

# A 100 $\mu$ W Decimator for a 16 bit 24 kHz bandwidth Audio $\Delta\Sigma$ Modulator

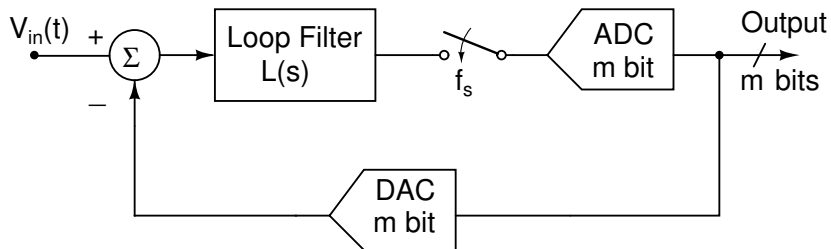
Shankar Parameswaran,  
Nagendra Krishnapura

Department of Electrical Engineering  
Indian Institute of Technology, Madras  
Chennai, 600036, India

2010 International Symposium on Circuits and Systems,  
Paris, France

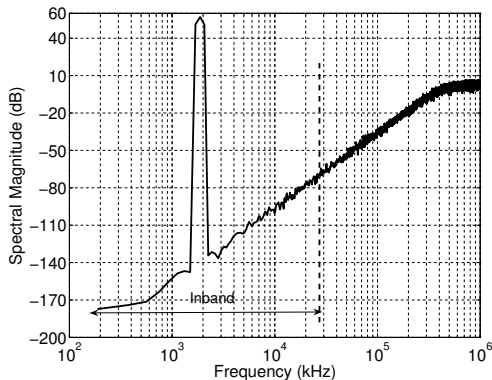
- Brief overview of  $\Delta\Sigma$  Modulators & Decimators
- Architectural optimizations in decimator to reduce power
- Simulation results
- Fabricated chip results
- Conclusions

# Block diagram - Continuous Time $\Delta\Sigma$ Modulator



- Oversampling
- Noise shaping
- Low resolution internal ADC

# Spectrum of the $\Delta\Sigma$ Modulator output



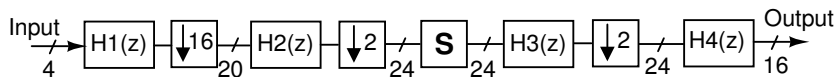
- 3<sup>rd</sup> order L(s) , 4 bit ADC,  $f_s = 3.072$  MHz
- Inband Signal to noise ratio (SNR) = 96 dB
- Decimator - Low pass filtering & downsampling

# Decimator requirements

Modulator[1]		Decimation Filter	
Order	3	Downsampling Factor	64
Sampling rate	3.072 MHz	Passband ripple	0.05 dB
Nyquist rate	48 kHz	Passband edge	21.6 kHz
SNR	93 dB	SNR	96 dB
Power	90 $\mu$ W	Power	< 100 $\mu$ W

[1] - S. Pavan, N. Krishnapura, R. Pandarinathan, and P. Sankar, "A power optimized continuous-time  $\Delta\Sigma$  converter for audio applications," *IEEE Journal of Solid-State Circuits*, vol. 43, no. 2, pp. 351-360, 2008

# Block diagram of the decimator



$H_1(z)$ : SINC4

$H_2(z)$ : First Halfband filter

$H_3(z)$ : Second Halfband filter

$H_4(z)$ : Equalizer

**S** : Scaling block

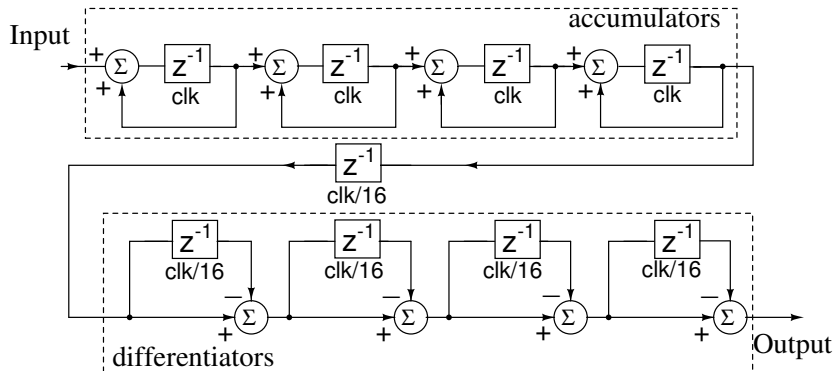
- Multistage decimation

- SINC,  $H(z)$  - 16 tap moving average filter
- SINC4,  $H^4(z)$  - Cascade of 4 SINC filters
- Removes quantization noise shaping
- Downsampling of 16 - Hogenauer structure

