INTRODUCTION

The PARAGON™ DIGITAL offers great flexibility in adjusting input/output characteristics as well as shaping of the system's frequency response. The input/output characteristics of the system are adjusted by up to 4 channels of independent signal processing, AGC-O, peak clipping and low level expansion (squelch) blocks. The system's frequency response can be controlled by adjusting individual channel's gain and the crossover frequencies between channels. Also, the system's frequency response can be controlled using PARAGON™ DIGITAL biquad filters.

The PARAGON™ DIGITAL hybrid has four biquad filters available. The first two of them, Pre1 and Pre2 are located before the Band Split Filter in the signal path. Those filters are cascaded and it is important to note that if the Pre1 filter is disabled, the hardware assumes that the Pre2 filter is also disabled. The other two filters, PostA and PostB are located before the Notch Filter and the Peak Clipping blocks respectively. Filters PostA and PostB can be used completely independently. Please refer to the GB3210-S02 Pal Configuration Block Diagram for locations of the biquad filters in the signal path.

BIQUAD FILTER STRUCTURE

Each of the PARAGON™ DIGITAL biquad filters is an IIR (Infinite Impulse Response) digital filter implemented as Direct Form 1 structure. The block diagram of the Direct Form 1 structure is shown in Fig 2.

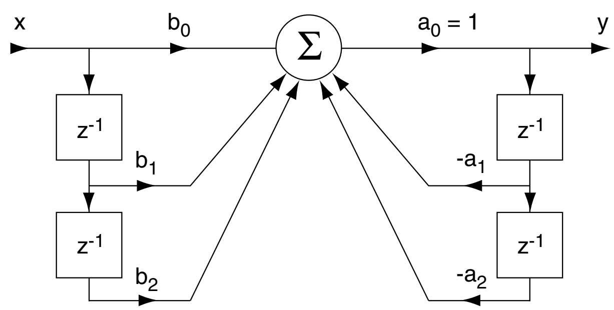

Fig. 2 Direct Form 1 IIR Structure Block Diagram

The transfer function of each of the biquad filter can be expressed as follows:

$$H(z) = \frac{b_0 + b_1 \times z^{-1} + b_2 \times z^{-2}}{1 + a_1 \times z^{-1} + a_2 \times z^{-2}}$$

By multiplying the numerator and the denominator by z^2 the transfer function can also be expressed as:

$$H(z) = \frac{b_0 \times z^2 + b_1 \times z^1 + b_2}{z^2 + a_1 \times z^1 + a_2}$$

Note that the coefficient a_0 is hard-wired to always be a 1. The coefficients are each 16 bits in length and include one sign bit, one bit to the left of the decimal point, and 14 bits

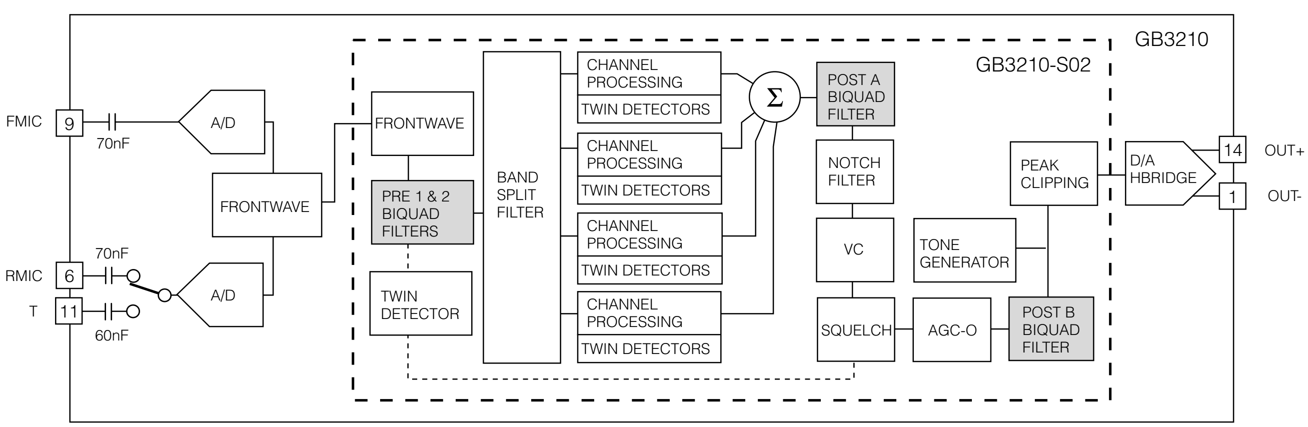

Fig. 1 GB3210-SO2 Pal Configuration Block Diagram

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All multiplications and additions are done with a single multiplier-accumulator (MAC) and therefore the following rules should be considered in order for the biquad filter to function properly:

1. All coefficients {b0, b1, b2, a1, a2} must be in the range [-2;+2].
2. The calculated filter's output signal level should not exceed the full scale representation in the filter. For example, if the input signal to the biquad is -10dBFS (10dB below Full Scale), therefore the biquad has only 10dB of headroom before it starts overflowing.

