## Chapter 13: Communication Systems Applications

13.1 Conventional Digital Down Converters  
13.2 Aliasing Digital Down Converters  
   13.2.1 IF Subsampling Example  
13.3 Timing Recovery in a Digital Demodulator  
13.4 Modem Carrier Recovery  
   13.4.1. Background  
   13.4.2. Modern Carrier Recovery  
13.5 Digitally Controlled Sampled Data Delay  
13.6 Interpolated Shaping Filter  
13.7 Sigma-delta Decimating Filter  
   13.7.1 Sigma-delta Filter  
13.8 FM Receiver and Demodulator  
   13.8.1 FM Band Channelizer  
   13.8.2 FM Demodulator  
   13.8.3 Stereo Decoding  

13.1 Conventional Digital Downconverters

- Conventional receiver with digital baseband demodulation
  - ADC at moderate sample rates (12kHz to 4 MHz) prior to baseband processing
  - IF filter sets receiver BW
  - Quadrature downconversion for complex processing
  - LPF for anti-aliasing

Figure 13.1 Standard Radio Receiver Architecture
Goals of Digitizing Receivers

• Receiver Limitations
  – Mixing and LPF: passband gain variation and differences, mixing harmonics. Matching analog components is difficult!
  – IF Bandpass filter: uniformity, bandwidth, transition bands
  – ADC Sample Rate: limit digital from moving into IF or higher

• Analog components
  – Must have sufficient precision and accuracy
  – Require tuning for most applications
  – Must be concerned with component coupling (capacitive, inductive)
  – RFI and EMI considerations
Digitized IF

- Higher rate ADC enables digital IF processing
  - ADC at higher sample rates (4 MHz to 64 MHz)
  - IF Bandpass ADC sampling and signal processing
  - Subsampling of band-limited IF possible

Figure 13.2 Second Generation Radio Receiver Architecture
Digital IF Processing

- Commercial Digital Downconverter IC components with multiple half-band filter decimators
  - Dedicated hardware for “repetitive mathematical tasks”
  - Software DSP for lower data rate tasks and “if-then-else” processing

**Figure 13.3** Standard Digital Down Converter with CIC and Half-band Filters
CIC and FIR Correction

- Figure 13.4 Frequency Response of 5-Stage CIC Filter, at Input Sample Rate, at Output Sample Rate Illustrating Main Lobe Folding due to 10-to-1 Resampling, and Main Lobe Response with Overlaid Compensating 4-to-1 Down Sample filter
Matlab Compensation Filter

\[ gg = \left(1 - \frac{1}{6} \theta^2 + \frac{1}{120} \theta^4 - \frac{1}{5040} \theta^6\right)^5 \]  

(13.1)

- Model of required compensation
  - Taylor Series Expansion of \( \sin(\theta) / \theta \) around \( \theta = 0 \)
- Suggested Matlab code to generate filter coefficients

```matlab
phi=[0.00 0.01 0.011 0.02 0.021 0.03 0.031... 0.04 0.041 0.05 0.051 0.06 0.061... 0.07 0.071 0.08 0.081 0.09 0.091 0.10];
phi=phi*pi;
tt=((1-(phi.^2)/6+(phi.^4)/120-(phi.^6)/1540)).^5);
hh=remez(20,[phi 0.4 0.5]/0.5,[1./tt 0 0]);
```

MATLAB Call for CIC Compensating Half-band Filter
Compensation Results

Figure 13.5 Spectra of CIC Main Lobe, of 21-Tap Compensating Half-band Filter, and the Composite Response
Clean-up Filter Examples

- See Chap13_1.m and Chap13_2.m
  - Taylor series expansion
  - Altera approach
Hardware Components

Digital Up and Down Converters using CIC and HBs

• Ettus Corporation USRP devices
  – Xilinx VHDL Software
• Analog Devices AD9856, AD9857, AD6654, AD6636
• Intersil HSP502014B, HSP50216, HSP50415
Analog Devices
VersaCOMM™ Products

