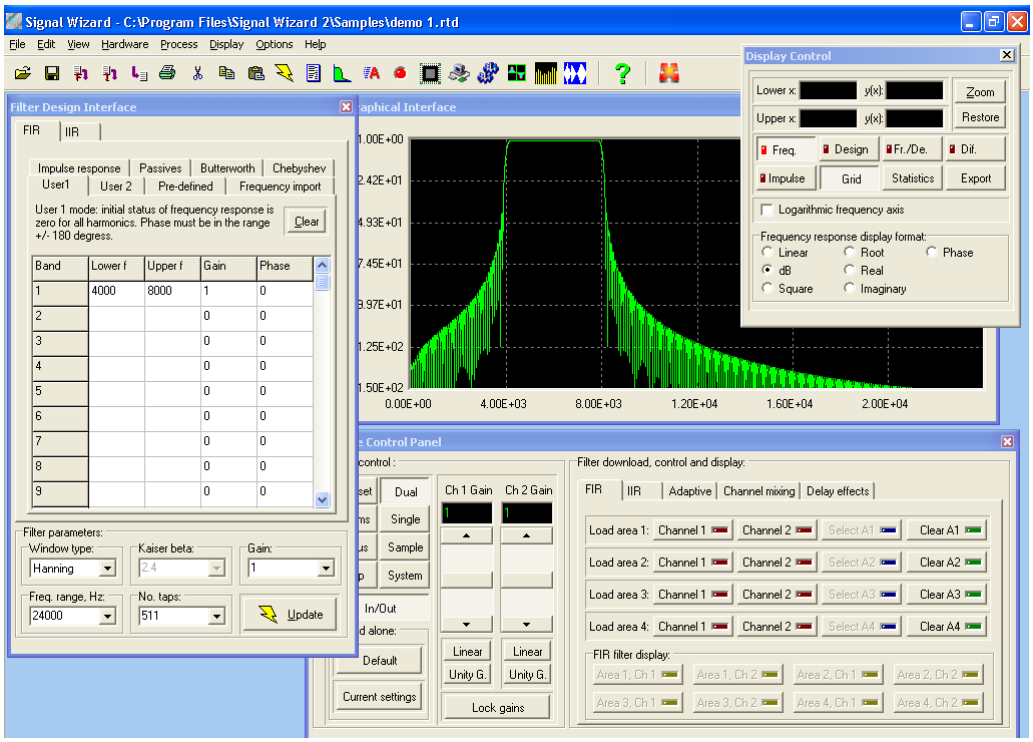
Introduction

The Signal Wizard 2 is a unique, integrated system for designing, downloading and running very high performance filters in real-time. It includes the high-level PC-based software interface that designs the filter according to the user's requirements, a hardware module based on an advanced digital signal processor and a low-level firmware operating system that implements the filtering operations. Once designed, an integrated software interface is used to download the filter to the hardware module via a serial link where it is executed on demand. Most important, the system requires no knowledge of digital signal processing (DSP) theory on the part of the user, or of the mathematics associated with digital filter design. The Signal Wizard is a total-solution package. Due to its flexibility, it is particularly well suited to the real-time processing of audio signals. High quality analogue signal conditioning and a stereo 24-bit resolution codec provide extremely high resolution, sufficient for the most demanding applications. In short, The Signal Wizard 2 brings the power of digital signal processing to any audio-bandwidth domain that requires electronic signal filtering. Applications include audio signal processing, sensor signal conditioning, signal analysis, vibration analysis, education and research in electrical, electronic and other physical sciences.

New features added for Signal wizard 2 include:

- True stand-alone operation (without PC) once configured
- IIR and adaptive filters
- True dual channel operation
- Impulse response import
- Real or complex frequency response import
- Delay options for in/out and FIR filter modes
- Real-time spectrum analyzer
- 6 x faster operation
- Off-line filter mode for wave (WAV) files

Low pass, high bass, multiple band-stop / band-pass filters may be combined to produce very complex filters for frequencies up to 24 kHz, as well as standard infinite impulse response (IIR) and adaptive types. The software can also accept measured responses to define a filter template. This can be used to provide measurement equalisation or to search out signal signatures in noisy environments. In fact, it is a simple matter to produce filters with completely arbitrary frequency magnitude and phase characteristics using the finite impulse response (FIR) method, with no phase distortion, no matter how sharp the filter is. Alternatively, arbitrary phase distortion can be introduced if this is desirable. It is even possible to design and execute real-time deconvolution (inverse) filters using the special invert mode. Because the processing module is so fast, it is possible to design filters with responses far beyond what is possible with traditional analogue techniques. The control program runs under Windows, and provides a user-friendly filter design tool that de-mystifies the process of specifying the filter. The filter design process simply becomes one of describing the desired frequency response. The design package indicates the response that will be produced and also the deviation from that specified. User designs may be stored for re-use and actual responses may be entered from measurements for simulation or equalisation purposes. The filters are calculated and downloaded to the hardware within seconds.

