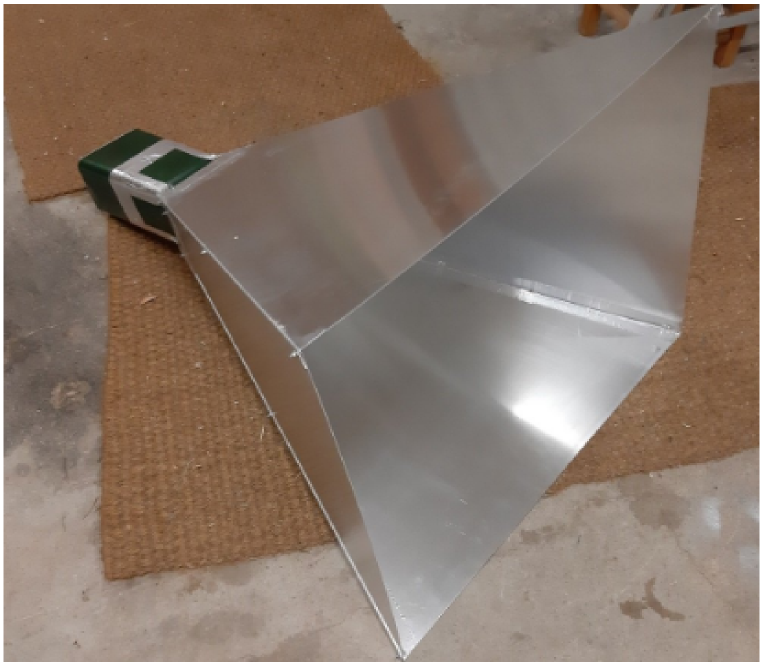
Cosmocode : Week 1

Introduction of Horn antenna



A horn antenna is a type of microwave antenna widely used in various applications, including radar systems, satellite communication, wireless communication, and radio astronomy. It is named "horn" due to its characteristic shape, which resembles an open-ended horn.

The horn antenna's design is based on waveguide technology, where the waveguide's dimensions determine the operating frequency range and radiation pattern. It is typically made of metal and can be either rectangular or conical in shape.

The working principle of a horn antenna is based on its ability to convert guided electromagnetic waves (usually in a waveguide) into free-space radiation. When electromagnetic waves travel through the waveguide and reach the open end (the mouth) of the horn antenna, they are radiated out into space. The shape and size of the horn antenna determine its directivity, gain, and beamwidth, which are crucial parameters in determining its performance.

Horn antennas are known for their wide bandwidth and low loss, making them suitable for high-frequency applications. They can be designed for various polarization types, such as linear, circular, or dual polarization, based on the application requirements.

Some key advantages of horn antennas include:

High directivity: Horn antennas can achieve high directivity, allowing them to focus energy in a particular direction.

Broad bandwidth: They can cover a wide range of frequencies, making them versatile for different applications.

Low loss: Horn antennas typically exhibit low insertion loss, which helps in efficient energy transfer.

Simple construction: Their straightforward design and construction make them cost-effective and relatively easy to manufacture.

Python in signal processing

In signal processing with Python, there are several libraries and tools that are commonly used to work with signals, process data, and analyze results. Here are some basics of Python libraries often used in signal processing tasks:

NumPy: NumPy is a fundamental library for numerical computing in Python. It provides support for large, multi-dimensional arrays and matrices, along with a vast collection of high-level mathematical functions to operate on these arrays efficiently. In signal processing, NumPy is used for data manipulation and basic operations on signals.

SciPy: SciPy is built on top of NumPy and offers additional functionality for scientific and technical computing. It includes modules for signal processing (e.g., filtering, spectral analysis), optimization, integration, interpolation, and more

Matplotlib: Matplotlib is a popular library for creating static, interactive, and animated plots in Python. It is often used to visualize signals, time-domain plots, frequency-domain plots, and various data representations in signal processing tasks.

Pandas: Pandas is widely used for data manipulation and analysis. It provides data structures like DataFrames and Series that are well-suited for handling and processing time-series data, which is common in signal processing applications.

Week 2

Signal Processing

Signal processing is a branch of electrical engineering and applied mathematics that deals with the analysis, manipulation, and interpretation of signals. In the context of signal processing, a signal is a representation of information in the form of a function that varies with time, space, or some other independent variable.

The main objectives of signal processing are to extract useful information from signals, improve signal quality, and make data more understandable for further analysis or transmission. Signal processing techniques are used in a wide range of fields, including telecommunications, audio and speech processing, image and video processing, radar and sonar systems, medical imaging, and more.

There are two primary domains in signal processing:

Time-domain signal processing: It involves analyzing signals in the time domain, where the variations occur along the time axis. Techniques such as filtering, convolution, and windowing are commonly used in time-domain signal processing.

Frequency-domain signal processing: It involves analyzing signals in the frequency domain, where the variations occur along the frequency axis. The most common tool used for this purpose is the Fourier Transform, which converts a signal from the time domain to the frequency domain. The Fast Fourier Transform (FFT) is a computationally efficient algorithm for calculating the Fourier Transform.

Some key concepts and techniques used in signal processing include:

Filtering: Filtering is the process of modifying or extracting specific frequency components from a signal. It is used for noise reduction, signal enhancement, and extracting relevant information.

