DESIGN OF AN AUDIO DAC

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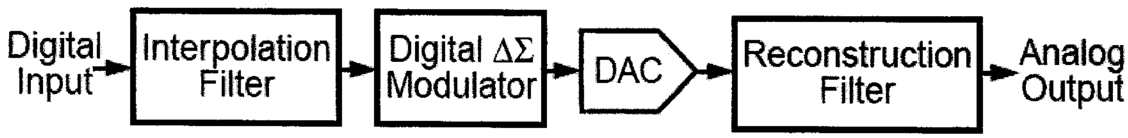
EXAMPLE: AUDIO DAC [1]

Specifications:

Parameter	Symbol	Value	Units
Input Sample Rate	$f_{s,in}$	44.1	kHz
Signal Bandwidth	f_{B0}	20	kHz
Output Signal-to-Noise Ratio	SNR	110	dB
Passband Flatness		0.1	dB
Image Attenuation		>90	dB
Modulator Bandwidth	$f_B = 2\,f_{s,in}$	88	kHz
Modulator Sample Rate	$f_s = 256\,f_{s,in}$	11.29	MHz
Modulator OSR	$OSR = f_s/(2f_B)$	64	
Number of Quantizer Steps	M	8	

DESIGN APPROACH

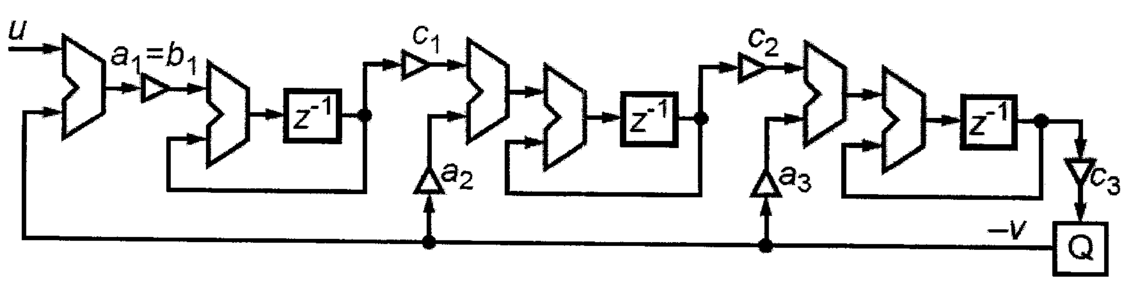
System:



To relax the analog filter specs, quadruple the passband width of the ΔΣ modulator to keep quantization noise away from f_{B0}.

MODULATOR DESIGN

3rd – order, 9-level loop gives an SNR = 115 dB over a 0-88 kHz band.



All NTF zeros at dc.

TOOLBOX CODE [2]

```
form = 'CIFB';
[a,g,b,c]=realizeNTF(NTF,form);
b(2:end) = 0;
ABCD= stuffABCD(a,g,b,c,form);
ABCDs = scaleABCD(ABCD,M+1);
[a,g,b,c]= mapABCD(ABCDs,form);
% a = [0.0331 0.0807 0.1626]
% g = 0
% b = [0.0331 0 0 0]
% c = [0.6176 1.3672 10.4879]
```

```
% scale for a(1) = 1/32
k = a(1) * 32;
a(1) = a(1)/k;
b(1) = b(1)/k;
c(1) = c(1)*k;
% scale for c(1) = 0.5
k = c(1)/0.5;
c(1) = c(1)/k;
a(2) = a(2)/k;
c(2) = c(2)*k;
% scale for c(2) = 1
k = c(2)/1;
c(2) = c(2)/k;
a(3) = a(3)/k;
c(3) = c(3)*k;
```


QUANTIZED COEFFICIENTS

Coefficient	Original Value	Transformed Value	Quantized Value	% Error
$a_1 = b_1$	0.0331	0.0312	$\frac{1}{32}$	0
a_2	0.0807	0.0617	$\frac{1}{16}$	1
a_3	0.1626	0.0909	$\frac{1}{16}+\frac{1}{32}$	3
c_1	0.6176	0.5000	$\frac{1}{2}$	0
c_2	1.3672	1.0000	1	0
c_3	10.4879	18.7539	16+2	–4

WORD LENGTH IN THE LOOP

For -130 dBFS truncation noise, the necessary word lengths can be found from the input-referred noise powers.

