DESIGN AND CONSTRUCTION OF AN EXPERIMENTAL SETUP FOR RESEARCH IN ADAPTIVE CANCELLATION OF RADIO FREQUENCY INTERFERENCE

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ABSTRACT

Radio frequency interference (RFI) is becoming a major problem for research in radio astronomy. Radio astronomy is the science of studying the universe by observing radio waves emitted by celestial objects. Radio astronomers have used RFI mitigation techniques developed for communication application to remove interference from their data. However, RFI suppression achieved in these techniques are not sufficient for radio astronomy applica-tion, since the radio signals from celestial objects are more than 1000 times weaker than that of the interfering signals. This limitation demands for developing new mitigation techniques with better RFI suppression.

One of the most successful approaches of RFI mitigation is the technique of adaptive interference cancellation. This technique requires data from two channels: (i) a primary channel to obtain the RFI affected astronomical signal and, (ii) a reference channel to obtain the RFI. In this technique the data from the reference channel is used to remove RFI in the primary channel using adaptive filters. The primary goal of the project is to design and construct an experimental setup for RFI mitigation which has the capability of acquiring data from a primary and a reference channel. We used the 1.4 GHz receiver system of the 10.4-m telescope at the Raman Research Institute as the primary channel. A reference antenna and receiver system were designed and constructed near the frequency 1420.5 MHz for obtaining the RFI signals. An interface circuit was also built to connect the telescope and reference antenna outputs to an existing Data Acquisition System. The DAS is capable of recording digital voltages from two channels, which are then used for further processing in MATLAB.

The second goal of the project is to study two aspects that affect RFI suppression in a conventional adaptive filter. They are :(i) the relative difference between bandwidth of RFI and that of the primary channel and,(ii) the number of bits used for quantization of signals in the primary and reference channels.

Both these aspects were studied using simulated data in MATLAB and astronomical data taken with the experimental setup.The main results of this study are (i)if the bandwidth of the RFI is smaller than that of the primary channel, then an improvement in RFI rejection can be achieved by restricting the bandwidth of the reference channel to the frequency range of RFI ,(ii) the RFI rejection in general increases with the number of bits used for quantizing the signals in the two channels and (iii) the RFI rejection is a sensitive function of the number of bits used for representing the reference signal.

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ϵ - error signal
ξ- mean square error

Δ - gradient

μ - step size in LMS algorithm

LIST OF ABBREVIATIONS

RFI	Radio Frequency Interference
ITU	International Telecommunications Union
FCC	Federation of Communications Commission
FIR	Finite Impulse Response
IIR	Infinite Impulse Response
MSE	Mean Square Error
LMS	Least Mean Square
LNA	Low Noise Amplifier
DA S	Data Acquisition System
ADC	Analog to Digital Converter
IF	Intermediate Frequency
LO	Local Oscillator
CPLD	Complex programmable logic device.
INR	Interference to Noise ratio.

CHAPTER 1

RADIO ASTRONOMY AND RADIO FREQUENCY INTERFERENCE

1.1 INTRODUCTION

Radio astronomy is the science that studies the universe by observing the radio waves emitted by the celestial objects. Radio astronomical observations are made using radio telescopes, which are basically very sensitive radio receivers. These radio telescopes operate at specific frequencies over the frequency range of 15 MHz to 300 GHz (wavelength - 0.001m to 30m) and measure the power of radio waves of celestial objects over some bandwidth centered at the operating frequency.

International unions such as ITU, FCC allocate and regulate the radio spectrum for a variety of scientific and commercial applications. Usually, radio astronomy observations are done from allocated frequency bands. The primary band allocation for radio astronomy observations restricts the radio frequency transmissions in it .The secondary band allows transmissions with a specified power level. Although RF spectrum is allocated for radio astronomy, the observations in the secondary band are affected by transmission of signals from other bands. The specified RFI power level in the secondary bands is not good enough for sensitive observations. Also the high sensitivity requirement of radio astronomy observations demands the use of frequencies outside the bands allocated for the radio astronomy. The signals affecting radio astronomical observations are referred to as *Radio Frequency Interference* (RFI). These interference can enter the radio telescope through the side lobes of the antenna.