Please take note that the underlying code in the product components automatically checks all of the filters in the system for stability (that is, the poles have to be within the unit circle) before updating the graphs on the screen or programming the coefficients into the hybrid. If the Interactive Data Sheet receives an exception from the underlying stability checking code, it will automatically disable the biquad being modified and display a warning message. When the filter is made stable again, it can be re-enabled.

DIGITAL SYSTEM LIMITATIONS

When designing and implementing an IIR filter, it is important to consider the limitations of any DSP system. The limitations described here might cause the IIR filter to become noisy, distorting or even unstable. In that case, it may be necessary to adjust some of the filter's parameters so it will perform as expected.

When the signal is converted from analog to digital form, the precision is limited by the number of bits available. The signals are not only subject to limited precision, filter coefficients follow the same rule - they are also subject to limited precision of digital systems. The limited precision of all digital systems leads to quantization errors.

Those errors due to limited precision are nonlinear and signal dependent. The nonlinearity can eventually lead to instability, particularly with IIR filters.

There is no method that can eliminate quantization errors. However those errors can be minimized by keeping values large so that the maximum number of bits is used to represent them. But, there is a limit to how large the numbers can be.

There is also another potential source of error in DSP systems, particularly in IIR filters. It is possible for a result of IIR filter processing to exceed the maximum number size. When that happens, the hardware may saturate or overflow. To avoid the overflow or saturation, the input signal and/or filter coefficients should be scaled down.

DIGITAL IIR FILTER DESIGN

The following examples illustrate one of the IIR filter design algorithms. Each filter is first designed as an analog filter, the filter's characteristics (order, corner frequency, passband gain) are translated into an analog transfer function. Poles and zeros of the analog transfer function are then matched to the poles and zeros of the digital filter's transfer function using bilinear transformation. The digital filter's coefficients (from the digital filter's transfer function) can then be entered into the Biquad Section of the Interactive Data Sheet to implement that particular filter. Those examples require some knowledge of complex math and "s" and "z" operators used in analog and digital transfer functions.

DIGITAL FILTER DESIGN EXAMPLES:

Example 1: 1st order low pass filter with 4kHz corner frequency and 0dB gain

The transfer function of the 1st order analog low pass filter with 0dB gain:

$$H(s) = \frac{\omega_C}{s + \omega_C} \quad \text{where } \omega_C = 2\pi f_C$$

f_C is -3dB corner frequency in Hz , $\pi = 3.1415935$

The transfer function has pole s_{p1} at $-\omega_C$ and zero s_{z1} at ∞

The bilinear transformation is used to convert the analog transfer function to a digital transfer function. The pole and the zero of the digital transfer function are matched to the pole and zero of the analog transfer function according to the following formula:

$$z = \frac{1 + \frac{T}{2} \times s}{1 - \frac{T}{2} \times s}$$

where T is the sampling period, $T = 1/f_S$, and f_S is the sampling frequency.

The corner frequency of the analog filter f_C = 4000Hz therefore ω_C = 25132. The transfer function of this analog low pass filter is:

$$H(s) = \frac{25132}{s + 25132}$$, the pole $s_{p1} = -25132$, the zero $s_{z1} = \infty$

Using the bilinear transformation, sampling frequency $f_s = 32000$Hz:

$$z_{p1} = \frac{1 + \frac{-25132}{64000}}{1 - \frac{-25132}{64000}} \qquad z_{p1} = 0.4360604 \qquad z_{z1} = -1$$

The transfer function of the first order digital filter is:

$$H(z) = G \times \frac{z - z_{z1}}{z - z_{p1}}$$

Gain parameter G has to be calculated so the digital filter has unity gain at DC:

$$H(z)\Big/_{z=1} = G \times \frac{1 - z_{z1}}{1 - z_{p1}} = 1 \quad \text{thus } G = \frac{1 - z_{p1}}{2}$$

G = 0.2819698

The transfer function of the digital filter:

$$H(z) = 0.2819698 \times \frac{z + 1}{z - 0.4360604} = \frac{0.2819698 \times z + 0.2819698}{z - 0.4360604}$$

The filter's coefficients have to be converted to 14-bit integer numbers. The conversion is done by multiplying the fractional coefficients of the transfer function by 2^{14} and rounding the results to the nearest integer number. The rounded integers can then be entered into one of the biquad sections in the Interactive Data Sheet and programmed into the PARAGON™ DIGITAL hybrid.