Sampling and Reconstruction: Signals are often continuous in nature, but in practice, they need to be converted into a discrete form for processing and transmission. Sampling involves converting continuous signals into discrete samples, and reconstruction is the process of converting discrete samples back into a continuous signal.

Modulation and Demodulation: Modulation is the process of embedding information in a carrier signal, while demodulation is the process of extracting the original information from the modulated signal. Modulation is commonly used in communication systems to transmit data over long distances.

Wavelet Transform: The Wavelet Transform is used for both time and frequency analysis, making it suitable for non-stationary and transient signal analysis.

Fourier Transform

The Fourier Transform is a fundamental mathematical tool used in signal processing, mathematics, physics, and various other fields. It allows us to analyze a signal or a function in terms of its frequency components. The transform is named after the French mathematician and physicist, Joseph Fourier, who introduced the concept in the early 19th century.

The Fourier Transform takes a function in the time domain (or spatial domain) and transforms it into the frequency domain. In the frequency domain, the signal is represented as a sum of sinusoidal components of different frequencies. This provides a different perspective on the signal, revealing its frequency content and allowing us to analyze its periodic and non-periodic characteristics.

The continuous Fourier Transform (CFT) is defined for continuous-time signals, and its formula is given by:

$$F(\omega) = \int_{-\infty}^{\infty} f(t)e^{-j\omega t}dt$$

where:

- $F(\omega)$ is the complex-valued function in the frequency domain.
- $f(t)$ is the function in the time domain.
- ω is the angular frequency (frequency in radians per unit time).
- j is the imaginary unit (equal to $\sqrt{-1}$).

For discrete-time signals (or sampled signals), we use the Discrete Fourier Transform (DFT) or its fast

algorithm called the Fast Fourier Transform (FFT). The DFT is given by:

$$X[k] = \sum_{n=0}^{N-1} x[n]e^{-j(2\pi/N)kn}$$

where:

- $X[k]$ is the complex-valued result in the frequency domain.
- $x[n]$ is the discrete-time signal in the time domain.
- N is the number of samples in the signal.
- k is an integer representing the frequency bin.

The main applications of the Fourier Transform include:

Frequency Analysis: It is used to analyze the frequency content of a signal, which is crucial in understanding the behavior and characteristics of a signal.

Filtering: Fourier Transform helps design filters to remove unwanted frequency components from a signal or to extract specific frequency components.

Signal Compression: In some cases, the Fourier Transform can be used for signal compression by representing a signal with fewer significant frequency components.

Image and Audio Processing: Fourier Transform is widely used in image and audio processing to analyze and manipulate images and audio signals.

Communication Systems: In communication systems, Fourier Transform is used for modulation, demodulation, and channel equalization

Usefulness of Fourier Transform in Signal Processing

- The Fourier transform is extensively used in signal processing for analyzing and manipulating signals.
- By decomposing a signal into its constituent frequencies, it enables the identification of specific frequency components, filtering out noise, and extracting useful information.
- It plays a crucial role in applications such as audio and image compression , equalization , filtering , and modulation / demodulation techniques.

Week 3

- **VISIT TO SSA LAB (21st june 2023)**
    1. VISIT TO RADIO AND ASTRONOMY LAB . INTERACTION WITH SENIORS .
    2. Got familiar with using horn antenna along with rtl-sdr and low noise amplifier.
    3. Installation of GQRX software because it gives averaged out results of many signals processed .
    4. Code used for signal processing: https://drive.google.com/file/d/1r0-kszAJmpt54WF4y_h7jAS0rQvQr_EK/view?usp=sharing

