Spectrometers and Signal Processing Basics 2021 GBO/AO Single Dish Workshop Ryan Lynch (GBO)







What is a Spectrometer and How Does it Work?

- A spectrometer measures intensity of electromagnetic radiation at different frequencies / wavelengths
 - In practical applications spectrometers have a finite frequency / wavelength *resolution* and a finite range of frequencies / wavelengths over which they operate
- Most astronomers are introduced to spectrometers at optical wavelengths
 - Use gratings (or prisms) to disperse light, i.e. physically separate different wavelengths
 - Measure intensity of dispersed light using a CCD
 - Position on CCD maps to wavelength
- This is not how radio-frequency spectrometers work
- Why not?

Spectrometers in Other Contexts

- Physical properties of the spectrometer often scale with wavelength (or it's inverse)
- Radio-wavelengths span ~10 m 1 mm
- Consider diffraction angle

$$\theta_m = \sin^{-1} \left(\frac{m \lambda}{d} \right), m = 1, 2, \dots$$



 A radio-frequency diffraction grating would have to be impractically large

Spectrometers in Other Contexts

- Physical properties of the spectrometer often scale with wavelength (or it's inverse)
- Or consider a refractive prism





- Index of refraction usually approaches 1 as wavelength increases
- Fractional change in wavelength $\Delta\lambda/\lambda$ is small
- Angle of refraction will be very small and not much separation of different wavelengths, leading to almost no dispersion

Spectrometers in Other Contexts

- Properties of a radio-CCD are also impractical
- Pixel size has to be $> \lambda$
- Band-gap energy of semiconductor needs to be < radiophoton energy
 - At radio frequencies photons have energies $\sim \mu eV$ meV
 - Most semiconductors have band-gap energies $\sim 1 \text{ eV}$
 - Thermal noise will swamp signal unless detectors are cooled to extremely low temperatures

Analog Radio Spectrometer

- Quantum devices like CCDs don't operate well at radio wavelengths, but analog electronic circuits do
- It is fairly easy to create electronic filters that attenuate power above/below/within certain radio frequencies



- We can envision sending copies of a signal through multiple bandpass filters, each with different frequency cutoffs, and then detecting power that passes through each filter
- A bank of filters \rightarrow a *filterbank*

Analog Radio Spectrometer

- Each signal path is known as a *channel*
- Each channel has some some narrow *channel bandwidth* over which it is sensitive
 - Analogous to resolution of an optical spectrometer
- In practical applications, channels are adjacent, with small gaps between them
- Difference between lowest and highest frequency channels defines *total bandwidth*

Analog Radio Spectrometer

- This setup is intuitively simple and illustrates basic concepts, but is not very practical
- What if you need a narrower channel bandwidth?
 - Need to duplicate the entire filterbank with a different set of filters
- What if you need a large total bandwidth and narrow channel bandwidth?
 - Need lots and lots of filters
- Can we do better?
- Yes! We can create a spectrometer using math...
- But first, what are we actually measuring?

Radio Telescopes Sample Electric Fields

- EM radiation is a time-varying electromagnetic field
- Radiation incident on a radio receiver causes a change in electric potential, i.e. a change in *voltage*



 Real-world signals have non-zero amplitude at many frequencies (i.e. polychromatic)

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Radio Telescopes Sample Electric Fields

- EM radiation is a time-varying electromagnetic field
- Radiation incident on a radio receiver causes a change in electric potential, i.e. a change in *voltage*
- For a *monochromatic* wave
 Initial Phase

$$\vec{V}(x,t) = V_0 \sin(2\pi f t + \phi_0) \qquad c = \lambda \cdot f$$
Amplitude
Frequency

 Real-world signals have non-zero amplitude at many frequencies (i.e. polychromatic)

Complex Voltage and Power

- Because incoming radiation is described by both an electric field amplitude and phase, it is convenient to represent it as a **phasor**
- This lends itself to using complex numbers to describe the voltage (recall Euler's formula)

$$Ae^{i\theta x} = A[\cos(\theta x) + i\sin(\theta x)]$$



Complex Voltage and Power

- The cosine and sine terms are often referred to as *I*(*t*) and *Q*(*t*) (i.e. *I/Q* values)
 - I corresponds to the real part of the complex voltage, and
 Q to the imaginary part
 - Don't confuse these with the I and Q of Stokes parameters!

Complex Voltage and Power

- This allows us to represent the signal with a real and imaginary part
 - Retains full amplitude and phase information so can be used for **coherent** processing
- In the final analysis we are usually interested in the **power** (which has non-zero mean), rather than the amplitude

$$P = |Ae^{i\theta x}|^2$$

- This step is usually referred to as detection
 - If we sample two polarization states, we can form Stokes parameters or other polarization products prior to detection
 - Note that we lose phase information at this stage!