In recent times, RFI poses a major problem for radio astronomers to pursue their research in radio astronomy. Therefore, active research to mitigate RFI has been initiated and many methods developed for suppressing interference in communications have been tried out. However, the suppression achieved is not good enough. Hence research to develop better methods to suppress RFI is essential.

1.2 SOURCES OF RADIO FREQUENCY INTERFERENCE

Radio frequency interference is either generated internal to the telescope or externally. Internal or local RFI is due to components associated with the radio telescope. The potential sources of RFI include networking devices, digital systems, oscillations, motors. All these devices are shielded well to reduce the effect of electromagnetic radiation from them. However, the observations are affected due to the proximity of the devices to the radio telescope.

Externally generated RFI can be due to natural source or man-made. Thermal emissions from ground, lightning, sun and other bright radio sources are a few examples of natural sources of RFI. Man-made RFI can be due to spurious, out-of band signals from commercial services such as communication networks, RADAR, domestic appliances which fall in the radio astronomy band of radio telescopes.

1.3 METHODS TO SUPPRESS RADIO FREQUENCY INTERFERENCE

Mitigation strategies for RFI depend on various factors such as type of radio telescope, type of observations and nature of RFI. Hence, there is no universal method that can be adopted for RFI mitigation. The RFI suppression techniques can be broadly classified into three categories.

1. Time domain processing: Strong and short bursts of RFI in a data can be removed by blanking. In this method, the observational data is sampled at a high frequency and thresholded to remove the data corrupted with interference. This method of miti-gating RFI is called blanking technique.

2. Frequency domain processing: Narrow band RFI with fixed frequency RFI can be removed using notch filters. Alternatively, the receiver bandwidth can be divided into many frequency channels and channels with RFI are removed from further data-processing. This method works better for cases where there is no astronomical spectral line information in the rejected channels.

3. Spatial processing: Spatial filtering methods isolate and remove the interference in spatial domain. The direction of arrival of the RFI can be determined in an array of antennas. In such arrays spatial nulls can be oriented in the direction of the RFI when forming the beam. Thus the antenna gain in the direction of the RFI is reduced considerably, thus suppressing RFI.

Since the position of the radio telescope, the frequency and position of the source of interference in general, are not stationary, adaptive techniques have been used in the implementations of all the above described processing.

1.4 ADAPTIVE INTERFERENCE CANCELLATION

The adaptive noise canceling technique in signal processing was first applied for radio astronomy application by Barnbaum & Bradley (Astronomical Journal, 1998). IN this technique, data is taken from two channels. The primary channel consists of the signal received by the radio telescope. The primary channel signal comprises both the astronomical signal and RFI. The auxiliary or the reference antenna will receive the RFI signal.

The astronomical signals in both the channels are uncorrelated due to the different propagation paths of the signals. The RFI signal in the reference channel is adjusted with an adaptive filter, such that the error signal (difference between the primary and reference channel signals) is zero. The error signal is the parameter used for adjusting the filter. The filter's coefficients are adjusted by an adaptive algorithm, which minimizes in the least squares sense the power output of the system. Results show that about 20 dB of suppression has been achieved by this scheme with preliminary tests.

1.5 OBJECTIVE OF THE PROJECT

The adaptive interference cancellation has been successful in mitigating RFI to some extent. This technique as mentioned earlier requires two channels to receive the RFI and the astronomical signal contaminated with RFI. The primary goal of this project is to design and construct an experimental setup for RFI mitigation, which can acquire data from a primary and reference channel. The second goal is to study about two aspects that affect RFI suppression in the adaptive interference cancellation system, viz., the number of bits used for quantization of signals in primary and reference channels and the bandwidth reduction of RFI over the primary channel signal with the real data.

Chapter 2 discusses about adaptive filters and their application for suppressing RFI. Chapter 3 describes the construction of the reference antenna, design and construction of amplifier and band-pass filter and the Data Acquisition System. The simulations to study the effect of quantization of bits and bandwidth restriction in the reference channel signal to suppress RFI are described in chapter 4, which also discusses the application of these effects in astronomical data.

