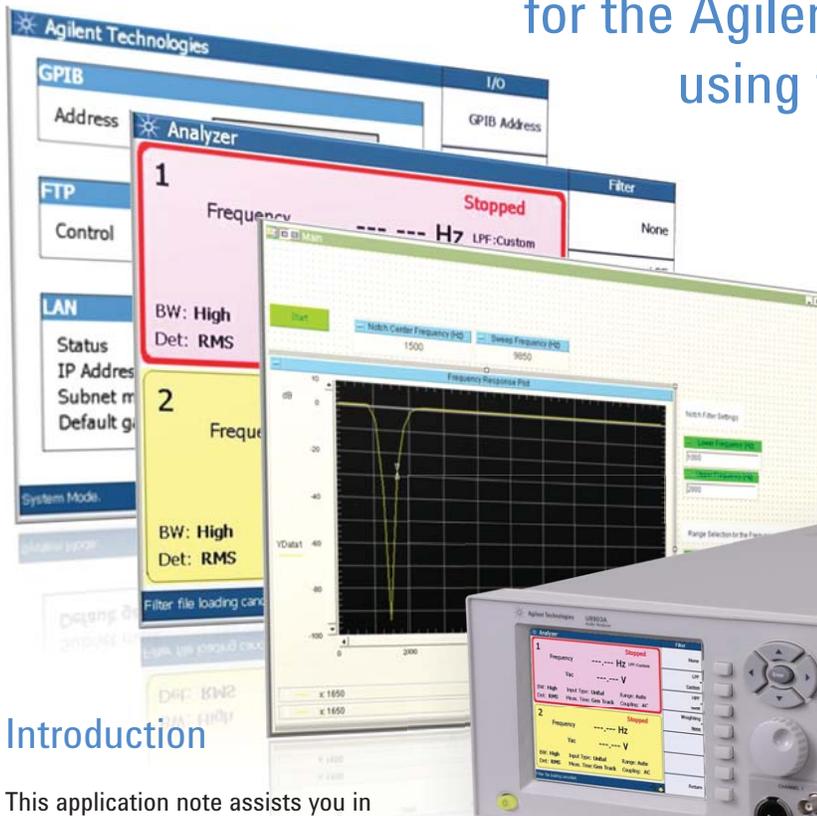


# Three Easy Steps to Create Your Own Notch Filter

for the Agilent U8903A audio analyzer  
using the Agilent VEE software

Application Note



## Introduction

This application note assists you in creating your own customized notch filter for your audio analysis by using a VEE program. The VEE program will help you to create an FIR digital filter to be used in your U8903A. You can have the option of installing the notch filter permanently into your U8903A nonvolatile memory or of recalling the filter when it is needed from a PC through IO connectivity.



Agilent Technologies

## Back to Basics

### What is an audio filter?

Any medium that an audio signal passes through, whatever its form, can be regarded as a filter. Filters are usually used to discriminate between a frequency or a band of frequencies from a given signal, which is normally a mixture of both desired and undesired signals. The undesired portion of the signal commonly comes from noise sources such as power line hum, which are not required for a particular application.

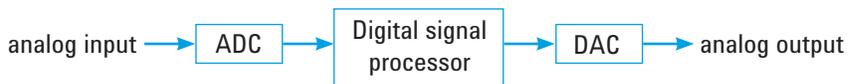
However, we do not usually think of something as a filter unless it can modify sound in some way. For example, a speaker wire is not considered a filter, but the speaker is. The tone control circuit in an ordinary car radio is a filter, as are the bass, midrange, and treble boosts in a stereo preamplifier. Graphic equalizers, echo devices, phase shifters, and speaker crossover networks are further examples of useful filters in audio. There are also examples of undesirable filtering, such as the uneven reinforcement of certain frequencies in a room with “bad acoustics.”