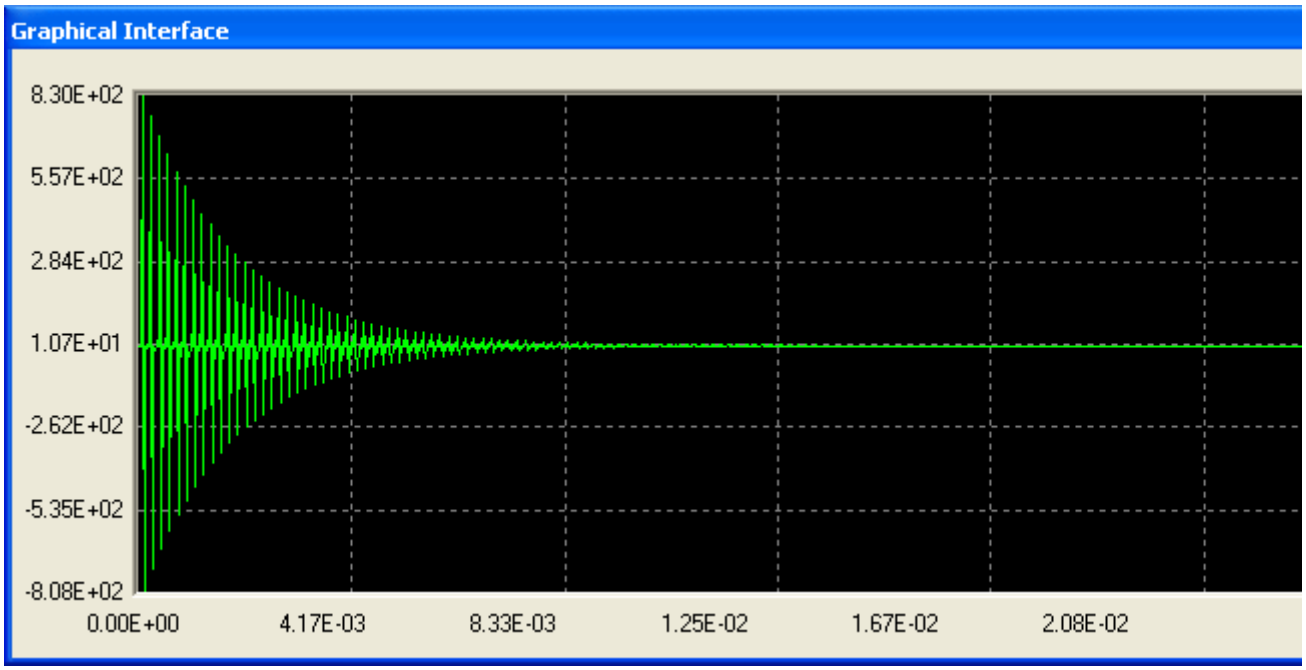


The Main User Interface

Real-time signal processing based on both general purpose microprocessors and fast digital signal processors is a technique that emerged in the 1970's, and is now widely considered one of the fastest growing application areas in the field of digital technology. Applications include biomedical signal analysis, image analysis, image coding and decoding, and audio signal enhancement. Typically for filtering, the analogue waveform is first digitized by an ADC, and the binary values are transmitted to a DSP device that filters them using an appropriate algorithm. The processed data are then sent to a DAC that outputs a filtered analogue signal.

Filters constructed using DSP technology offer many advantages over traditional analogue methods. Most important, they are inherently flexible, since changing the characteristics of the filter merely involves changing the program code or filter coefficients; with an analogue filter, physical reconstruction is required. Furthermore, they are immune to the effects of aging and environmental conditions, since the filtering process is dependent on numerical calculations, not mechanical characteristics of the components. This makes them particularly suited for very low frequency signals. For the same reason, the performance of digital filters can be specified with extreme precision, in contrast to analogue filters where a 5% figure is considered excellent. However, there are significant investments in terms of time and intellectual effort required to understand the functions and instruction set of a particular device, construct the system, and write the algorithms. This cycle can take many months. Contrast this with designing and fabricating a 2nd order analogue filter based on two resistors, two capacitors and one op-amp, a process that might take fifteen minutes. Perhaps for this reason, scientists and engineers who wish to use a particular filter will first attempt an analogue solution. DSP filters in contrast, tend to be used by individuals who are both familiar and comfortable with the art of DSP, in terms of the electronics, coding and mathematics.

The Signal Wizard dispenses completely with this learning curve. In five minutes, you will be able to design and run filters that you might have thought impossible in practical circumstances.