$$H(z) = \frac{1 - z^{-16}}{1 - z^{-1}}$$

$$H(f) = \frac{\sin(16\pi f)}{\sin(\pi f)}$$

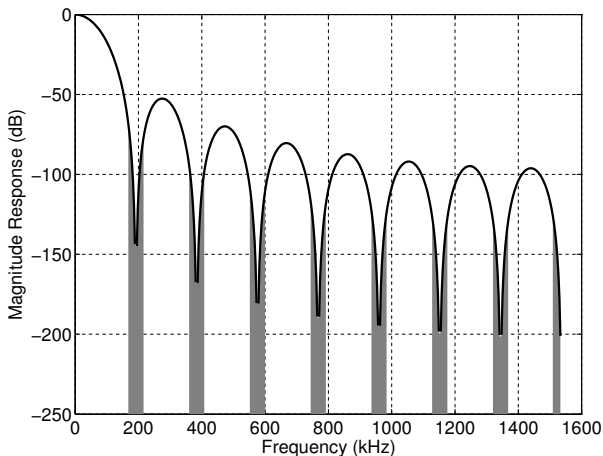
# SINC4 - Hogenauer structure



- Retiming, Pipelining save 46% power in SINC4
- Optimal Datawidth =  $B_{in} + k \log_2 N = 20$



# Frequency response of SINC4



Shaded Area - Noise aliasing bands

# Halfband filters

- FIR filters
  - 6 dB bandwidth =  $\frac{f_s}{4}$
  - Alternate tap weights are 0
- Two halfband filters downsample by 2 each
- First halfband filter
  - Initial filtering
  - 10<sup>th</sup> order
- Second halfband filter
  - Sharper filtering
  - 50<sup>th</sup> order

# Halfband filter implementation

- Polyphase structure, downsampling by 2 within filter
- Tap weights in Canonical Signed Digits (CSD)

$$\text{Example: } 0.875 = 2^{-1} + 2^{-2} + 2^{-3} = 2^0 - 2^{-4}$$

(Reduces Multiplication Complexity)

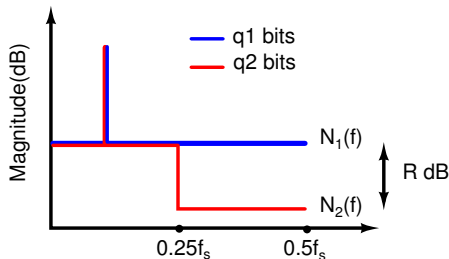
- Nested Multiplication, Horner's rule

$$\text{Example: } 2^{-5} - 2^{-7} = 2^{-5}(1 - 2^{-2})$$

(Reduces Truncation Error)

- $2^{-1}$  is dropping a bit in the multiplicand

# Datawidth in halfband filters



$$N_1(f) = 10^{-\frac{6q_1}{10}}$$

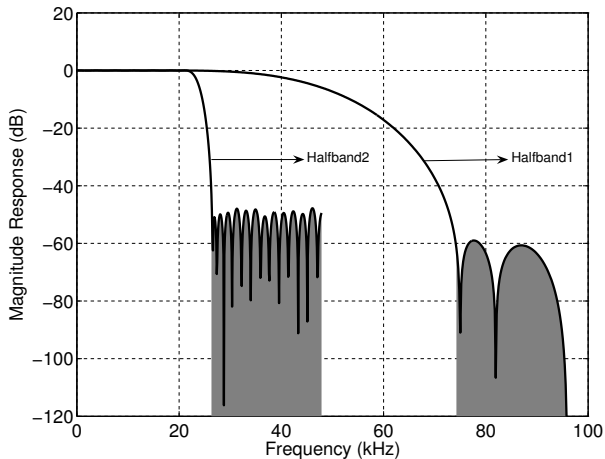
$$N_2(f) = 10^{-\frac{6q_2}{10}}$$

$$R = 10 \log \frac{N_2}{N_1}$$

$$q_2 = q_1 + \frac{R}{6}$$

To attenuate a portion of quantization noise floor of a  $q_1=16$  bit signal by 48 dB in some band needs  $q_2=24$  bit in the filter.