CHAPTER 2

ADAPTIVE FILTERS

2.1 INTRODUCTION

Digital signal processing refers to the operations that can be performed on discrete signal to modify the input-output relationship of the signals. A discrete signal can be obtained from an analog signal by sampling the analog signal and quantizing it. A discrete-time sequence x(n) generated by uniform sampling of an analog signal x(t) is given by $x(n) = x(t)|t = nT = x(nT)$. The sampling frequency f_s is the reciprocal of the sampling period, T. Sampling theorem states that for a band-limited signal with maximum frequency f_{max}, the sampling frequency must be at least twice that of the maximum frequency $f_{max}(f_s \geq 2f_{max})$ in order to avoid aliasing. When the signals are under-sampled, the higher frequency components of the signals overlap with the lower frequency components causing distortion of the signals .The distortion caused is called *aliasing*. The sampled signals, which are in the form of continuous-amplitude, discrete-time signals are converted into discrete-amplitude discrete-time signals by *quantisation*.

This chapter briefly outlines some aspects of signal processing such as DFT, adaptive filter theory and their applications in RFI mitigation.

2.2 BASICS OF SIGNAL PROCESSING

2.2.1 Fourier Transforms

Fourier transform is a unique mathematical tool useful in the analysis and design of linear time-invariant systems. Discrete Fourier transforms represent a signal in the frequency domain as a sum of its complex exponential components. For a discrete sequence x(n) of N samples placed apart t_s seconds apart, over $T = Nt_s$ observation intervals, the DFT is given by

$$X(k) = \sum_{n=0}^{N-1} x(n)e^{-j2\pi kn/N} \tag{2.1}$$

and the inverse DFT is given by

$$x(n) = \frac{1}{N}\sum_{k=0}^{N-1} X(k)e^{j2\pi kn/N}, k=0\ \text{N - 1} \tag{2.2}$$

The X(k) represents the harmonic of the signal and is spaced at $1/T = 1/Nt_s = f_s/N$ from the next harmonic.

2.2.2 Z transform

Z transform is a generalization of the Fourier transform for discrete-time signals.The unilateral Z-transform of a discrete sequence x(n) is defined as

$$X(z) = \sum_{n=0}^{\infty} x(n)z^{-n} \tag{2.3}$$

where z is a complex variable. The power series representing z transform is infinite and does not converge for all values of z.

2.2.3 Power Spectral Analysis

The power spectrum of a signal is a function in the frequency domain and is given by

$$P_x = | X(k)|^2 \tag{2.4}$$

The power spectrum of a signal defines the power in each frequency component of a signal.Evaluating a power spectrum is a way to isolate noise.

2.2.4 Filters

Filters are linear time-invariant (LTI) systems used as frequency selective devices to allow certain frequencies and block the other frequencies. The LTI systems can be described by a linear constant coefficient equation as

$$y(n) = \sum_{k=0}^{M} a_k x(n-k) - \sum_{k=1}^{N} b_k y(n-k) \tag{2.5}$$

The frequency response of a filter describes the behavior of the system in frequency domain. The transfer function of the filter in *z* domain is given by

$$H(z) = \frac{y(z)}{X(z)} = \frac{\sum_{k=0}^{M} a_k z^{-k}}{\sum_{k=0}^{N} a_k z^{-k}} \tag{2.6}$$

where H(z) is the z-transform of the impulse response sequence h(n) of the filter. Depending on the impulse response ,the filters are classified as :*Finite Impulse Response (FIR) filters* and *Infinite Impulse Response (IIR) filters.*

FIR filters:

The transfer function of FIR filter is given by

$$H(z)=\sum_{n=0}^{N-1}h(n)z^{-n} \tag{2.7}$$

The FIR filter have finite impulse response coefficients and are stable .The FIR filters can be realized in direct forms,cascade forms or by polyphase forms by the basic building blocks such as adder,multiplier and a delay element.

IIR filters:

The IIR filters are described by

$$y(n)=\frac{\sum_{n=0}^{N-1}b_k x(n-k)}{\sum_{n=1}^{M}a_k y(n-k)} \tag{2.8}$$

The IIR filters, though unstable are able to converge faster than FIR filters.