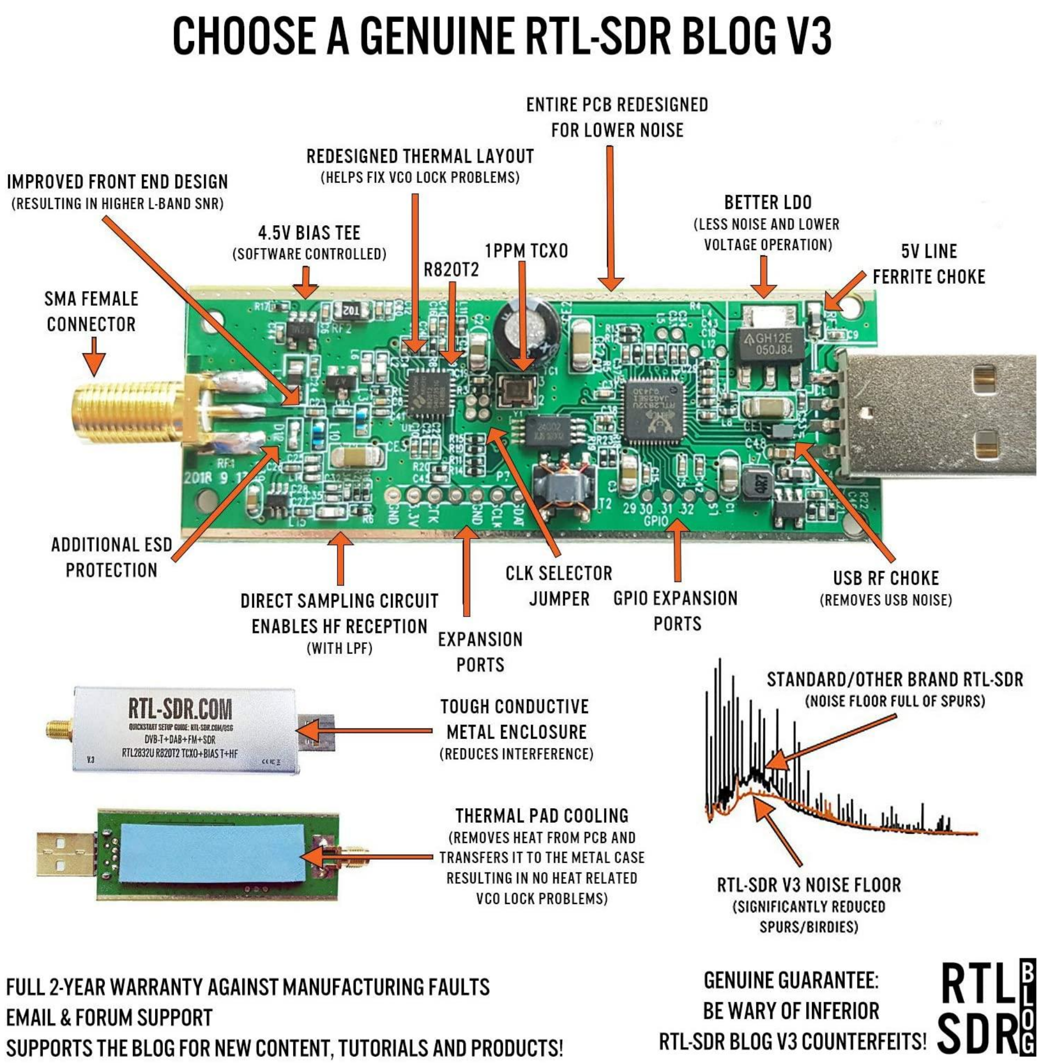
Components of rtl -sdr

RTL-SDR (Software-Defined Radio) refers to the use of a certain type of low-cost digital TV tuner based on the Realtek RTL2832U chipset as a wideband software-defined radio receiver. While the RTL-SDR itself is a complete device, there are a few key components that make up the system. Here are the main components of an RTL-SDR setup:

RTL2832U Chipset: The RTL-SDR is built around the Realtek RTL2832U chipset, which is a USB 2.0 digital TV tuner. It was originally designed for receiving digital television signals, but it was discovered that it could also be repurposed as a software-defined radio receiver.

Antenna: An external antenna is required to capture radio signals and feed them into the RTL-SDR receiver. The type of antenna used depends on the specific frequency range and application. Common choices include whip antennas, dipole antennas, discone antennas, and specialized antennas for specific frequency bands.Drivers: Depending on your operating system, you may need to install specific drivers to enable communication between the RTL-SDR device and your computer. The most commonly used driver is called librtlsdr, which provides a software interface for accessing the RTL-SDR hardware.

Additional Accessories (Optional): Depending on your specific needs, you may use additional accessories such as filters, amplifiers, and attenuators to improve the performance of your RTL-SDR setup. These accessories can help reduce interference, increase sensitivity, or adapt to specific frequency ranges.



Low noise amplifier :

A low noise amplifier (LNA) is an electronic device that amplifies weak signals while introducing minimal additional noise. It is commonly used in communication systems and radio frequency (RF) applications where signals are weak and need to be boosted before further processing or transmission.

LNAs are typically used at the front end of RF systems, such as in wireless receivers, radar systems, satellite communication systems, and sensitive measurement equipment. They are designed to operate at high frequencies and provide gain to overcome losses in subsequent stages of the system.

Key characteristics of a low noise amplifier include:

Low noise figure: The noise figure of an LNA represents the amount of noise it adds to the signal. Lower noise figures indicate better performance, as it means the amplifier is adding less noise to the original signal.

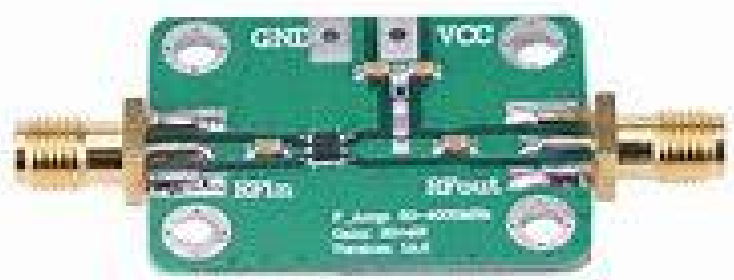
High gain: LNAs are designed to provide significant gain to amplify weak signals. Higher gain helps to increase the signal strength without introducing excessive noise.

Wide bandwidth: LNAs should have a broad operating bandwidth to accommodate a range of frequencies commonly encountered in RF systems.

High linearity: Linearity refers to an amplifier's ability to maintain a linear relationship between the input and output signals. A high linearity LNA ensures that the amplified signal faithfully represents the input signal, without distortion or non-linear behavior.