Impulse Responses may be imported and convolved in real time

Features

In summary then, the software has the following features:

- Runs under Windows 98, Me, 2000, NT and XP.
- FIR filters: Multiple pass, stop or arbitrary filters; impulse or frequency response import mode; Butterworth, Chebyshev filters (all major types) to any order; analogue passive networks; zero-phase distortion or arbitrary phase.
- Rectangular, Bartlett, Hamming, Hanning, Blackman or Kaiser Window functions.
- Deconvolution (inverse) or flipped filter options.
- IIR filters: Butterworth and Chebyshev filters (all major types); arbitrary (pole-zero placement); IIR to FIR translation.
- Adaptive filters: true (dual input) adaptive filter, for broadband or narrowband noise cancellation.
- True dual channel operation (independent filters in each channel) with 24-bit resolution.
- Real-time dual channel scope and spectrum analyzer function with data capture.
- Plots impulse and frequency responses as magnitude, dB, square, root, real, imaginary or phase. Log or linear frequency axis. Pole-zero plots and coefficient export.
- Channel mixing and inversion.
- Channel delays from 21 microseconds to 1.8 seconds.
- Extensive filter analysis statistics.
- Animate facility for tap adjustment.
- Impulse and frequency responses exportable as text files for off-line processing and spreadsheet analysis.
- Real-time gain and sample rate control.
- Filter module holds up to 8 filters in non-volatile memory.
- Stand alone filter operation on start-up or connect to PC.
- Twelve sample rates from 48 kHz down to 4 kHz.
- Off-line filter option for wave files.
- Maximum input and output level 2 V peak-to-peak.
- Serial interface operating at 115.2 kilo baud (auto-selected).

In brief, the software performs the following major functions:

It designs the filter according to the user's specifications. The filter is expressed as a set of FIR coefficients or taps (there is no need to be concerned with this if you don't want to be), IIR coefficients or as an adaptive expression. Collectively, the coefficients of a filter are known as the impulse response.

It converts the floating-point representation of the coefficients to 24-bit fractional arithmetic (again, this is invisible to the user).

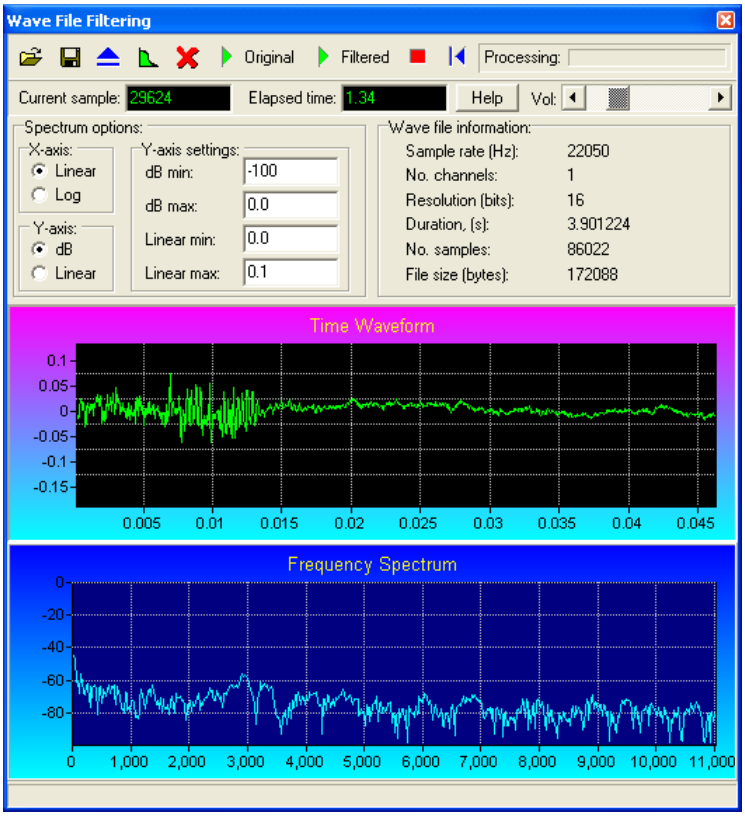
It automatically downloads the low-level firmware operating system to the hardware via a serial link.

It automatically communicates with the firmware embedded in the digital system to download the filter coefficients via the same serial link.

It allows the user to alter the system's gain, sampling rate and delay during run-time.