First integrator:

$$\frac{(2^{-N_1})^2}{3a_1^2 OSR} < 10^{-13}(M^2/2)$$

This leads to $N_1 = 21$.

Second integrator:

$$N_2 > -\log_2(Ma_1c_1\sqrt{0.45\times10^{-13}(OSR)^3}) = 16.2$$

This gives $N_2 = 17$.

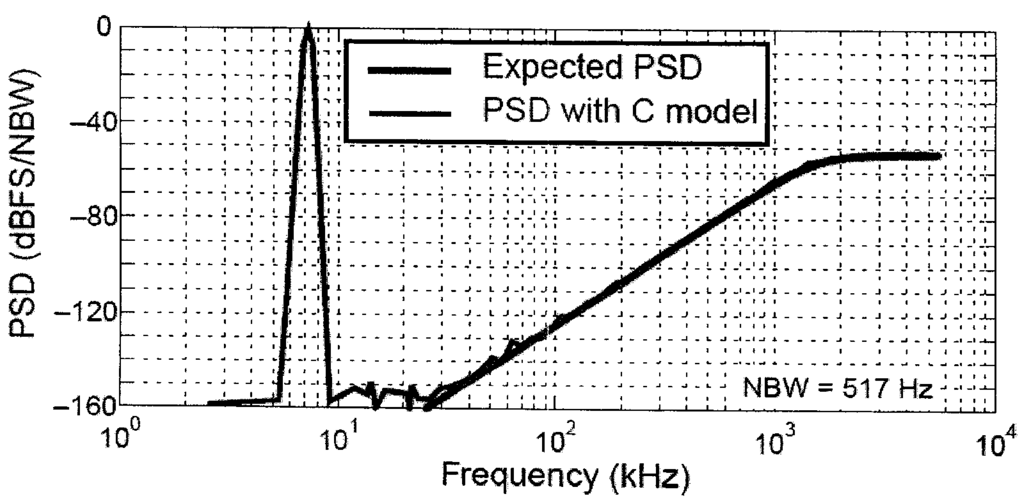
Third integrator:

$$N_3 > -\log_2(Ma_1c_1c_2\sqrt{0.077\times10^{-13}(OSR)^3}) = 11.4$$

This gives $N_3 = 12$.

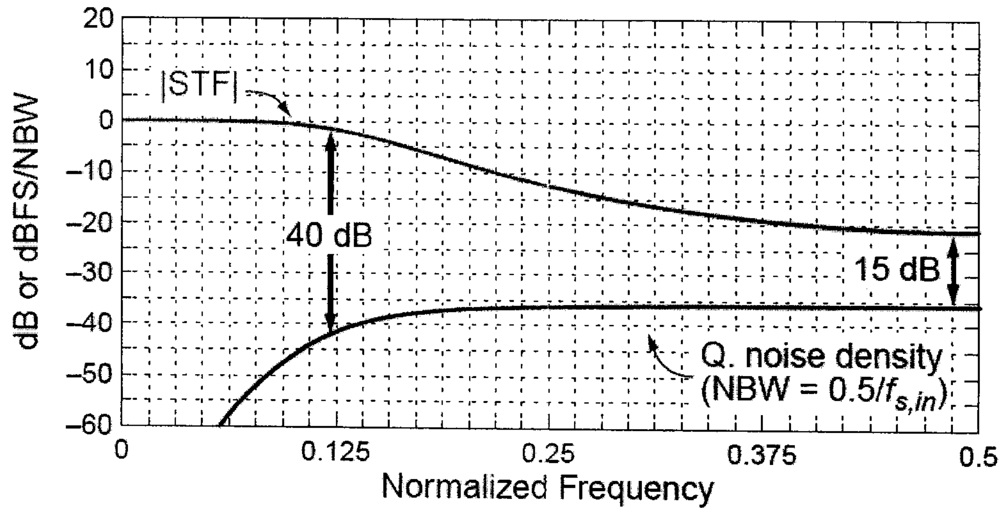
SIMULATED PSD

For a -1 dBFS input, the truncation noise over 0-20 kHz is -137 dBFS.