Low power consumption: In many applications, power efficiency is crucial. LNAs should be designed to consume minimal power while providing the necessary amplification.

Overall, the primary purpose of a low noise amplifier is to increase the signal level while maintaining a high signal quality, minimizing noise contributions, and improving the overall sensitivity and performance of RF systems.



Local oscillator and its function :

In signal processing, a local oscillator (LO) is an important component used in various applications, particularly in frequency conversion and modulation/demodulation processes. The primary function of a local oscillator is to generate a stable and controllable oscillating signal at a specific frequency that is used as a reference or a carrier for the signal processing operation.

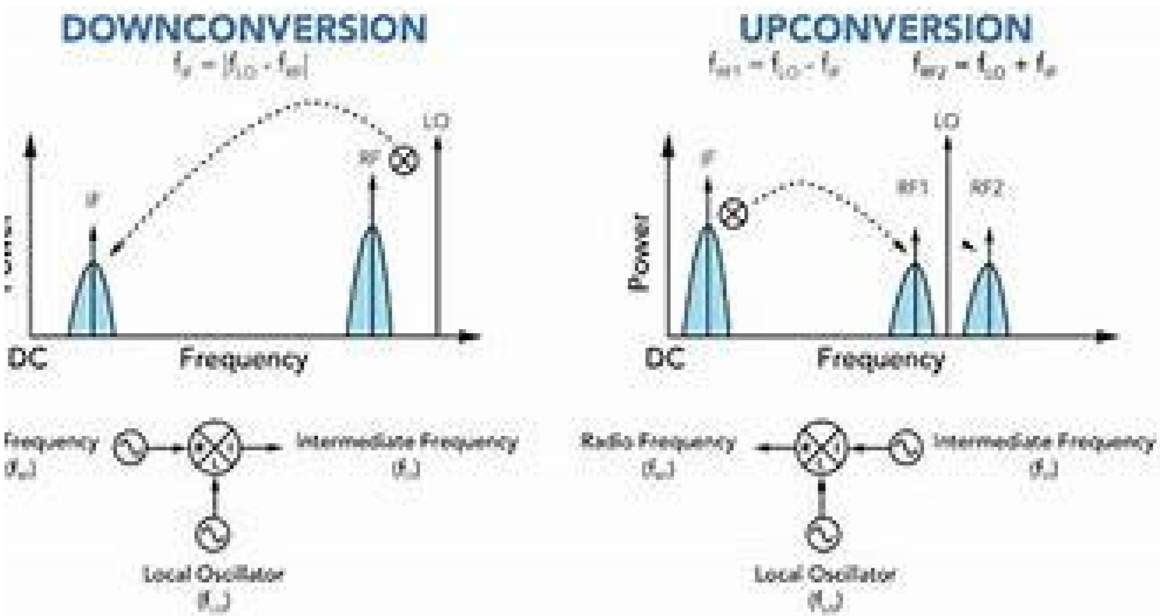
Here are a few common applications of local oscillators in signal processing:

Frequency Conversion: In frequency conversion, the local oscillator is used to shift the frequency of a signal from one frequency band to another. This process is essential in applications such as heterodyne receivers, upconversion/downconversion in wireless communication systems, and frequency translation in mixers. The local oscillator generates a frequency that, when mixed with the input signal, produces an intermediate frequency (IF) or a desired output frequency.

Modulation and Demodulation: Local oscillators play a crucial role in modulation and demodulation processes. In modulation, the local oscillator generates a high-frequency carrier signal that is modulated with the information-bearing signal to produce a modulated signal for transmission. In demodulation, the local oscillator is used to recover the original information from the modulated signal by mixing it with the received signal to extract the desired frequency components.

Oscillators in RF and Microwave Systems: Local oscillators are widely used in RF and microwave systems for signal generation at specific frequencies. They are employed in applications such as radar systems, satellite communication, wireless transmitters, and RF signal generators. The local oscillator provides a stable and precisely controlled signal that serves as a carrier or reference frequency for these systems.

Overall, local oscillators are essential components in signal processing systems, providing stable reference signals, carrier frequencies, or intermediate frequencies for various operations such as frequency conversion, modulation, demodulation, and frequency synthesis.



The Nyquist sampling theorem

The Nyquist sampling theorem, also known as the Nyquist-Shannon sampling theorem, is a fundamental concept in signal processing and digital communications. It establishes the minimum sampling rate required to accurately reconstruct a continuous-time signal from its samples.

The theorem, formulated by the American engineer Harry Nyquist and later extended by Claude Shannon, states that in order to accurately reconstruct a band-limited continuous-time signal, the sampling frequency must be at least twice the highest frequency component present in the signal. Mathematically, it can be stated as follows:

Fs ≥ 2 * Fmax

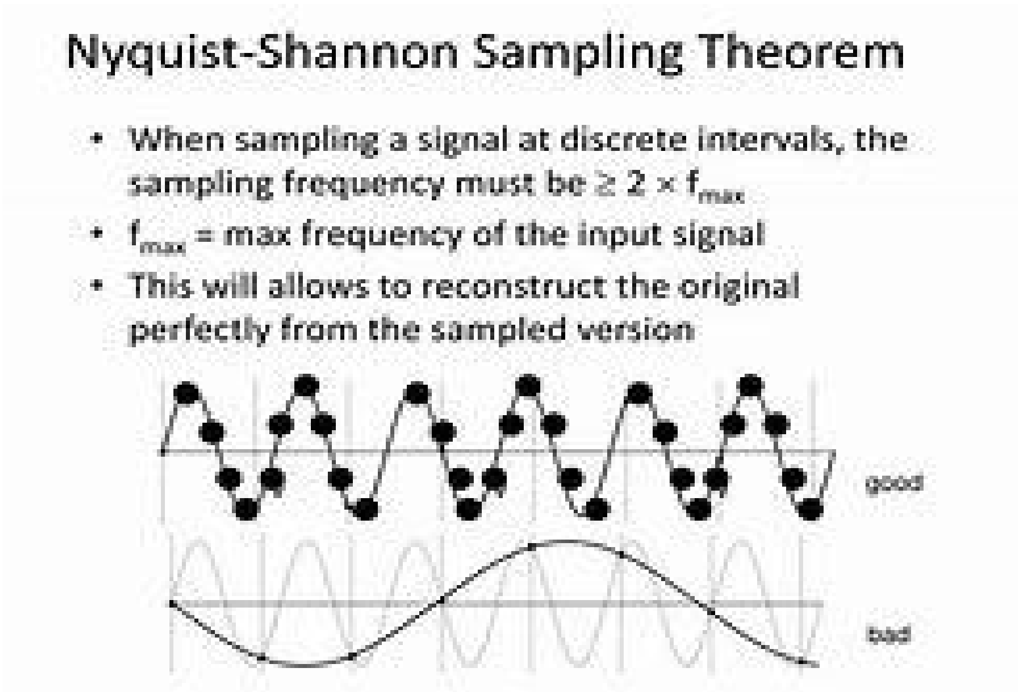
where Fs is the sampling frequency and Fmax is the maximum frequency component in the signal.

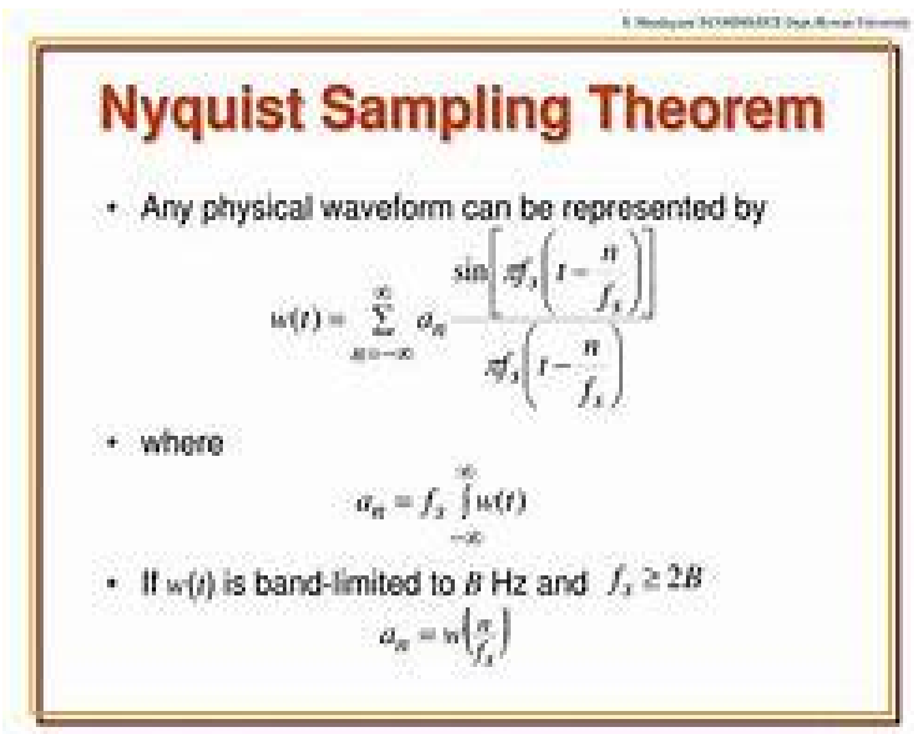
According to the Nyquist sampling theorem, the sampling rate must be greater than the Nyquist rate, which is twice the highest frequency present in the signal. This ensures that all the information contained in the original continuous-time signal is preserved and can be accurately reconstructed from its samples.

If the sampling rate is less than the Nyquist rate, a phenomenon called aliasing occurs. Aliasing causes overlapping and distortion of frequency components, leading to a loss of information and an inaccurate representation of the original signal.

In practical terms, the Nyquist sampling theorem has significant implications for digital signal processing and the design of analog-to-digital converters (ADCs) in various applications. It helps determine the required sampling rate for a given signal bandwidth and ensures that sufficient samples are taken to faithfully represent the original analog signal.

It is worth noting that while the Nyquist sampling theorem provides a minimum sampling rate, in practice, higher sampling rates are often used to provide additional margin and improve the quality of the reconstructed signal.