COEFFICIENT	REAL VALUE	14-BIT INTEGER VALUE
b0	0.2819698	4620
b1	0.2819698	4620
b2	0	0
a0	1	16384
a1	-0.4360604	-7144
a2	0	0

Example 2: 1st order high pass filter with 500Hz corner frequency and 0dB gain

The transfer function of the 1st order analog high pass filter with 0dB gain:

$$H(s) = \frac{s}{s + \omega_c} \quad \text{where } \omega_c = 2\pi f_c$$

f_c is -3dB corner frequency in Hz , $\pi = 3.1415935$

The transfer function has pole s_{p1} at $-\omega_c$ and zero s_{z1} at 0. The bilinear transformation is used to convert the analog transfer function to a digital transfer function. The pole and the zero of the digital transfer function are matched to the pole and zero of the analog transfer function according to the following formula:

$$z = \frac{1 + \frac{T}{2} \times s}{1 - \frac{T}{2} \times s}$$

where T is sampling period, $T = 1/f_s$, and f_s is the sampling frequency.

The corner frequency of the analog high pass filter $f_c = 500$Hz therefore $\omega_c = 3141$.

The transfer function of this analog low pass filter is:

$$H(s) = \frac{s}{s + 3141} \qquad s_{p1} = -3141,\ s_{z1} = 0$$

Using the bilinear transformation, sampling frequency $f_s = 32000$Hz:

$$z_{p1} = \frac{1 + \frac{-3141}{64000}}{1 - \frac{-3141}{64000}} \qquad z_{p1} = 0.9064189 \qquad z_{z1} = 1$$

The transfer function of the first order digital filter:

$$H(z) = G \times \frac{z - z_{z1}}{z - z_{p1}}$$

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Gain parameter G has to be calculated so the digital filter has unity gain at infinite frequency:

$$H(z)\Big|_{z=-1} = G \times \frac{-1 - z_{z1}}{-1 - z_{p1}} = 1 \quad \text{thus } G = \frac{1 + z_{p1}}{2}$$

G = 0.9532094

$$H(z) = 0.9532094 \times \frac{z - 1}{z - 0.9064189} = \frac{0.9532094 \times z - 0.9532094}{z - 0.9064189}$$

The filter's coefficients have to be converted to 14-bit integer numbers. The conversion is done by multiplying the fractional coefficients of the transfer function by 2^{14} and rounding the results to the nearest integer number. The rounded integers can then be entered into one of the biquad sections in the Interactive Data Sheet and programmed into the PARAGON™ DIGITAL hybrid:

COEFFICIENT	REAL VALUE	14-BIT INTEGER VALUE
b0	0.9532094	15617
b1	-0.9532094	-15617
b2	0	0
a0	1	16384
a1	-0.9064189	-14851
a2	0	0

Example 3: 2nd order low pass filter with 3000Hz corner frequency and 0dB gain

The transfer function of a 2nd order analog low pass filter with 0dB gain:

$$H(s) = \frac{\omega_c^2}{s^2 + 2 \times \zeta \times \omega_c \times s + \omega_c^2}$$

ζ = damping ratio (0 to 1)
$\omega_c = 2\pi f_c$, f_c = corner frequency

The transfer function has 2 poles located at

$s_{p1,p2} = -\zeta \times \omega_c \pm j \times \omega_c \times \sqrt{1-\zeta^2}$ and 2 zeros $s_{z1} = s_{z2}$ at ∞

The corner frequency of the 2nd order analog filter f_c = 3000Hz therefore ω_c = 18850. For the 2nd order filter to have -3dB at the corner frequency the damping ratio should be set to:

$$\zeta = \frac{\sqrt{2}}{2} = 0.7071068$$

The transfer function of the analog low pass filter:

$$H(s) = \frac{3.553 \times 10^8}{s^2 + 2.6657 \times 10^4 \times s + 3.553 \times 10^8}$$

the poles $s_{p1,p2} = -13329 \pm j \times 13329$,
the zeros $s_{z1} = s_{z2} = \infty$

Using the bilinear transformation, sampling frequency f_s = 32000Hz:

$$z_{p1} = \frac{1 + \frac{(-13329 + j \times 13329)}{64000}}{1 - \frac{(-13329 + j \times 13329)}{64000}}$$

$$z_{p1} = 0.6075147 + j\,0.2770771$$

$$z_{p2} = \frac{1 + \frac{(-13329 - j \times 13329)}{64000}}{1 - \frac{(-13329 - j \times 13329)}{64000}}$$

$$z_{p2} = 0.6075147 - j\,0.2770771$$

$z_{z1} = z_{z2} = -1$

The second order digital filter transfer function:

$$H(z) = G \times \frac{(z - z_{z1}) \times (z - z_{z2})}{(z - z_{p1}) \times (z - z_{p2})}$$

$$H(z) = G \times \frac{z^2 + 2 \times z + 1}{z^2 - 1.2150293 \times z + 0.4458458}$$