More Realistic Signals



Fourier Transform Spectrometer

- We measure V(t) but want to measure power vs frequency
- We can harness the Fourier Transform to do this mathematically. Recall that for a continuous signal

$$X(f) = \int_{-\infty}^{+\infty} x(t) e^{-2\pi i f t} dt$$





Fourier Transform Spectrometer

- We measure V(t) but want to measure power vs frequency
- We can harness the Fourier Transform to do this mathematically. For a discrete signal

$$X_k = \sum_{0}^{N-1} x_n e^{-2\pi i k n/N}$$





Fourier Transform Spectrometer

- FT spectrometers are highly flexible and can be implemented on computers *if* we can fully **digitally** sample the voltage time series
- To do this we have to measure amplitude *and* phase
- We also have to sample quickly enough to detect rapidly varying (i.e. high-frequency) signals

Continuous vs Discrete Signals

- Incoming radiation is a continuous change in electric field over a continuous range of frequencies
- Digital systems work on discrete values that can be represented with some finite number of bits
- 1 bit = 2^1 values (0,1)
- 2 bit = 2² values (0 3)
- 8 bit = 2^8 = 256 values (0 255)

- The number of bits used to sample the signal defines the dynamic range
 - Smaller bit depth / resolution provides less granularity (1 bit = high or low)
 - Higher bit depth captures both weak and strong inputs
- This introduces some error, as perfect reconstruction is not possible with a finite number of bits
 - Bit depth chosen to keep quantization errors at or below an acceptable level





Analog to Digital Converters

- An **analog-to-digital converter** (ADC) is a device for converting continuous signal to discrete, digital signal
- ADCs are characterized (in part) by the number of bits they use and the maximum *sampling rate*
- The sampling rate determines the *bandwidth* of the sampler

Nyquist-Shannon Sampling Theorem

- To perfectly reconstruct a time varying signal, we must sample at a critical rate, f_N, that is **twice the highest frequency** contained in the signal
 - A signal at a frequency $f > f_N$ will be **aliased** into our sampling band at a lower apparent frequency
- *f_N* is known as the **Nyquist frequency**



Nyquist-Shannon Sampling Theorem

- This is not just a time/frequency phenomenon
- Spatial variations can be decomposed into spatial frequencies
 - Sharp features contain higher frequency components
- Nyquist sampling in spatial domain is important in mapping





Sampling Rate and Bandwidth

- To avoid aliasing, we must apply an analog filter to suppress power outside some desired bandwidth *B*
- We then use our ADC to sample at a frequency $f_s = 2 \times B$
 - Example: We want to sample 800 MHz bandwidth
 - Downcovert to **baseband** and apply low-pass filter
 - Sample at 1.6 Gsps
- Remember: filters are not perfectly sharp
 - Filter roll-off needs to start below f_s/2 to ensure aliasing is kept below an acceptable level



Why Don't Optical Spectrometers Work This Way?

- If a FT spectrometer is so useful, why not use them at shorter wavelengths / higher frequencies?
- Radio frequencies have to advantages for this approach
 - Signals can be *down-converted* from high to low frequency via *mixing*
 - Down-converted signals have frequencies that are low enough to sample with modern electronics
- Optical-frequency mixing is much harder, and electronics can't sample at the native optical frequencies

Dynamic Range for Wideband Systems

- Note that ADCs are **total power** devices
 - We have not yet sampled the power contributed at individual frequencies
- As the bandwidth goes up, so to do does the total power contributed by noise, RFI, and signal of interest
- Resolution / bit depth becomes increasingly important for wideband systems
 - Strong signals can push ADCs into non-linear regime

Types of Spectrometers

- There are different ways of implementing these general principles
- We will talk more about two examples
 - Auto-correlation spectrometer (ACS)
 - Polyphase filterbank (PFB)

Weiner-Kinchin Theorem

Relates the power spectrum to the **autocorrelation** of the incoming time series

$$S(f) = \int_{-\infty}^\infty r_{xx}(au) e^{-2\pi i f au} \, d au.$$

• r_{xx} is the autocorrelation, defined as

$$r_{xx} = \int_{-\infty}^{\infty} f(u) f^*(u - \tau) du$$

- τ is known as the **lag**, and * denotes the complex conjugate
- In words, the power spectrum is the Fourier transform of the integral of the input signal multiplied point-wise by a time-delayed version of itself

Autocorrelation Spectrometer

- An **autocorrelation spectrometer** is highly flexible in terms of total bandwidth and channel bandwidth
 - The sampling interval $\Delta \tau$ and total number of lags N completely determine these parameters

$$B = \frac{1}{2\Delta\tau}; \Delta f = 1.2\frac{B}{N}$$

The factor of 1.2 comes from the **windowing function**, which is simply a hard cutoff at $t > \Delta \tau N$ (i.e. w(t) = 1 for t <= $\Delta \tau N$, else 0)

- The observed power spectrum is a convolution of the true spectrum with the Fourier transform of *w*

 $\widetilde{S}(f) = S(f) \circ W(f)$

Autocorrelation Spectrometer

- Because the Fourier transform of a top-hat is a sinc function, the channel shape of an ACS is itself a sinc, defined by it's FWHM
 - This is where the factor of 1.2 comes from
- While an ACS is flexible and easy to implement, this frequency response is undesirable
 - Power can leak into adjacent channels
 - For very strong signals, leakage can impact significant part of band
- Can we do better?