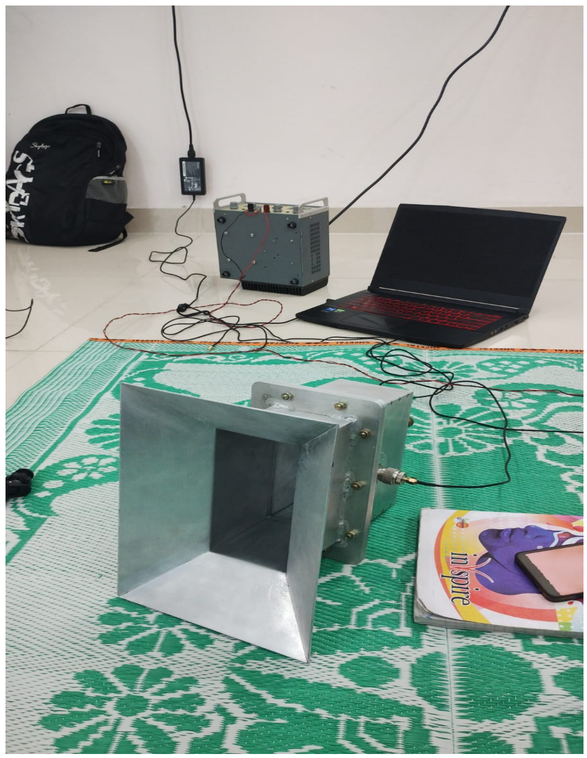

Nyquist-Shannon Sampling Theorem
• When sampling a signal at discrete intervals, the sampling frequency must be ≥ 2 × f_{max}
• f_{max} = max frequency of the input signal
• This will allows to reconstruct the original perfectly from the sampled version
good
bad


Nyquist Sampling Theorem
• Any physical waveform can be represented by
$w(t) = \sum_{n=-\infty}^{\infty} a_n \frac{\sin\left[\pi f_s\left(t - \frac{n}{f_s}\right)\right]}{\pi f_s\left(t - \frac{n}{f_s}\right)}$
• where
$a_n = f_s \int_{-\infty}^{\infty} w(t)$
• If $w(t)$ is band-limited to B Hz and $f_s \geq 2B$
$a_n = w\left(\frac{n}{f_s}\right)$

Week 4

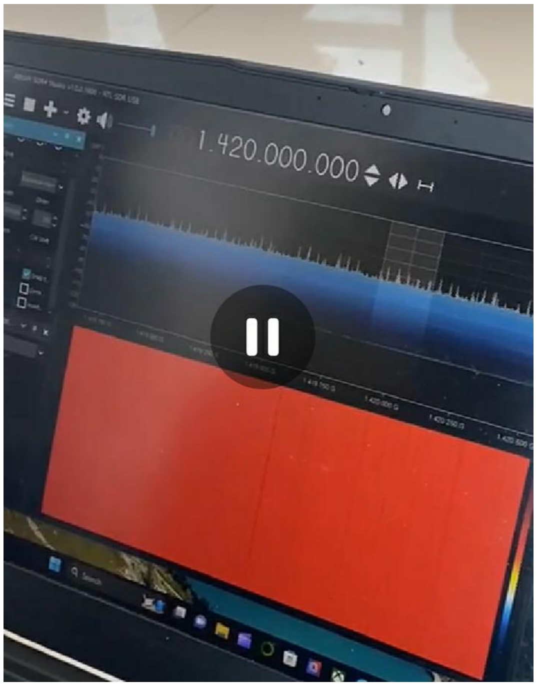
Testing of Horn antenna :

- Small horn antenna :

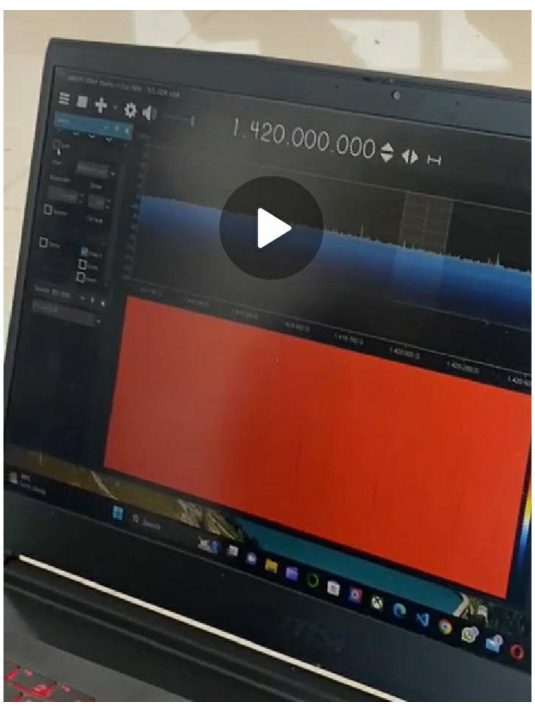


Testing of small horn antenna was done by using it to collect radiation from atmosphere and from lab walls . Marked difference in both cases were observed .

