General purpose filter engine white paper

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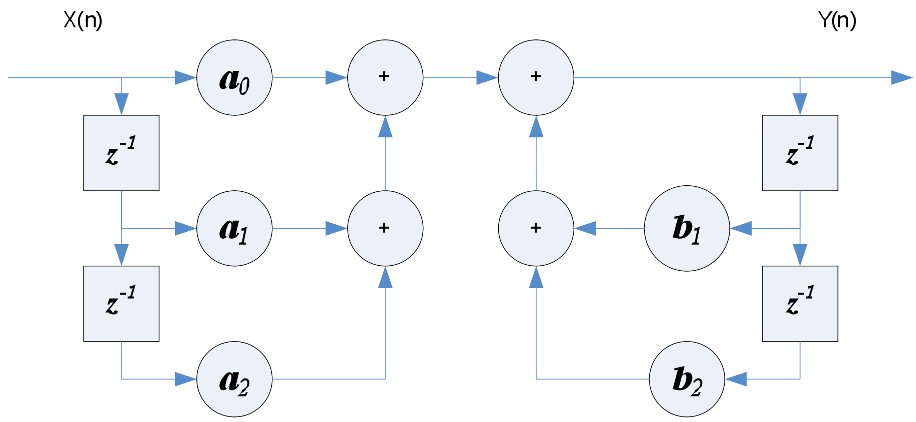
Abstract
Recent years have witnessed huge expansion in the market for portable devices with hi-fi audio playback capability such as MP3 players, multimedia phones and hand-held games consoles. However, such devices often use low cost transducers, and are operated in noisy environments; hence the audio quality experienced by the user is lower than the inherent quality of the programme material. Mobile telephones are also expected to operate as media players, and have similar requirements, as well as the need to improve legibility during voice calls.

It is possible to improve the perceived audio quality by compensating for transducer frequency response; in mobile phone use, it is possible to suppress some of the environmental noise by judicious use of filtering. These types of filtering operations can usefully be performed in the codec itself, freeing any host processor from the burden of real-time audio filtering. Such filtering operations must be completely programmable however, since the filters required will alter with the application and equipment. A general purpose filtering engine allows these, and many other improvements to be made.

Introduction
The small form factor in portable media devices prohibits proper acoustic design. And, it is important to be able to improve the audio acoustic design late in the product design cycle. Using dedicated IC's to perform DSP take up board space, which is usually at a premium in modern portable devices. Dialog's approach is to offer general purpose filtering within the codec device this allows effects such as bass enhancement, acoustic transducer filtering, ambient noise suppression, equalisation and stereo widening to be implemented without modifying the host processor software.

Programmable filters
One of the most useful general purpose digital filters is the biquad section, which can implement all second-order filter topologies. There are several canonical forms for the biquad. The direct-form type I architecture is straightforward to understand from simple filter theory and has several advantages when used in hardware digital signal processing such as insensitivity to arithmetic overflow during computation. A diagram of the implementation architecture of the biquad is shown below.

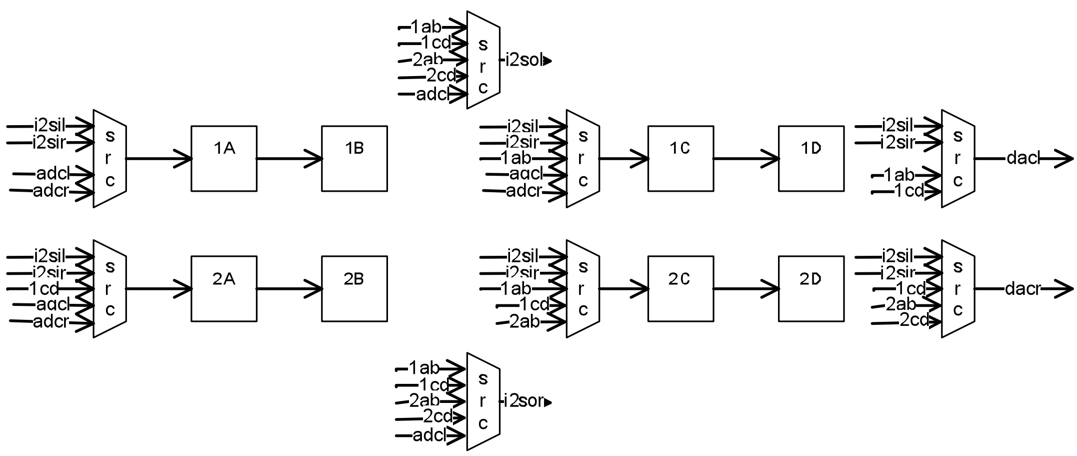


X(n) represents the sampled input to the filter and Y(n) the sampled output at a given sample time, n. The forward coefficients are a_0, a_1, a_2 and the reverse coefficients are b_1, b_2. A single sample delay is represented by z^{-1}.