Gain parameter G has to be calculated so the digital filter has unity gain at DC:

$$H(z)\Big|_{z=1} = G \times \frac{(1 - z_{z1}) \times (1 - z_{z2})}{(1 - z_{p1}) \times (1 - z_{p2})} = 1$$

$$\text{thus } G = \frac{(1 - z_{p1}) \times (1 - z_{p2})}{4}$$

G = 0.0577041

The transfer function of the digital filter:

$$H(z) = \frac{0.0577041\, z^2 + 0.1154082\, z + 0.0577041}{z^2 - 1.2150293 \times z + 0.4458458}$$

The filter's coefficients have to be converted to 14-bit integer numbers. The conversion is done by multiplying the fractional coefficients of the transfer function by 2^{14} and rounding the results to the nearest integer number. The rounded integers can then be entered into one of the biquad sections in the Interactive Data Sheet and programmed into the PARAGON™ DIGITAL hybrid:

COEFFICIENT	REAL VALUE	14-BIT INTEGER VALUE
b0	0.0577041	945
b1	0.1154082	1891
b2	0.0577041	945
a0	1	16384
a1	-1.2150293	-19907
a2	0.4458458	7305

Example 4: 2nd order high pass filter with 1000Hz corner frequency and 0dB gain

The transfer function of a 2nd order analog high pass filter with 0dB gain:

$$H(s) = \frac{s^2}{s^2 + 2 \times \zeta \times \omega_c \times s + \omega_c^2}$$

ζ = damping ratio (0 to 1)
$\omega_c = 2\pi f_c$, f_c = corner frequency

The transfer function has 2 poles located at $s_{p1,p2} = -\zeta \times \omega_c \pm j \times \omega_c \times \sqrt{1-\zeta^2}$ and 2 zeros $s_{z1} = s_{z2}$ at 0

The corner frequency of the 2nd order analog filter f_c = 1000Hz therefore ω_c = 6283

For the 2nd order filter to have -3dB at the corner frequency the damping ratio should be set to:

$$\zeta = \frac{\sqrt{2}}{2} = 0.7071068$$

The transfer function of the analog high pass filter:

$$H(s) = \frac{s^2}{s^2 + 8.8858 \times 10^3 \times s + 3.9478 \times 10^7}$$

the poles $s_{p1,p2} = -4443 \pm j \times 4443$, the zeros $s_{z1} = s_{z2} = 0$

Using the bilinear transformation, sampling frequency f_s = 32000Hz:

$$z_{p1} = \frac{1 + \frac{(-4443 + j \times 4443)}{64000}}{1 - \frac{(-4443 + j \times 4443)}{64000}}$$

$$z_{p1} = 0.86232501 + j \times 0.1208905$$

$$z_{p2} = \frac{1 + \frac{(-4443 - j \times 4443)}{64000}}{1 - \frac{(-4443 - j \times 4443)}{64000}}$$

$$z_{p2} = 0.86232501 - j \times 0.1208905$$

$z_{z1} = z_{z2} = 1$

The second order digital filter transfer function:

$$H(z) = G \times \frac{(z - z_{z1}) \times (z - z_{z2})}{(z - z_{p1}) \times (z - z_{p2})}$$

$$H(z) = G \times \frac{z^2 - 2 \times z + 1}{z^2 - 1.7246502 \times z + 0.7582191}$$

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Gain parameter G has to be calculated so the digital filter has unity gain at infinity:

$$H(z)\Big|_{z=-1} = G \times \frac{(-1 - z_{z1}) \times (-1 - z_{z2})}{(-1 - z_{p1}) \times (-1 - z_{p2})} = 1$$

$$\text{thus } G = \frac{(-1 - z_{p1}) \times (-1 - z_{p2})}{4}$$

G = 0.8707173

The transfer function of the digital filter:

$$H(z) = \frac{0.8707173 \times z^2 - 1.7414346 \times z + 0.8707173}{z^2 - 1.7246502 \times z + 0.7582191}$$

The filter's coefficients have to be converted to 14-bit integer numbers. The conversion is done by multiplying the fractional coefficients of the transfer function by 2^{14} and rounding the results to the nearest integer number. The rounded integers can then be entered into one of the biquad sections in the Interactive Data Sheet and programmed into the PARAGON™ DIGITAL hybrid:

COEFFICIENT	REAL VALUE	14-BIT INTEGER VALUE
b0	0.8707173	14266
b1	-1.7414346	-28532
b2	0.8707173	14266
a0	1	16384
a1	-1.7246502	-28257
a2	0.7582191	12423

DOCUMENT IDENTIFICATION
INFORMATION NOTE
The product is in a development phase and specifications are subject to change without notice. Gennum reserves the right to remove the product at any time. Listing the product does not constitute an offer for sale.

REVISION NOTES:
Corrections to errors in filter design examples.

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