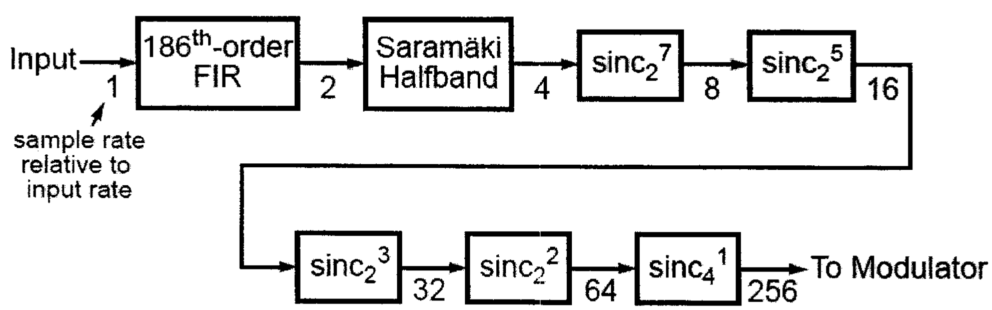
INTERPOLATION FILTER DESIGN

It reduces the unneeded spectral replicas between f_{B0} and $f_s - f_{B0}$, to a level below the quantization noise Q_N. Since the modulator STF also contributes attenuation, the filter needs to provide $-|Q_N|^2$(dB) + $|STF|^2$(dB). Near f_B, provide 90 dB.



FILTER STRUCTURE

A cascade of interpolate – by – 2 sections:



Here, the sinc_N^k filter has the transfer function

$$H(z) = \left[\frac{(1 - z^{-N})}{N(1 - z^{-1})}\right]^k$$

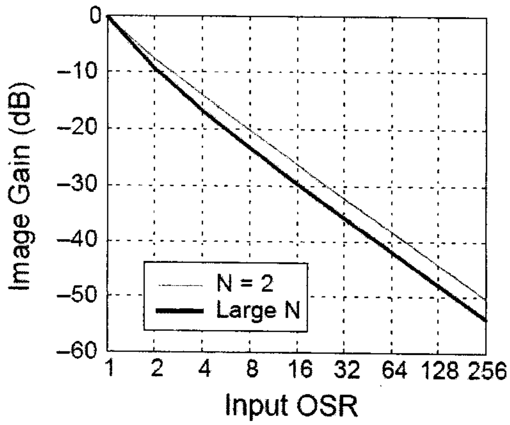
where z refers to the output f_s.

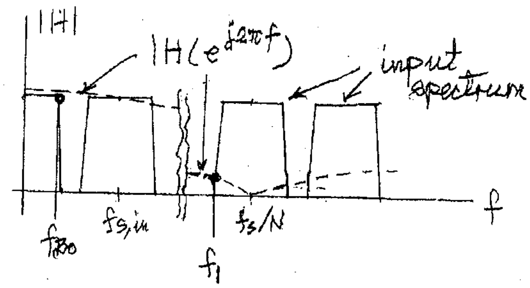
IMAGE GAIN

Defined as:

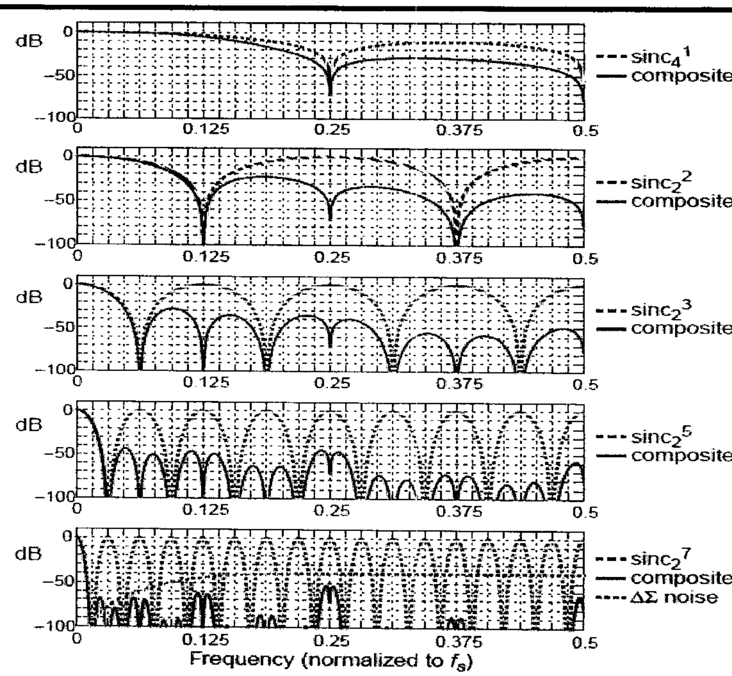
$$G = \frac{H(e^{j2\pi f_1})}{H(e^{j2\pi f_{B0}})}$$