2.3 ADAPTIVE SIGNAL PROCESSING

An adaptive filters models the input-output relationship between two signals in an iterative manner. In linear filter theory, a linear filter is used to model the input-output relation .In the statistical approach of linear filtering, statistical parameters as mean and correlation of desired and noisy signals are assumed to be well defined in a stationary sense. The linear filter has to be designed with a noisy input data .

The optimum filter is usually achieved by minimizing the mean square value of a error signal(MSE),where the error signal is obtained from the difference between desired response and the actual filter output.For stationary inputs the adaptive system converges to a solution known as optimal Wiener solution, which is optimum in a mean square sense. The mean square error is a function of the adjustable parameters of the filter. Therefore, the mean square error (MSE) can be plotted against the adjustable parameters of the filter to generate the error performance surface. The minimum point of the performance surface represents the Wiener solution.

The Wiener filter requires *a priori* information about the statistics of the data to be processed, in other words the signal should be stationary. When these *a priori* informations about the signals are not known, the *adaptive filtering* technique is used.

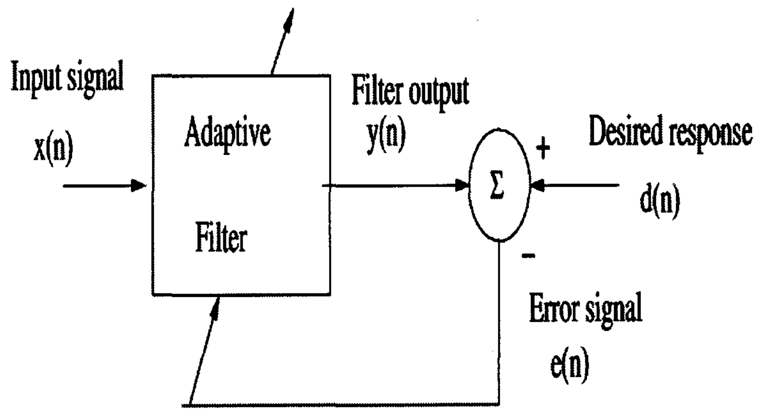

Fig 2.1: Schematic of adaptive filter.

2.3.1 Theory

An adaptive filtering algorithm involves two basic processes:

(i) a filtering process to produce an output in response to an input ,and (ii)an adaptive process,to adapt the adjustable filter parameters used in the filtering process.

The adaptive filtering algorithm can be derived by any of the two distinct approaches, viz., *stochastic gradient method* and *least squares estimation.* Least Mean Square(LMS) algorithm is a simple and widely used algorithm (Widrow and Hoff) for descending the performance surface.

An adaptive processing element or adaptive filter is shown in Fig.2.1.

The input vector to the adaptive filter is

$$X_k=[\, x_{0k}x_{1k}.....x_{Lk}] \tag{2.9}$$

and the corresponding weight vector is

$$W_k = [w_{0k}w_{1k} \quad w_{Lk}]^T. \tag{2.10}$$

The input -output relationship will be

$$y_k = X_k^T W_k = W_K^T X_T \tag{2.11}$$

The error signal of the adaptive filter is,

$$\varepsilon_k = d_k - y_k \tag{2.12}$$

Substituting for y_k, the expression is now

$$\varepsilon_k = d_k - X_K^T W_K = d_k - W_K^T X_k \tag{2.13}$$

Squaring the above error term to obtain the instantaneous squared error,

$$\varepsilon_k^2 = d_k^2 + W^T X_k X_k^T W - 2d_k X_k^T W \tag{2.14}$$

Assuming that ε_k,d_k and X_k are stationary statistically, the expected value of the above equation over k will be

$$E[\varepsilon_k^2] = E[d_k^2] + W^T E[X_k X_k^T]W - 2E[d_k X_k^T]W \tag{2.15}$$

The expectation $E[X_k X_k^T]$ is equal to the autocorrelation function of the filter input and is designated as R. Similarly, the expectation *P* is the cross correlation between the filter input and desired response d(n).Minimizing with respect to W_k gives the Wiener -Hoff equations for the optimum filter coefficients

$$RW_{opt}^T = P \tag{2.16}$$

The mean square error can now be expressed as

$$MSE \equiv \xi = E[\varepsilon_k^2] = E[d_k^2] + W^T RW - 2P^T W. \tag{2.17}$$

The mean square error ξ is a quadratic function of the filter weights. As mentioned earlier , the mean square error (MSE) as a function of the filter weights is a bowl shaped performance surface having a unique minimum.