VERSACOMM PRODUCT FAMILY
Receive Signal Processors (RSP)
- AD6620
  Single/Dual Channel
  65 MSPS
- AD6624
  Single/Dual/Quad Channel
  80 MSPS
- AD6634
  Dual Channel W-CDMA
  80 MSPS

Transmit Signal Processors (TSP)
- AD6622
  Single/Dual/Quad Channel
  75 MSPS
- AD6623
  Single/Dual/Quad Channel
  104 MSPS

Quadrature Digital Upconverter/Modulator (QDUC)
- AD9853
  5-65 MHz QPSK/16-QAM
  Digital Modulator with 10-bit DAC
- AD9856
  200 MSPS Quadrature
  Digital Upconverter with 12-bit DAC
- AD9857
  200 MSPS Quadrature
  Digital Upconverter with 14-bit DAC

Intersil HSP 50415
Wideband Programmable Modulator
Intersil HSP 50216
Programmable Digital Downconverter

Features
- Up to 70MSPS Input
- Four Independently Programmable Downconverter Channels in a single package
- Four Parallel 16-Bit Inputs - Fixed or Floating Point Format
- 32-Bit Programmable Carrier NCO with >115dB SFDR
- 110dB FIR Out of Band Attenuation
- Decimation from 8 to >65536
- 24-bit Internal Data Path
- Digital AGC with up to 96dB of Gain Range
- Filter Functions
  - 1 to 5 Stage CIC Filter
  - Halfband Decimation and Interpolation FIR Filter
  - Programmable FIR Filter
  - Resampling FIR Filter
- Cascadable Filtering for Additional Bandwidth
- Four Independent Serial Outputs
- 3.3V Operation
- Pb-Free Available (RoHS Compliant)

Applications
- Narrow-Band TDMA through IS-95 CDMA Digital Software Radio and Basestation Receivers
- Wide-Band Applications: W-CDMA and UMTS Digital Software Radio and Basestation Receivers

13.2 Aliasing Digital Downconverter (Bandpass Sampling)

\[ f_S = 2f_{BW} + f_{\Delta} \]  \hspace{1cm} (13.2)

Signals collected from the output of the anti-alias filter at this data rate are said to satisfy the Nyquist sampling criterion. The Nyquist criterion is sometimes stated as: “The sample rate must be greater than twice the highest frequency of the input signal.”

This is an over restrictive or a narrow interpretation of the Nyquist criterion. The less restrictive interpretation is that the sample rate must exceed the two-sided bandwidth of the signal. This second interpretation is important when we have a narrow bandwidth signal centered on a high frequency carrier.

The bandpass sampling theorem states that the Nyquist rate must only be maintained for the passband.
13.2.1 Subsampling ADC

When we have a narrow bandwidth signal centered on a high frequency carrier. In a digital receiver, the signal processing following the data collection process removes the carrier to extract the complex envelope of the narrowband signal on the carrier. A digital down conversion process normally performs this task. Since the carrier frequency is discarded as part of the signal extraction, there is no need to preserve it during the data sampling process.

We are thus free to “violate” the LPF Nyquist criterion for the carrier frequency as long as we satisfy the criterion for the bandwidth of its complex envelope.

For convenience, the sampling rate is also selected based on the signals carrier frequency, such that:

\[ f_C = k f_s \pm \frac{1}{4} f_s \quad (13.3) \]
“Bandpass” Nyquist Criteria

\[ f_c = kf_s \pm \frac{1}{4} f_s \quad (13.3) \]

To minimize the cost of the analog signal-conditioning filter we arrange for the signal band of interest to alias to one-fourth of the selected sample rate.

Aliasing to the quarter sample rate maximizes the separation between the positive frequency alias and the negative frequency alias, which permits the maximum transition bandwidth of the analog band-pass filter. Aliasing the center frequency \( f_c \) to the quarter sample rate during the sampling process is assured if the sample rate satisfies (13.3).