where $f_1 = (f_s / N) - f_{B0}$. This is a function of N and $OSR_{in} = \frac{f_s/N}{2f_{B0}}$





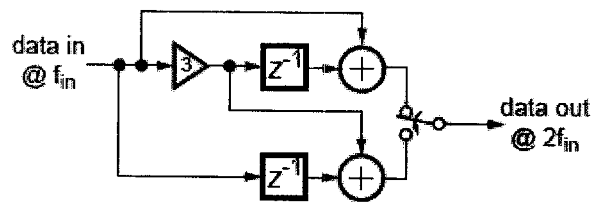
SINC FILTER PERFORMANCE



SINC FILTER COMPLEXITY

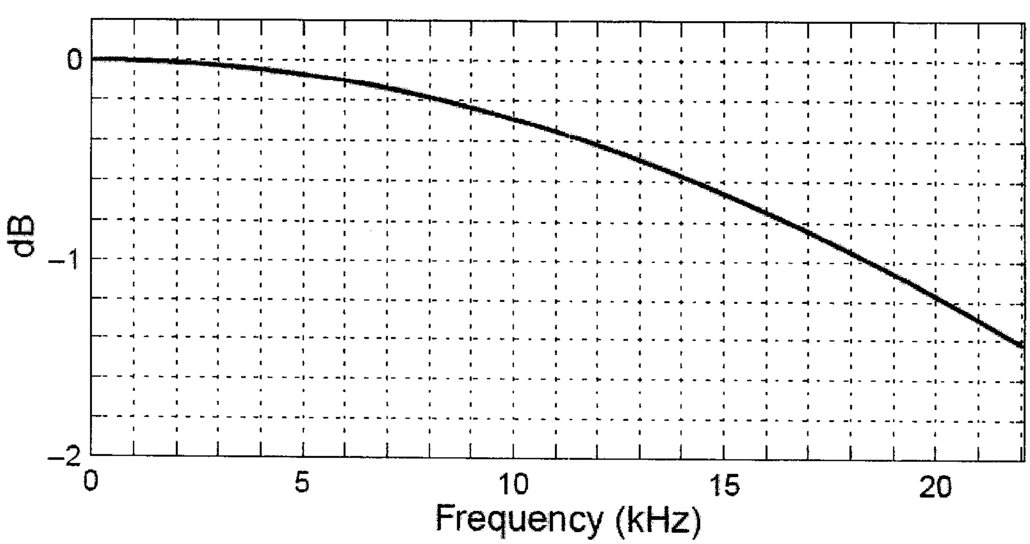
Parameter	$sinc_2^7$	$sinc_2^5$	$sinc_2^3$	$sinc_2^2$	$sinc_4^1$
Output Sample Rate	$8\ f_{s,in}$	$16\ f_{s,in}$	$32\ f_{s,in}$	$64\ f_{s,in}$	$256\ f_{s,in}$
Sinc order	7	5	3	2	1
Number of Additions at the output sample rate (= Number of registers)	6	4	2	1	0
Number of Additions at $f_{s,in}$	48	64	64	64	0

Polyphase implementation of $sinc_2^3$:



The modulator needs 2,048 additions at $f_{s,in}$!

SINC FILTERS' PASSBAND



May be equalized in the first two stages.