Radiation pattern from atmosphere



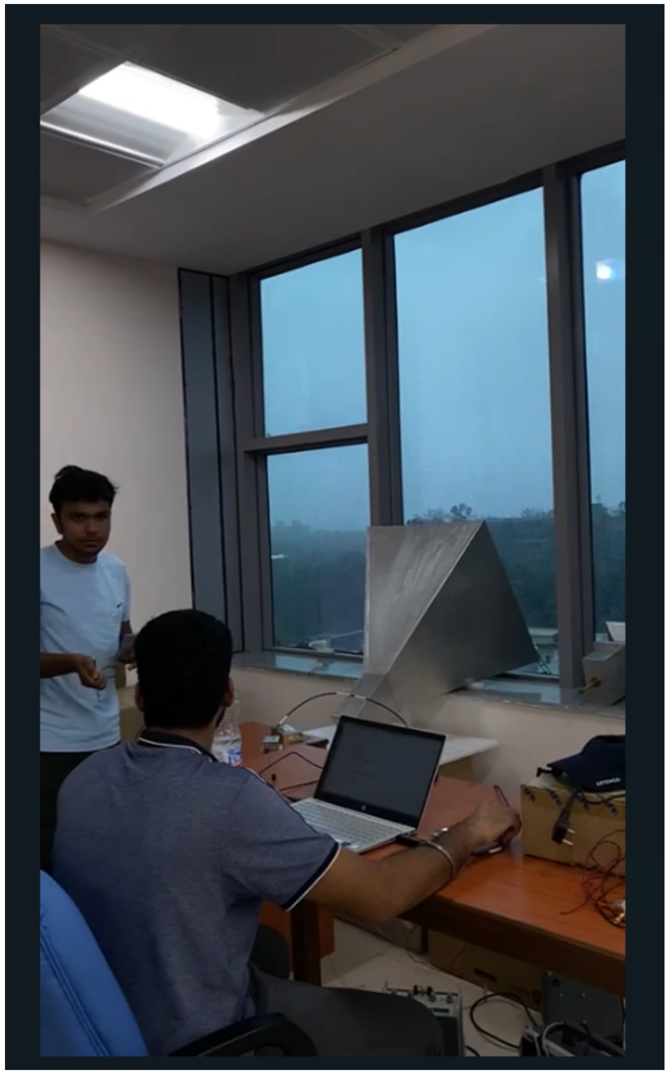
Radiation pattern from walls



- **Large Horn antenna : Large Horn antenna was used to see hydrogen spectrum from atmosphere and observe** High Galactic latitude polarized emission at 1.4 GHz and implications for cosmic microwave background observations **and the signals were converted into frequency domain by spectrum analyzer .**

HYDROGEN SPECTRUM

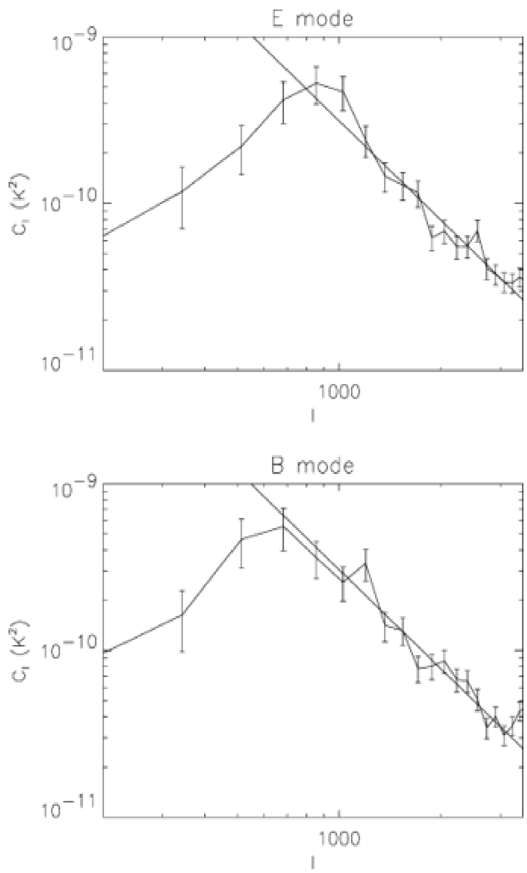
One of the most important contributions of radio astronomy to our understanding of the universe is the detection and analysis of the 21-centimeter line of atomic hydrogen.Atomic hydrogen emits and absorbs radiation at a characteristic frequency corresponding to a wavelength of approximately 21 centimeters (1420.4058 MHz). This emission and absorption line arise from the energy transition of the electron in a hydrogen atom flipping its spin due to the intrinsic magnetic properties of the electron. This phenomenon is known as the hyperfine transition.Radio telescopes are designed to observe and detect radio waves emitted by various astronomical sources, including hydrogen gas in interstellar space. The 21-centimeter line is crucial for studying the structure, dynamics, and distribution of hydrogen in galaxies, interstellar medium, and even the early universe.By observing this spectral line, astronomers can gather information about the velocity and distribution of hydrogen gas in galaxies. This data is invaluable for studying galaxy rotation curves, identifying the presence of dark matter, and understanding the processes of galaxy formation and evolution.Radio astronomers use specialized radio telescopes and radio receivers to detect the weak signals emitted by atomic hydrogen. The information collected from these observations helps build a comprehensive picture of the large-scale structure of the universe.