Flexibility
In order to perform functions that are useful, the general purpose filtering capabilities provided by a device must satisfy the following criteria:

- Provide a useful number of biquad sections
- Allow biquad sections to be interconnected in a flexible manner.

The DA7210 provides eight independent biquad sections, grouped into four sets; each set may be connected flexibly from multiple input sources, including other filter sections. In addition, a mixing function is provided, that allows various combinations of biquads to be mixed together. The coefficients are represented as 16 bit values, which may be programmed into 8-bit configuration registers on the DA7210. The diagram below shows the four groups of biquads (note that not all possible connections are shown). There are also four paths that can be connected to the filter engine: the left and right DAC output channels, and the digital audio interface channels. Each of these can be connected to the outputs of filter groups, or to the raw digital inputs.



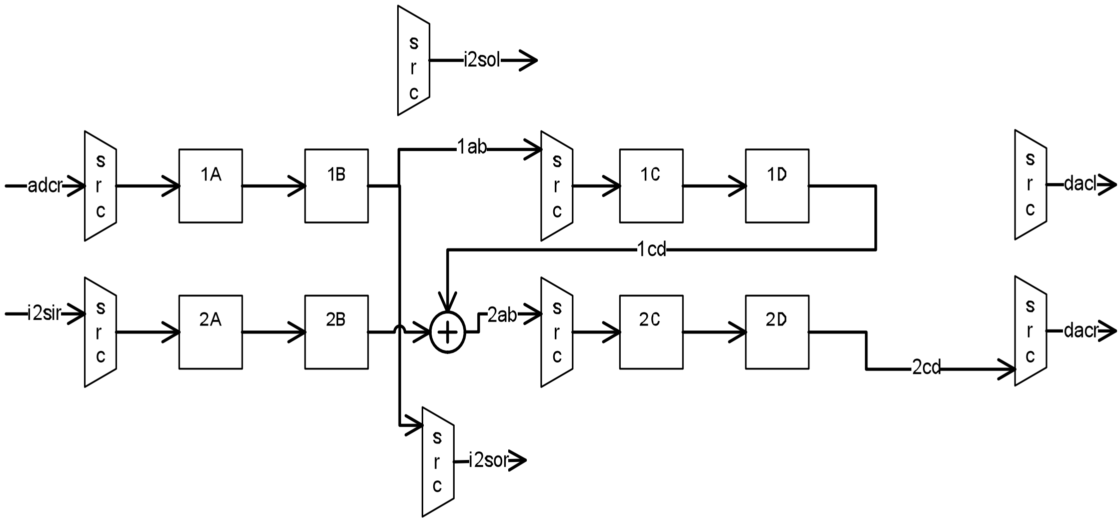
Filter groups may also be disabled in order to reduce power consumption.

The routing capabilities of the filter engine are also available even when all filters are disabled.

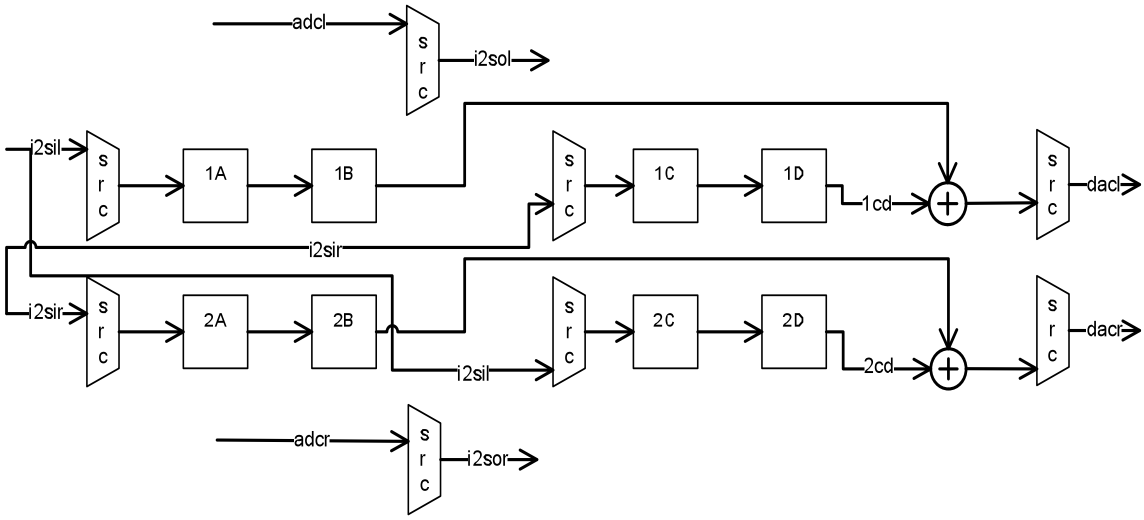
A simple example of the filter topologies that can be realised for equalisation purposes is an eight biquad filter (16^{th} order) that filters a digital input stream before output from a DAC channel whereby each DAC channel may have an eighth order filter applied for stereo playback and each input and output may have a fourth order filter applied for stereo record and playback.