The \( k+1/4 \) option aliases the signal to the positive quarter sample rate while the \( k-1/4 \) option aliases the signal to the negative quarter sample rate. Use of the two options simply makes available a larger set of possible sample rates with either option equally acceptable. Note that \( -1/4, k+1 \) and \( +3/4, k \) are equivalent.
IF Subsampling Example

- Signal 2-Sided Bandwidth 10 kHz
- Center Frequency 450 kHz
- Signal Dynamic Range 80 dB
- Output Sample Rate 20 kHz

\[ f_c = k \cdot f_s \pm \frac{1}{4} \cdot f_s \]

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\( f_c \) | 450 kHz

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IF Subsampling

- Usually some “nice values” pop out
  - 24 kHz, 40 kHz, 72 kHz, 120 kHz, and 200 kHz
  - Decimation rates to get to 20 kHz
    - 1.2, 2, 3.6, 6, and 10 (prefer integers)

- 40 kHz (decimate by 2)
  - Close to output rate, but may complicate digital filters

- 120 kHz (decimate by 6)
  - Nice, spectral inversion?!

- 200 kHz (decimate by 10)
  - Authors favorite and his example
Subsampling Figures

Figure 13.6 Aliasing Spectra at 2.25 fs in Second Nyquist Zone to 0.25 fs in Zeroth Nyquist Zone by IF Sampling at fs

Figure 13.7 Periodic Spectra on Scaled Circle Showing how 450 kHz From Second Nyquist Zone Folds through 250 kHz in First Nyquist Zone to 50 kHz in Zeroth Nyquist Zone

Aliasing at fs=200 kHz
Moving $fs/4$ to Baseband

- Complex mixing the signal to baseband and applying a low pass filter (half-band) creates a complex output for signal processing.

$$\exp(- j \cdot 2\pi \cdot f \cdot t) = \exp\left(- j \cdot 2\pi \cdot \left(\frac{f_s}{4}\right) \cdot \left(\frac{n}{f_s}\right)\right)$$

$$\exp(- j \cdot 2\pi \cdot f \cdot t) = \exp\left(- j \cdot \frac{\pi}{2} \cdot n\right) = \cos\left(\frac{\pi}{2} \cdot n\right) - j \cdot \sin\left(\frac{\pi}{2} \cdot n\right)$$

$$\exp(- j \cdot 2\pi \cdot f \cdot t) = (- j)^n = \{1 \quad -j \quad -1 \quad j \quad 1 \quad -j \quad -1 \quad j \quad \cdots\}$$
Half-Band Complex LPF with Decimation

- For a complex signal, the output sample rate is at least 2x the Nyquist rate. Therefore, the half-band with decimation approach would always pass the real component (the center tap of the HB filter) and filter the complex components (odd coefficients).
Text Approach #1

- High sample rate, quadrature downconversion to baseband, CIC and two half-band filters. The old way …
Text Approach 2

- ADC 200 kHz, subsampling or bandpass sampling
- Real-to Complex Polyphase filter-decimate by 4
- Rational Rate Change by 5/2
Figure 13.10 Signal Processing Structure of Down Converter Using Aliasing to and from Quarter Sample Rate by Subsampling and Polyphase Filtering
Text Approach 2

- Polyphase filter-decimate by 4
- Rational Rate Change by 5/2

Figure 13.8 Four Path Polyphase Filter: Simultaneously Translates Frequency Bar From Quarter Sample Rate to Baseband, Down Samples 4-to-1, and Converts Real Input to Complex Output

Figure 13.9 Indexing of Successive Zero-packed Data Samples to 5-Path Polyphase Filter
Text Approach 3

- ADC 200 kHz, subsampling or bandpass sampling
- Complex Polyphase filter-decimate by 5 (complex modulated)
  - Alias 50 kHz to 10 kHz
- Complex modulate to baseband
- Half-Band filter-decimate by 2
Figure 13.12 Signal Processing Structure of Down Converter Using Aliasing to Quarter Sample Rate by Subsampling and Again by 5-Path Polyphase Filter
13.3 Digital Demodulation: Timing Recovery

- There is a random time-of-flight and unmatched crystals; therefore, timing recovery is necessary.