# Frequency response of halfband filters



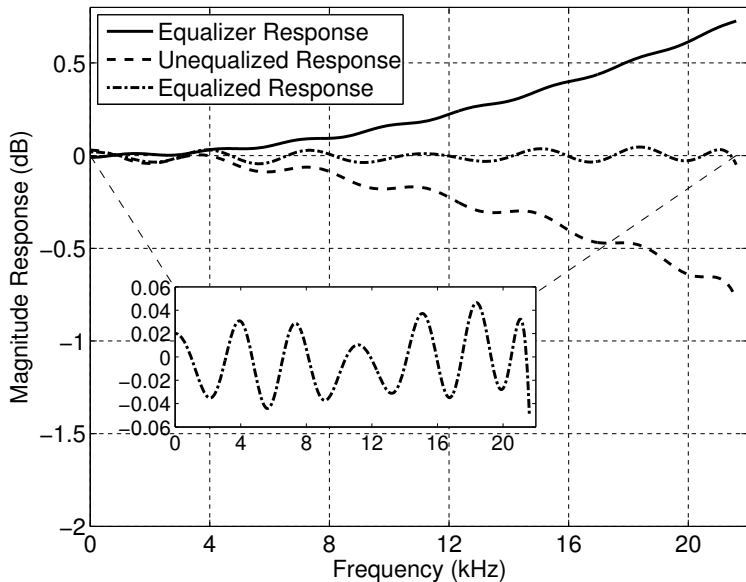
Shaded Area - Noise aliasing bands

# Scaling block

- $\Delta\Sigma$  Modulator - Maximum Stable Amplitude (MSA=85%)
- Signal swing at modulator output is  $MSA \times \text{fullscale}$
- 96 dB SNR at Nyquist rate only with fullscale amplitude
- Scale by  $MSA^{-1}$  after first halfband filter
- Lesser number of bits from modulator (4) and high frequency noise prevents scaling at the initial stages of decimator
- CSD & Nested Multiplication

- Droop of SINC4 in the passband
- Maximum passband ripple = 0.05 dB
- Inverse SINC4 designed with *Parks McClellan* method
- 48<sup>th</sup> order filter
- CSD & Nested Multiplication

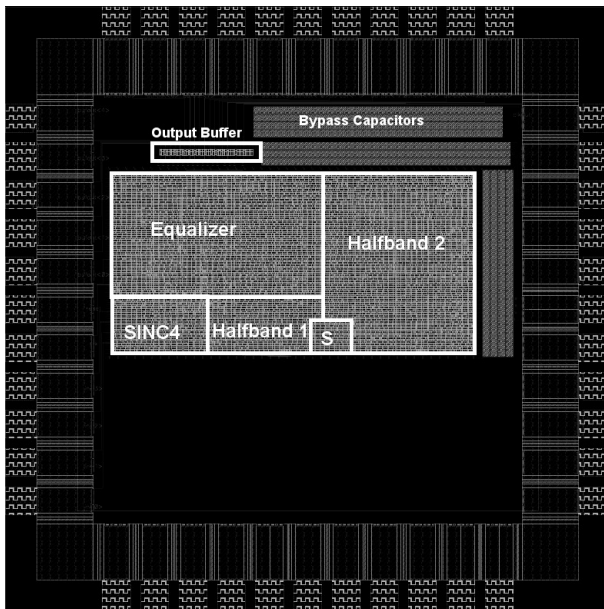
# Frequency response of the equalizer



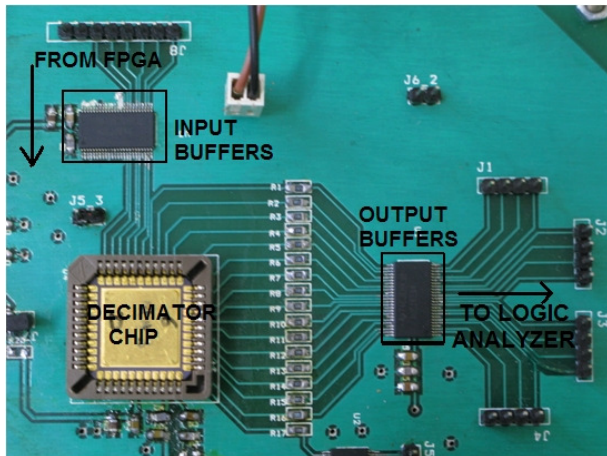


- Technology: 1.8 V standard cells UMC 0.18  $\mu\text{m}$  CMOS
- CAD tools
  - Design Synthesis - Design Compiler
  - Place & Route - SoC Encounter
- Power Consumption - PrimePower
  - 96.7  $\mu\text{W}$  -  $\Delta\Sigma$  modulator input is a tone at 1.6 kHz
  - 100  $\mu\text{W}$  -  $\Delta\Sigma$  modulator input is a white noise
- Active area = 0.46  $\text{mm}^2$