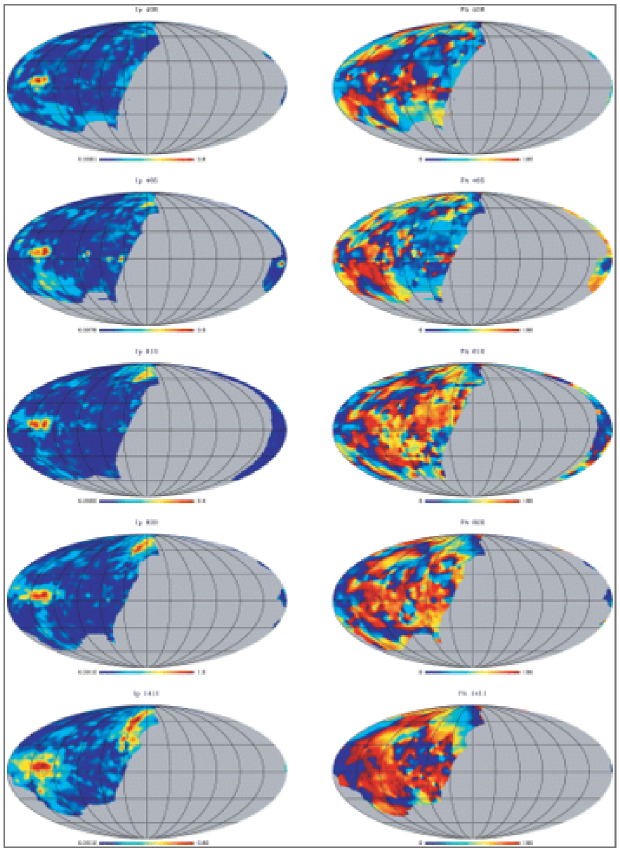
High Galactic latitude polarized emission at 1.4 GHz

High Galactic latitude polarized emission at 1.4 GHz refers to the presence of polarized radio emission coming from regions of the sky with high Galactic latitudes. In astronomy, Galactic latitude is a coordinate system used to describe the position of an object in the sky relative to the plane of the Milky Way galaxy.The polarized emission at 1.4 GHz is typically associated with synchrotron radiation. Synchrotron radiation is produced by high-energy charged particles, usually electrons, spiraling around magnetic field lines at relativistic speeds. This phenomenon is prevalent in astrophysical environments where strong magnetic fields and relativistic particles are present.At 1.4 GHz (or 1400 MHz), the emission falls within the radio frequency range. Radio telescopes equipped with polarization-sensitive receivers are used to detect and study this polarized emission. These instruments are designed to measure both the total intensity of the radio emission and the polarization properties, such as the polarization angle and the degree of polarization.High Galactic latitude regions are away from the plane of the Milky Way galaxy, where the amount of interstellar material (gas, dust, etc.) is relatively low. These regions are less affected by Galactic foreground emission, making them valuable for studying extragalactic sources and cosmological phenomena.

The study of polarized emission at 1.4 GHz from high Galactic latitude regions can provide valuable insights into various astrophysical processes, such as the magnetic field structure in the interstellar medium, the properties of cosmic-ray electrons, and the large-scale structure of the universe. It is an essential tool for astronomers studying the polarization of radio sources and cosmic microwave background radiation, among other things.



E- (top) and *B*-mode (bottom) angular power spectra of the polarized emission at 1.4 GHz in the observed patch.



Polarized intensity *Ip* (left) and polarization angle maps (right) formed by interpolating the Brouw & Spoelstra (1976) data. The maps correspond to 408, 465, 610, 820 and 1411 MHz (from top to bottom). The units are Kelvin and degrees, respectively. The maps, in Galactic coordinates centred on the Galactic centre, have been convolved with a 4° FWHM Gaussian filter.

Spectrum analyzer :

o plot signals in the frequency domain, you would typically use a spectrum analyzer. A spectrum analyzer is a specialized electronic instrument that displays the frequency spectrum of an input signal. It allows you to visualize the various frequency components of a signal and their corresponding amplitudes.

There are two main types of spectrum analyzers:

Real-time Spectrum Analyzer: This type of spectrum analyzer provides a real-time display of the frequency spectrum. It can capture and analyze signals in the frequency domain as they happen, making it suitable for analyzing time-varying or dynamic signals. Real-time spectrum analyzers are commonly used in applications like wireless communications, radar, and other real-time signal analysis scenarios.

Swept-Tuned Spectrum Analyzer: This type of spectrum analyzer works by sweeping through a range of frequencies and measuring the signal power at each frequency point. The results are then displayed as a spectrum plot. Swept-tuned spectrum analyzers are suitable for analyzing continuous waveforms and are commonly used in RF (radio frequency) and microwave applications.

Modern spectrum analyzers often come with additional features like the ability to perform various types of signal measurements, such as channel power, adjacent channel power, and occupied bandwidth measurements. They may also have built-in functionalities for demodulation and analysis of specific modulation schemes in communication systems.Additionally, there are software-based spectrum analyzers that use the processing power of a computer to perform the spectrum analysis. These software analyzers often work in conjunction with hardware like RF receivers or sound cards to capture and process signals.

Whether using a hardware-based or software-based solution, spectrum analyzers are powerful tools for understanding the frequency content of signals and are widely used in fields such as telecommunications, audio engineering, radio astronomy, and many other applications where signal analysis in the frequency domain is crucial.