- Yes!



Polyphase Filterbank

- In a direct discrete Fourier transform (DFT) we start with a rectangular windowing function (in time) and end with a sinc response (in frequency)
- We prefer to have a rectangular (i.e. flat) response in frequency across a channel
 - Use the Fourier inverse as the time-domain window, i.e. a sinc filter
- In practice, to obtain an N-point spectrum, use M = N x P points
 - *P* is the number of phases in the **polyphase** filterbank, also referred to as the number of taps

Polyphase Filterbank



- After multiplication by an *M*-point filter, each phase is added to produce an *N*-point input to the DFT
- The DFT can now be taken, the result squared, and then accumulated to produce a power spectrum

Polyphase Filterbank



Image credit: Jayanth Chennamangalam

- Caveats
 - In pratice, the sinc window must be truncated so the frequency response is not perfectly flat
 - We typically multiply the sinc window by an finite impulse response (FIR) filter to improve frequency response
 - Using more taps also improves response
- PFB is more computationally intensive (~1.5x) than direct DFT but improved spectral response is usually worth the trade-off

Astronomical Spectrometers

- Note that the frequency resolution we obtain is determined by the number of points in the FFT
 - The sampling theorem is also relevant here: we need 2N time samples for N frequency channels
- This creates an inverse relationship between time and frequency resolution
- In typical spectral line observing, we are more concerned with frequency resolution than time resolution
- In pulsar observing we are usually more concerned with time resolution that frequency resolution

Astronomical Spectrometers

- The last* step is typically to detect and accumulate power spectra for some integration time
 - The choice of integration time depends on the stability of the instrument and scientific goals
 - Typically use ~0.1 10 s for spectral line observing to allow efficient excising of RFI
 - Typically use 10s μs in pulsar observing to retain sensitivity to fast pulsars

*Additional signal processing often performed in pulsar observing (e.g. dedispersion, folding)

Polarization Products

- Most receivers sample two polarization states (typically linear [X/Y] or circular [L/R])
- Everything described above must be duplicated for each polarization channel
 - 2x ADCs, 2x spectrometer engines
- The polarization products that one records depends on science goals
 - Typically sufficient to record each channel's self-products independently (e.g. $|X|^2$ and $|Y|^2$)
- For strongly polarized sources, typically record Stokes parameters or self and cross terms

Polarization Products

- Stokes parameters allow complete recovery of polarized signal
 - For a linear basis:

$$I = |X|^{2} + |Y|^{2} \text{ (total intensity)}$$

$$Q = |X|^{2} - |Y|^{2}$$

$$U = 2 \text{ Re}(X*Y)$$

$$V = 2 \text{ Im}(X*Y)$$

- |V| = circular polarization $|L| = \sqrt{(Q^2 + U^2)} = linear polarization$
- We may also record the self and cross terms directly, [i.e. |A|², |B|², Re(A* B), Im(A* B)]

A Note on Complex Voltages

- There are some applications in which it is desirable/necessary to record pre-detection complex voltages
 - Very long baseline interferometry requires phase information for correlation
 - Offline analysis may be needed to form spectra with different resolutions for different applications
- This comes at the expense of very high data rates, requiring lots of storage

Hardware for Modern Digital Backends

- Modern systems are typically implemented with a combination of field programmable gate arrays (FPGAs) and GPU-equipped high performance computers running specialized digital signal processing software
- GBT currently uses five primary backends
 - Digital continuum receiver
 - Mark VI VLBI baseband recorder
 - VEGAS (spectral line/pulsar observing)
 - JPL Radar Backend
 - Breakthrough Listen (baseband recording for SETI, etc.)

Hardware for Digital Backends

- VEGAS, BTL developed through CASPER (Collaboration for Astronomical Signal Processing and Electronics Research)
- VEGAS uses 8x ROACH2 boards and NVIDIA GPUs
 - Integrated ADCs, FPGAs, 10 gigabit ethernet, serial communication ports, onboard flash memory perform initial conditioning, supply channelized data or I/Q values
 - Additional spectral line / pulsar processing performed on GPUs/CPUs
 - Data stored on beeg-fs distributed filesystem
 - 8 independent spectrometer banks for maximum frequency coverage/flexibility

Hardware for Digital Backends







Questions?



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