For mobile telephone applications, the general purpose filter engine provides a flexible solution for digital sidetone generation, as shown below. The microphone input is fed from the ADC right channel, then equalised to correct for transducer characteristics, in filter set 1AB. This is sent to the TX channel over the digital audio interface. The RX channel is received over the digital audio interface right channel, equalised in filters 2CD and then sent to the right DAC channel.

The equalised microphone output is also filtered again in filters 1CD and then mixed with the RX data before being equalised for output. The relative gain of the sidetone channel is set by the coefficients for filters 1CD and the frequency characteristics are chosen to eliminate 'howl round' and ambient noise reduction.

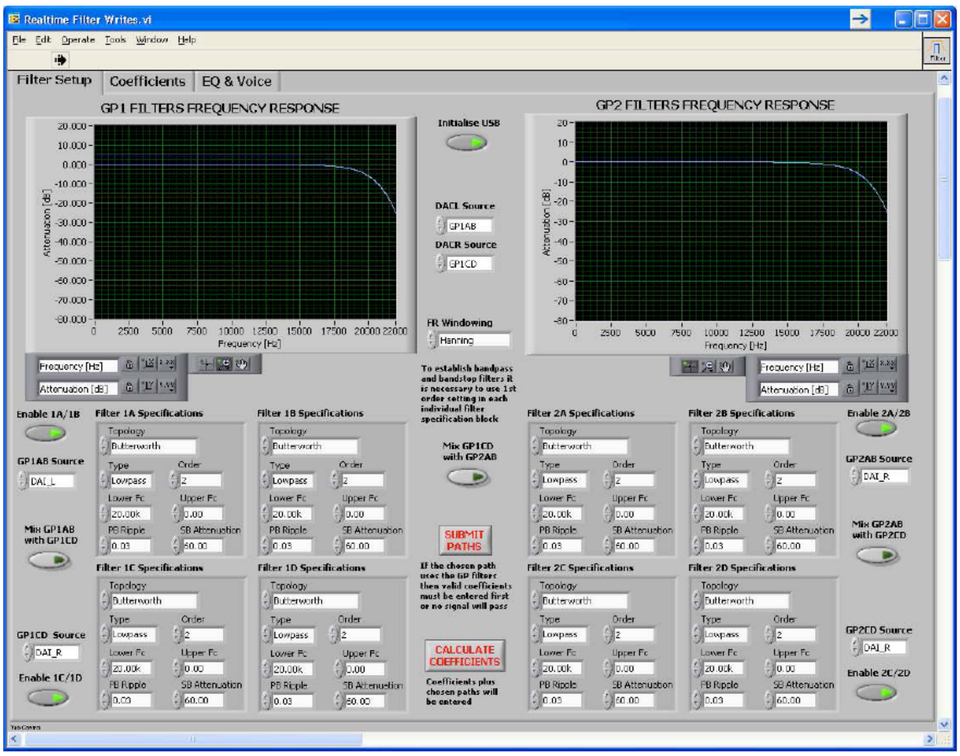


Enhanced stereo width control is also possible. In the configuration below, both left and right digital input channels are sent to two sets of filters. The left and right outputs are formed by mixing the outputs from the filtered versions of the left and right input signals. By choosing suitable weightings and frequency responses an enhanced stereo effect is created; this sounds more natural than fully separated left and right signals when listening through headphones.



Design aids

When developing filtering applications for devices it is vital to be able to prototype new filter responses quickly. Whilst there are a plethora of filter design tools available it is most useful to have a tool that can generate the required configuration register values for a specific device. A design tool is available for the DA7210 that can generate the required coefficient values for many standard filters, translate these coefficients into values suitable for programming into the DA7210 and then download them into a device. This considerably shortens the design development time. The tool also understands the many routing and special purpose options in the DA7210 filter engine. A screenshot of the tool is shown below.



If a designer wishes, it is possible to calculate filter coefficients for a special purpose filter, and then input these coefficients. The tool will then provide a graphical calculation of the frequency response, and translate the values for the DA7210.

Summary

Consumers expect both improved audio quality and increased functionality from mobile audio products. While individual components are constantly being developed to improve quality and performance it is often challenging to deliver these without increasing the load on the system processor, which would lead to increased power drain. The DA7210 enables the system designer to deliver enhanced audio at the codec, without the need to burden the system processor. The DA7210 family of audio codecs offer exceptionally low power and high quality enhanced audio.

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