### Digital or analog filter?

An analog filter is any filter that operates on continuous-time signals. In other respects, it is just like a digital filter. Analog filters were used successfully for decades before digital filters come along. Although analog filters are excellent in some aspects, especially in cost, they do have some serious drawbacks. One of the drawbacks of analog filters is the non-linear phase characteristics. This does not affect most of the common applications, but it is heavily involved in applications like telecommunications and voice processing. Another drawback in analog filters is the lack of sharpness at the cut-off frequency. It is possible to increase roll-off rate by cascading filter stages, but this will increase the cost and complexity of the system. On the other hand, it is possible to achieve all these characteristics fairly by using a digital filter.

A digital filter is just a filter that operates on digital signals, such as sound represented inside a computer. It is a computation, which takes one sequence of numbers (the input signal) and produces a new sequence of numbers (the filtered output signal). The filters mentioned are not for digital signals only as they can be operated on signals that are not digital as well. It is important to realize that a digital filter can do anything that a real-world filter can do. That is, all the filters alluded to above can be simulated to an arbitrary degree of precision digitally.

Similar to any other digital signal processing system, the implementation of the digital filter also requires an ADC, DAC, and a processor. A simple system with these devices are shown below.



Here the filtering action is accomplished through the software running in the digital signal processor. The algorithms used for this purpose are known as *digital filter algorithms*. This algorithm takes an input from the ADC, calculates the output, and sends it to the DAC. There are mainly two types of filter algorithms:

- **Finite Impulse Response filter (FIR)** — The impulse response is “finite” because there is no feedback in the filter. If you put in an impulse (i.e., a single 1 sample followed by many 0 samples), zeroes will eventually come out after the 1 sample has made its way past all the coefficients in the delay line.
- **Infinite Impulse Response filter (IIR)** — The impulse response is “infinite” because there is feedback in the filter; if you put in an impulse, an infinite number of non-zero values will come out (theoretically).

## FIR filters vs. IIR filters

Compared to IIR filters, FIR filters offer the following advantages:

- They can easily be designed to operate in linear phase. Linear-phase filters delay the input signal, but do not distort its phase.
- They are simple to implement because for most DSP microprocessors, the FIR calculation can be done by looping a single instruction.
- They are suited to multi-rate applications. These are operations that include either "decimations" (reducing the sampling rate), "interpolations" (increasing the sampling rate), or a combination of both.
- They have suitable numeric properties. In practice, all DSP filters must be implemented using "finite-precision" arithmetic; that is, a limited number of bits. The use of finite-precision arithmetic in IIR filters can cause significant problems due to feedback, but FIR filters have no feedback, so they can usually be implemented using fewer bits. Therefore, the designer faces fewer practical problems related to non-ideal arithmetic.
- They can be implemented using fractional arithmetic. Unlike IIR filters, it is always possible to implement an FIR filter using coefficients with magnitudes less than 1.0. (The overall gain of the FIR filter can be adjusted at its output, if desired.) This is an important consideration when using fixed-point DSP, because it makes the implementation much simpler.

## What is a notch filter?

A notch filter is a filter that passes all frequencies except those in a stopband centered on a center frequency. It is the opposite of a band-pass filter. A notch filter is a band-stop filter with a narrow stopband (high **Q** quality factor). Notch filters are used in live sound reproduction (public address systems, also known as PA systems) and in instrument amplifiers (especially amplifiers or preamplifiers for acoustic instruments such as the acoustic guitar, mandolin, and bass instrument amplifier) to reduce or prevent feedback, while having little noticeable effect on the rest of the frequency spectrum. Other names include *band limit filter*, *T-notch filter*, *band-elimination filter*, and *band-reject filter*.