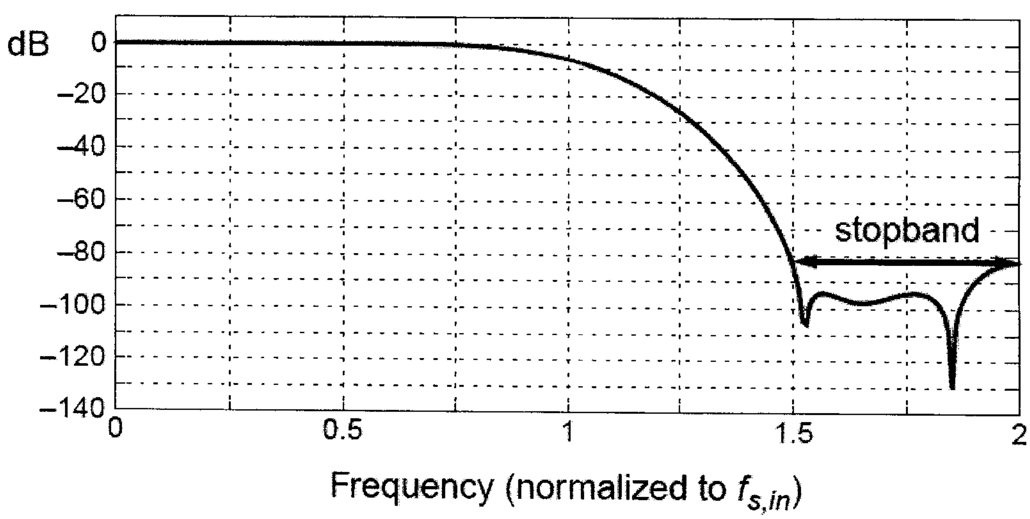
SECOND FILTER STAGE

Saramaki halfband FIR filter [2][3]:

```
[f1,f2,info] = designHBF(0.125,undbv(-70));
figure(1); clf
f = linspace(0,0.5,256);
plot(f*4, dbv(frespHBF(f,f1,f2)),'b','Linewidth',1.5)
figureMagic([0 2],0.25,2, [-140 10],10,2);
printmif('HBF_freq', [5 2.5], 'Helvetica10')

N = (2*length(f1)-1)*2*(2*length(f2)-1)+1;
y = simulateHBF([1 zeros(1,N-1)],f1,f2);
stem([0:N-1],y);
figureMagic([0 N-1],5,2, [-0.2 0.5],0.1,1)
printmif('HBF_imp', [5 2], 'Helvetica10')
```


SARAMAKI FILTER RESPONSE



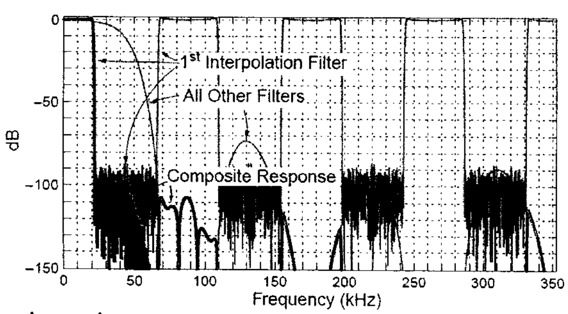
82 dB image attenuation. It requires 44 additions, 50 registers. Halfband filter (symmetric response, fewer tap weights).

INPUT FILTER STAGE

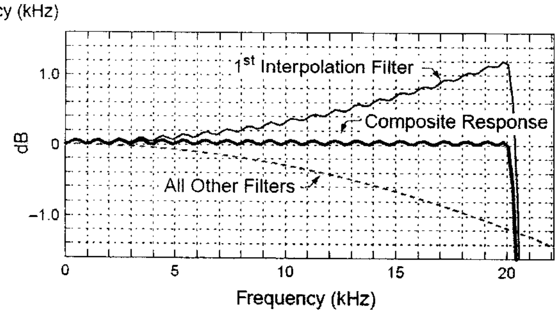
- Passband 0 – 20 kHz, stopband suppression -90dB from $f_{s,in}/2$ to $2f_{s,in} - f_{B0}$,
- Cannot be halfband; stopband starts at $f_{s,in}/2$.
- Last designed, so it can equalize the droop of the following stages.
- Usually, the stopband only starts at $f_{s,in} - f_{B0}$ (here, 24.1 kHz instead of 22.05 kHz). Much less complex!
- MATLAB's remez function can be used to design it: 186^{th}-order FIR, 445 additions, 186 registers, 60% of filter power dissipation.

