

(Digital) Signal Processing

Chris Phillips | 25/9/2023



Digital Signal Processing - DSP

- Why digital:
 - Stability (time and environment)
 - Superior performance (e.g. filter shape)
 - Cost can mass produce
- However:
 - Loss of dynamic range and SNR
 - Artifacts aliasing & birdies
 - Cross talk and reliability
 - High power requirements







Nyquist/Shannon Sampling

- A bandlimited signal can be perfectly reproduced if it is sampled at a frequency at least twice rate of maximum bandwidth
 - Critical sampling rate Nyquist
 - 100 MHz bandwidth signal needs to be sampled at 200 MHz
- *Aliasing* occurs if sampling at a lower rate
 - Frequencies can be lost, folded and mixed

Nyquist, H. (1928). "Certain topics in telegraph transmission theory." Shannon, C. E. (1949). "Communication in the presence of noise."



By Pluke - Own work, CCO, https://commons.wikimedia.org/w/index.php?curid=18423591









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Aliasing and Nyquist zones

- Aliasing can be useful in combination with filtering
 - It acts to reflect everything above the critical frequency
 - This is effectively a form of down-conversion
 - Needs bandpass filter before digitizer to reduce range of frequencies
 - Digitisers often operate in these higher-order Nyquist zones
 - Virtually all ATNF digitizer systems filter in Zone 2 or 3





Quantisation

- DSP represents samples with limited number of bits
- 1 or 2 traditional
- 8 or 16 common now
- More bits give (slightly) more "efficiency" and more dynamic range (e.g. robustness against RFI)
 - 4 bits give 98.8% efficiency





Complex Voltages

- There are multiple ways of representing a signal
 - One representation is not more "right" than another



- Complex numbers are a convenient way to represent e/m waves or voltages
 - Nyquist rate is halved data rate is unchanged
- Analogue to digital converters can produce real or complex output
- Digital filterbanks more naturally output complex values
- Complex numbers have a number of advantages
 - Slightly more efficient FFT implementation
 - "Natural" form for phase corrections



The Fourier transform

- The Fourier transform switches between the time and frequency domains
 - Or the image and spatial frequency (UV) domains, depending on context
- Often convenient to switch between representations to simplify the maths or interpretation
- FFT (Fast Fourier Transform is an algorithm to compute fourier transform efficiently

$$f(x) = \int_{-\infty}^{\infty} F(k)e^{2\pi i k x} dk$$
$$e^{ix} = \cos x + i \sin x$$
 Euler's formula



Standard Transforms



Fourier Theorems

Fourier shift theorem

 Applying a phase gradient in Fourier space shifts a function

Convolution theorem

 Convolution of two functions is equivalent to multiplication of their Fourier transforms



Spectral Ringing (Gibbs Phenomenon)

- Fourier theory assumes an infinite length function
- Realistic transforms (FFTs) limited size
- Equivalent of multiplying with a box car
- FT of boxcar sinc function
- "Sharp" features will convolved with sinc...





Polyphase Filterbanks (PFB)

- FFTs have poor spectral response
 - "Leakage" of signal from one channel to another
- Tapering of the signal in the time domain of smoothing (convolving) in frequency domain can reduce ringing
- PFB applies "pre-filter" to *multiple* FFT windows then averages data before performing FFT
- Less spectral leakage and "squarer" filter shape
- Similar results can be achieved with larger FFTs and averaging output channels in frequency



PFB Implementation





FFT Compared to PFB





Image Credit: John Tuthill

Technologies



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Image Credit: John Tuthill





Complex Visibilities

$$V_{\upsilon}(\mathbf{b}) = R_C - iR_S = \iint I_{\upsilon}(s) e^{-2\pi i \nu \mathbf{b} \cdot \mathbf{s}/c} d\Omega$$

$$R_{C} = \iint I_{E}(\mathbf{s})\cos(2\pi v \mathbf{b} \cdot \mathbf{s}/c)d\Omega$$

$$R_{S} = \iint I_{O}(\mathbf{s}) \sin\left(2\pi v \,\mathbf{b} \cdot \mathbf{s} \,/c\right) d\Omega$$

$$V_f(\mathbf{r}_1,\mathbf{r}_2) = E_f(\mathbf{r}_1) \times E_f^*(\mathbf{r}_2)$$



FFT Correlator - FX





PFB Correlator - FX





Lag Correlator - XF $\mathcal{F}{f}.\mathcal{F}{g} = \mathcal{F}{f^*g}$





Practical Implementation of Correlator

- Receive time tagged digitized voltages
- Select appropriate sample, allowing for geometric delay
 - Delay often represented as Nth order polynomial
- Apply time based phased correction
 - "Fringe Rotation" equivalent to Doppler correction
- FFT or PFB
- Fine delay correction
 - Linear phase ramp in frequency
- Cross multiple all antenna and polarisation pairs
- Time average



BIGCAT

- New backend for ATCA
- Doubles total bandwidth to 8 GHz
- Hybrid FPGA/GPU design
 - Very flexible configurations
- FPGA ADCs and coarse filterbank
 - Streamed over commodity 100 Gbps Ethernet
- To be installed first half 2024
- 1.75 Tbps total data rate from telescopes!!!
 - 117 Gbps per GPU



BIGCAT Design





Thank you

Space and Astronomy

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