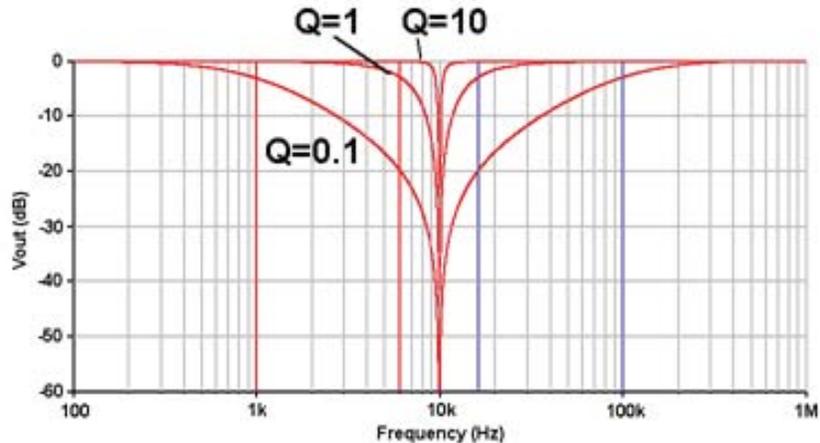


Figure 1. Phase responses of a notch filter

The figure above shows the phase responses of a notch filter where the greatest rate of change is at the center frequency. The rate of change becomes more rapid as the **Q** of the filter increases. The group delay of a notch filter is greatest at the center frequency, and becomes longer as the **Q** of the filter increases.

## Introduction to the U8903A Audio Analyzer

The U8903A is a digital signal processing (DSP) based audio measurement system that combines both an audio generator and analyzer. This instrument consists of two channels each for the audio generator and analyzer; the output and input configurations are fully independent of each other.

The U8903A audio generator has a frequency range of 5 Hz to 80 kHz, and the sine wave amplitude range can cover from 0 Vrms to 8 Vrms (11.3 Vp) for the unbalanced test output configuration and 0 Vrms to 16 Vrms (22.6 Vp) for the balanced output configuration. When using the audio generator, you can generate not only a basic sine waveform, but also square, dual sine, variable phase, noise, DC, multitone, and arbitrary waveforms.

The U8903A audio analyzer has a frequency measurement range of 10 Hz to 100 kHz with an amplitude measurement range of 1  $\mu$ Vrms to 140 Vrms (200 Vp).

In addition, the U8903A is equipped with frequency and time domain graph functions, as well as sweep capability for frequency, amplitude, and phase. This allows you to perform a wide range of audio parameter measurements consisting of voltage, frequency, total harmonics distortion plus noise (THD + N) signal, noise and distortion (SINAD), signal-to-noise ratio (SNR), noise level, SMPTE inter-modulation distortion, difference frequency distortion (DFD), phase, and crosstalk.

The U8903A also supports the industry standards of instrument connectivity such as GPIB, USB, and LAN.

To address challenging audio applications, the U8903A is equipped with standard and custom filters. A selection of built-in filters simplifies audio measurements by providing weighting networks required by international standards. These include CCIR, CCIR/ARM, and CCIT weighting filters, a C message filter, and an ANSI "A" weighting filter. In addition to the standard filters, you can create custom filters using applications such as MATLAB or Agilent VEE and upload the filters through the USB port of the audio analyzer. The U8903A also includes selectable 15 kHz, 20 kHz, and 30 kHz low-pass filters to reject unwanted out-of-band signals and noise.



# Customizing Your Notch Filter

## Prerequisite:

A PC installed with the Agilent VEE Pro (version 8.0 and above)

### Step 1

#### Define your settings and requirements.

##### Notch filter settings:

1. Determine the upper frequency,  $f_u$ .
2. Determine the lower frequency,  $f_l$ .

The delta (center frequency) between the upper and lower frequency is the notched frequency. It will be calculated automatically based on the following formula.