# Layout of the fabricated chip



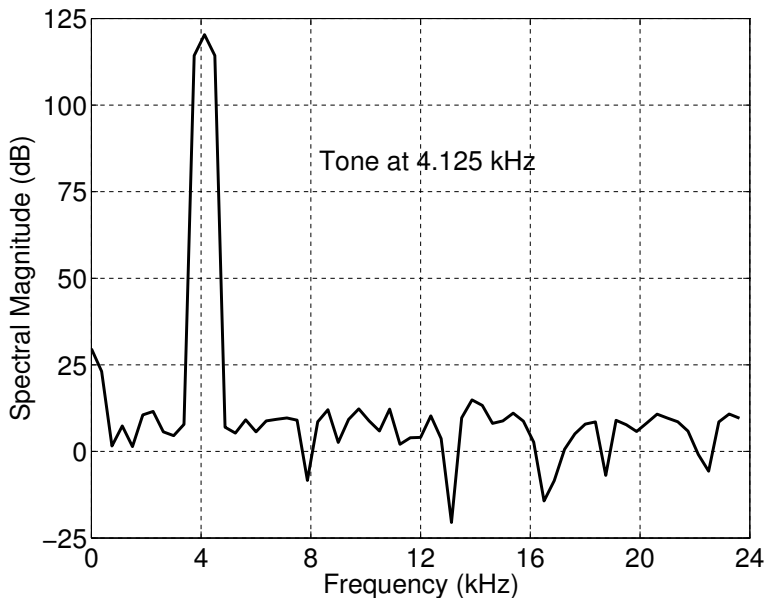
# Picture of the test board



# Measurement results from the chip

- Chip works correctly at 1.8 V supply and works reliably down to 0.9 V at 3.072 MHz
- Current consumed with 1.8 V supply =  $58 \mu\text{A}$  ( $104.4 \mu\text{W}$ )
- Current consumed with 0.9 V supply =  $26 \mu\text{A}$

# Decimated output spectrum from the chip



# Conclusion

- 100  $\mu$ W decimator for audio  $\Delta\Sigma$  ADC
- No handcrafted circuits - completely implemented with automated CAD tools
- Works down to 0.9 V supply
- 50% power reduction with a linear power regulator

# References



S. Pavan, N. Krishnapura, R. Pandarinathan, and P. Sankar, "A power optimized continuous-time  $\Delta\Sigma$  converter for audio applications," *IEEE Journal of Solid-State Circuits*, vol. 43, no. 2, pp. 351-360, 2008



S. Pavan and P. Sankar, "A 110  $\mu$ W Single Bit Audio Continuous-time Oversampled Converter with 92.5 dB Dynamic Range" *Proceedings of the European Solid State Circuits conference*, September 2009.



Ronald E. Crochiere, Lawrence R. Rabiner, "Interpolation and Decimation of Digital Signals - A Tutorial Review," *Proceedings of the IEEE* vol. 69, NO. 3, March 1981.



Carol J. Barrett, *Low-power decimation filter design for multi-standard transceiver applications*, Master of Science Thesis, University of California, Berkeley.



James C. Candy, "Decimation for Sigma Delta Modulation," *IEEE transactions on communications*, vol. com-34, no. 1. January 1986.



Eugene B. Hogenauer, "An economical class of digital filters for decimation and interpolation," *IEEE transactions on acoustics, speech, and signal processing*, vol. assp-29, no. 2, April 1981.



Keshab K. Parhi, *VLSI Digital Signal Processing Systems: Design and Implementation*, John Wiley & Sons, 1999.



Reid M. Hewlitt, Earl S. Swartzlander, Jr., "Canonical signed digit representation for FIR digital filters," *IEEE Workshop on Signal Processing Systems, 2000, SiPS 2000*.