**Figure 13.13** Signal Flow in Modulator and Demodulator for Communicating Through Band-limited AWGN Channel.
Timing Recovery First Generation

- Replicate baud processing for early, prompt and late
Polyphase Timing Recovery

Figure 13.17 Signal Flow for Timing Recovery with Polyphase Interpolator Processing and Shifting Asynchronous Samples to Desired Time Locations

Figure 13.18 Signal Flow for Timing Recovery with Polyphase Matched Filter. Timing Recovery Selects Matched Filter Path Aligned with Input Sample Positions
Polyphase Address Pointer

Figure 13.22 Polyphase Address Pointer for Timing Recovery Loop With 40-Path Filter and Two Samples Per Symbol. In Upper Figure Pointer Does Not Cross Address Pointer Boundary, In Lower Figure Pointer Crosses Address Pointer Boundary and Converts Address 0 to Address 40 and Switches from Even-indexed to Odd-indexed Data Sample.
13.4 Modem Carrier Recovery

**Figure 13.23** Block Diagram of Quadrature Up Converter at Transmitter and Quadrature Down Converter at Receiver.
Band-Edge Considerations

Figure 13.25 Spectra at Input and Output of Band-edge Filter for Centered Input Signal and for Input Signal with Spectral Offset
Band-Edge Filter and Freq. Detector

Figure 13.26 Band-edge Filter and Frequency Error Detector Appended to Carrier Recovery Process
Band-Edge Filter

Figure 13.28 Signal Flow for Matched Filter and Upper and Lower Band Edge Filters

Figure 13.29: (a) Desired and Undesired Spectra at Input to Matched Filter and Complementary Matched Filter, (b) Response of Complementary Matched Filter to Both Inputs, and (c) Response of Polyphase Filter to Both Inputs
13.8 FM Receiver and Demodulator

Multirate signal processing has had a significant influence at the physical or hardware layer of modern communication systems. In particular, multirate signal processing is found at the core of communication systems that couple the Software Defined Radio (SDR) and Software Communications Architecture (SCA) to reconfigure system resources for operation over a wide range of modulation formats and waveforms.

In this section we demonstrate one particularly efficient multirate signal processing solution to the task of implement a radio configured to select and extract a single FM channel from the commercial FM band, to down convert and demodulate that channel, and to perform stereo demodulation and separation of the resulting baseband signal.
FM Signal Structure

Figure 13.74 Input and Output Spectra of Stereo FM Receiver and Block Diagram of Conventional FM Receiver and Stereo Demodulator
RF Downconversion

Figure 13.75 Block Diagram of DSP-Based FM Channelizer

Figure 13.76 Spectral Characteristics of Prototype Low Pass Filter in Polyphase Receiver
FM Receiver

**Figure 13.79** Signal Flow Diagram of Modified Polyphase Filter to Permit Frequencies Offset by Quarter of Output Sample Rate to Alias to Baseband
FM Demodulator

\[
\dot{\theta}(n) = \frac{d}{dn} \arctan\left( \frac{I(n)}{Q(n)} \right) = \frac{I(n)\dot{Q}(n) - \dot{I}(n)Q(n)}{I^2(n) + Q^2(n)}
\] (13.18)

- Taking the derivative of the phase of the input, digitally.

**Figure 13.80** Signal Flow Diagram of Digital FM Discriminator
Project

- Due in two weeks!
- Presentation start 1 week from Thursday
- Are there any problems, see me ASAP.