$$f_N = (f_u - f_l) / 2$$

##### Range selection for the Frequency Response plot:

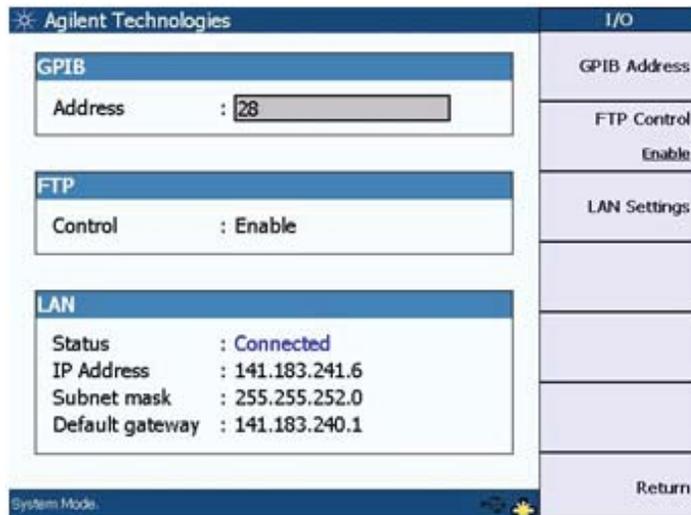
1. Determine the start frequency. (Please select a frequency value higher than the lower frequency.)
2. Determine the stop frequency. (Please select a frequency value higher than the upper frequency.)
3. Determine the step size.

### Step 2

#### Set up your connection.

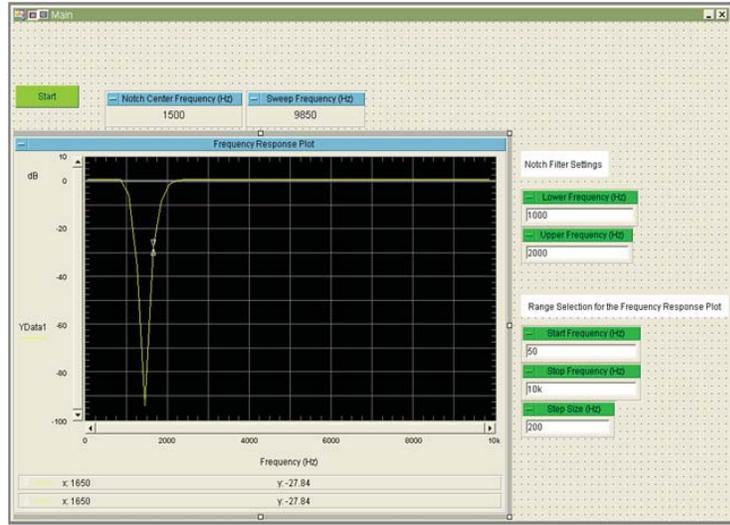
1. Save the VEE program notch filter design to your PC.
2. Connect the U8903A audio analyzer to your PC with either a USB/LAN or GPIB connection.
3. Acquire the U8903A IO connection information (as shown in the figure below).
4. Open the VEE program (your notch filter design program).

Remember to check the connection for the U8903A with the VEE program before you run the notch filter design program. For example, if you are using LAN connection, ensure that the IP address in the VEE program matches with the IP address given by the U8903A under the System Mode.



**Step 3** Run and save.

1. Key in the values for the Lower Frequency and Upper Frequency in the Notch Filter settings.
2. Key in the values of the Start Frequency, Stop Frequency, and Step Size in the Range Selection for the Frequency Response Plot.
3. Click the **Start** button to execute the program.
4. A pop-up window will prompt you to select a destination to save the raw data file. Select the location on your PC where you want to save the raw data file and insert the ".juf" extension behind your file name. (Example: Notch.juf)



**Descriptions on the VEE program above:**

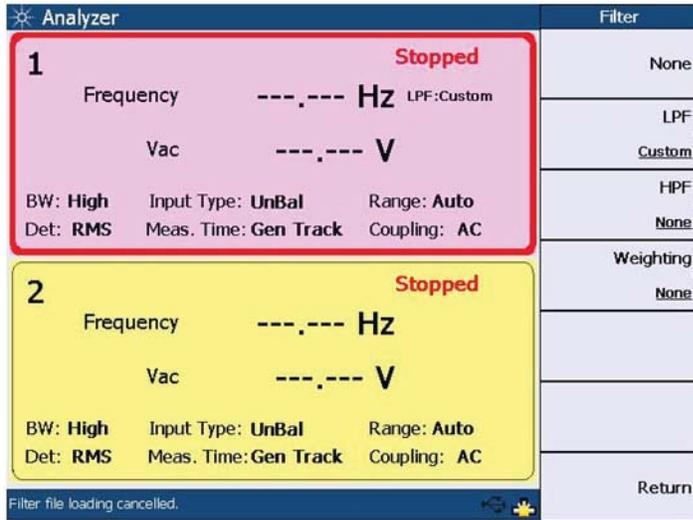
Item	Descriptions
Notch Center Frequency (Hz)	The frequency that you want to notch. Automatically calculated based on the delta between the Lower Frequency and the Upper Frequency values.
Sweep Frequency (Hz)	This displays the current sweep point.
	These markers indicate the value for the xy location in the sweep graph.