FREQUENCY RESPONSES

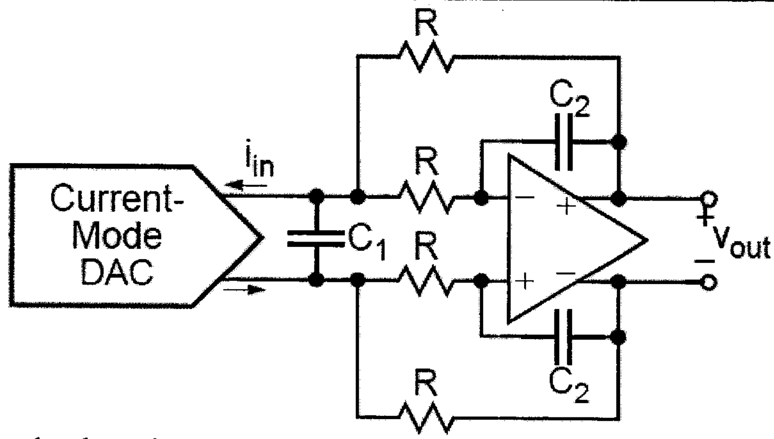
Overall response:



Passband response:



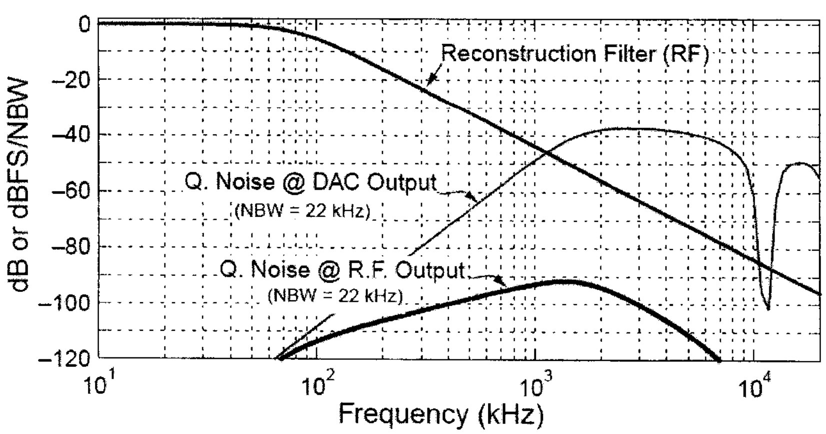
DAC AND RECONSTRUCTION FILTER



Transfer function:

$$\frac{V_{out}(s)}{I_{in}(s)} = \frac{2R}{\left(\frac{s}{\omega_0}\right)^2 + \left(\frac{s}{\omega_0 Q}\right) + 1} \quad \text{where} \quad Q = \sqrt{\frac{C_1}{2C_2}} \qquad \omega_0 = \frac{1}{\sqrt{2C_1C_2}R}.$$

FILTER DESIGN



- For -90 dBFS / 22 kHz noise PSD, a Butterworth response with f_{3db} = 80 kHz may be used. Then, $Q = 1/\sqrt{2}$ and $C_1 = C_2 = C$, $RC \approx 1.41 \times 10^{-6}$ s.
- Output noise PSD is 16 kTR at low frequencies. For SNR = 116 dB, R = 1 kΩ and C = 1.41 nF may be used.
- Then for a FS $V_{out} = 0.7\ V_p$, $i_{in} = 0.35\ mA_p$. 8 unit elements, each 39 μA.

REFERENCES

[1] R. Schreier and G. C. Temes, "Understanding ΔΣ Data Converters", *IEEE Press*, 2004, Sec. 9.6.

[2] Op. cit., Appendix B.

[3] T. Saramäki, "Design of FIR filters as a tapped cascaded interconnection of identical subfilters," *IEEE Transactions on Circuits and Systems*, vol.34, no. 9, pp. 1011-1029, Sept. 1987.