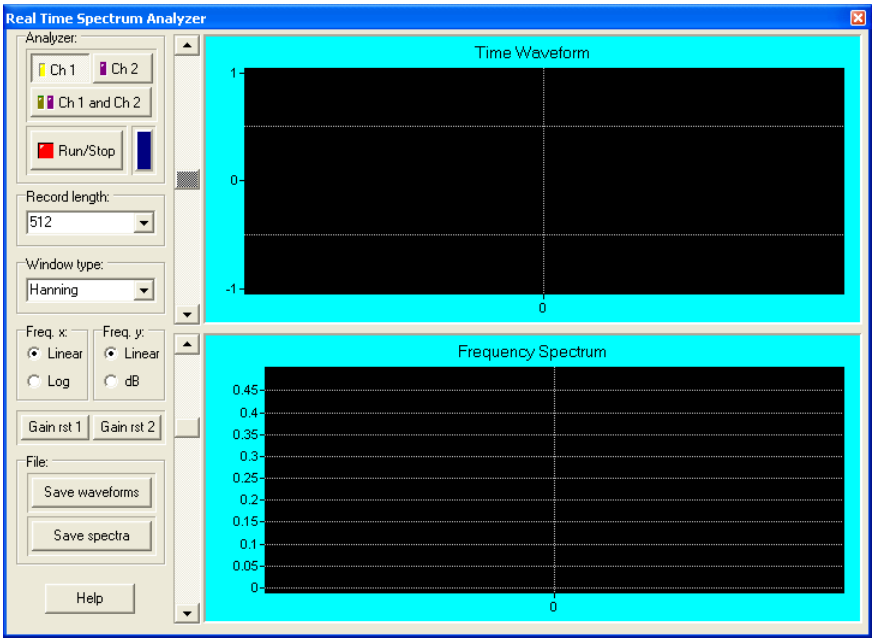
It allows the user to switch instantaneously between a filtered or non-filtered signal at run-time.

Most important, the software includes a flexible filter design section, a window that displays the frequency or impulse response of the realised filter in various formats, and a hardware control section for downloading a filter and generally communicating with the filter module. During design-time, the user simply enters the frequency range of the incoming signal, the pass or stop bands of the filter with their magnitudes, phases (optional) and frequency cut-off points, the number of taps (coefficients) in the filter and the windowing function required. The number of taps and the windowing function determine how closely the final filter will match the user's desired ideal. The user can manually specify the arbitrary width and magnitude/phase of up to one hundred such filter bands. It is even possible to import an ASCII text file representing an arbitrarily shaped impulse or frequency response (real or complex).



The Wave File Filtering Interface

In addition, the software has numerous utilities for comparing the accuracy of the final filter with that of the desired result. The frequency response of the filter the system obtains (as opposed to the ideal design information supplied), plotted in the display area, is what the hardware will actually produce, to a very high order of precision. In general, the more taps the user specifies, the closer the actual filter will be to the ideal design.



The Real Time Spectrum Analyzer Interface

The software has an additional facility that allows the user to export the frequency and impulse responses of the filter, instead of downloading them to the hardware module. This allows off-line processing to be performed on previously acquired and stored data. Filtering in this way will yield identical results to those produced by the real-time system. Alternatively, single channel wave (WAV) files may be filtered directly using any FIR filter designed by the system.

The Hardware

The hardware module connects to the PC via a standard serial (RS232) link. The connections are shown below.

Essentially, the module comprises signal pre- and post-conditioning circuitry, a high-resolution stereo codec, a high-speed DSP device, memory, timing and control sub-systems. The 24-bit over-sampling stereo codec system is configurable by the user to any one of twelve sample rates, ranging from 48 kHz down to 4 kHz. In terms of input signal frequency ranges, this equates to 24 kHz down to 2 kHz. The codec accepts or generates a 2 V peak-to-peak signal. The power of the hardware module depends on its own operating system, which is invisible to the user, but communicates with the PC software.

Exactly how fast can the filters operate, and how many filter coefficients can they practically employ? In order to understand how we arrive at the answers to these questions, you need to know a little bit about filter theory and MIP rates of microprocessors. The maximum number of taps permissible when operating at the highest frequency range of 24 kHz and in single channel mode, is 527. (Remember that at this range, the system is sampling at 48 kHz. The sample rate is always double the frequency range.) This represents a very sharp filter indeed. Using a frequency range of 12 kHz (sample rate of 24 kHz) in the same modes, the system can operate a filter with a maximum of 937 taps. At any range below this, it can operate a filter with a maximum of 1024 taps. Incidentally, the performance of a 1024-tap filter is so extremely sharp that it is quite unlikely that you would ever need to use it.

Minimum System Requirements

- 400 MHz PC running Windows 98, Me, 2000, XP or NT
- SVGA display (800 X 600 pixels) or higher, using normal size fonts
- CD ROM drive
- 20 Mbytes of free hard disc space
- Serial port, capable of 115.2 k bps

The demo version of Signal Wizard 2 has all the functionality of the full version, with the exception that files cannot be saved. The zipped file is 5.8 Mbytes and unzips to 11.1 Mbytes.

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