**NOTE**

After the VEE program execution is completed, the notch filter is temporary stored into the volatile memory of the U8903A. The notch filter will be erased after a power cycle.

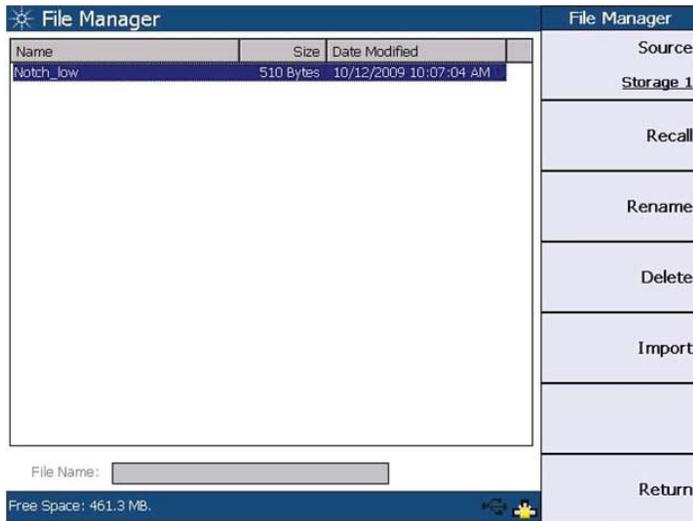
**NOTE**

The notch filter is not a standard filter. Therefore, it will be saved as Analyzer Channel 1 Custom filter category in the Low Pass Filter (LPF) section (as shown below).



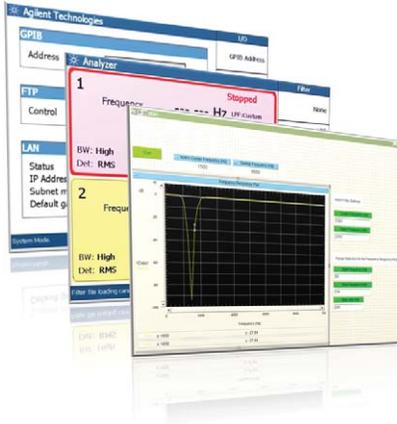
**NOTE**

If you wish to permanently store the notch filter into the nonvolatile memory of the U8903A, save the ".juf" file from the PC into your USB Flash Memory device and upload it to the U8903A. Select *Custom* in either the low pass, high pass, or weighting filters section so that you can upload a user-defined filter file via the File Manager (as shown below).



## Summary

This application note is written to assist you in creating a simple customized notch filter using the VEE program for audio applications. This VEE program will help you to carry out your day-to-day engineering tasks effectively and efficiently, especially if your job focus is in R&D designs and QA inspections.



## References

1. <http://www-ccrma.stanford.edu/~jos/filters/>
2. [http://logix4u.net/DSP/Digital\\_Filters/Digital\\_filters.html#](http://logix4u.net/DSP/Digital_Filters/Digital_filters.html#)
3. <http://www.dspguru.com/info/faqs/iirfaq.htm>
4. <http://www-k.ext.ti.com/SRVS/Data/ti/KnowledgeBases/analog/document/faqs/notch.htm>

## Related Agilent Literatures

1. Agilent U8903A Audio Analyzer Data Sheet (5990-3831EN)
2. Agilent U8903A Audio Analyzer User's Guide (U8903-90002)



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