2.3.2 LMS adaptive filter

An algorithm used for finding the minimum of the error surface for the Wiener solution is also used to find the minimum in the case of an adaptive system. The least mean square(LMS) algorithm is used for descending on the error surface, and is computationally easy and simple to use.

The LMS algorithm considers ε_k^2 as an estimate of ξ_k. Thus, for each iteration in the adaptive system, the gradient of the error surface can be estimated from,

$$\Delta_K = \begin{bmatrix} \frac{\delta\varepsilon_K^2}{\delta\omega_0} \\ \vdots \\ \frac{\delta\varepsilon_K^2}{\delta\omega_l} \end{bmatrix} = 2\varepsilon_K \begin{bmatrix} \frac{\delta\varepsilon_K^2}{\delta\omega_0} \\ \vdots \\ \frac{\delta\varepsilon_K^2}{\delta\omega_l} \end{bmatrix} = -2\varepsilon_K X_K \quad (2.18)$$

where L is the number of filter tap weights. Starting with this estimate and using the method of gradients we have the LMS algorithm as

$$W_{k+1} = W_k - \mu\nabla_k = W_k + 2\mu\varepsilon_k X_k \quad (2.19)$$

The parameter μ is the gain constant that regulates the speed and stability of adaptation. The smaller the step size, smaller is the error in W_K but takes a longer time for the solution to converge.

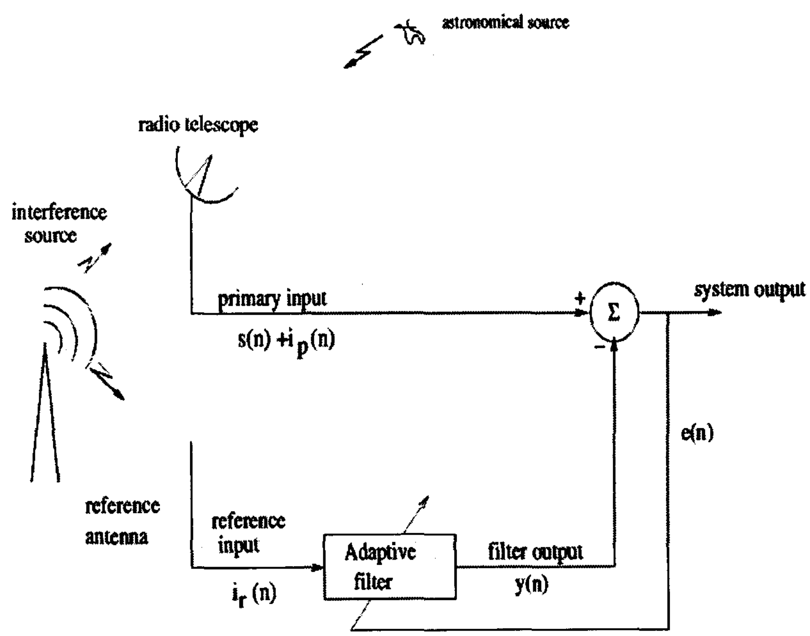

Fig 2.2: Schematic of adaptive interference canceler.

The weight vector converges to an optimal solution when

$$0 < \mu < \frac{1}{(L+1)\text{signalpower}} \tag{2.20}$$

2.4 ADAPTIVE FILTERS IN RADIO ASTRONOMY FOR RFI MITIGATION

Adaptive interference cancellation has been used in communications to reduce the interference. The same technique has been tried out in radio astronomy to suppress RFI (Barnbaum & Bradley, Astronomical Journal ,1998).The adaptive interference cancellation method to mitigate RFI is shown in the Fig.2.2.This technique requires two channels of RF signals for processing. The primary channel ,for example can be the 10.4 m radio telescope (Raman research Institute, Bangalore) which receives both the astronomical signal *s(n)* entering through the prime focus and the interference $i_p(n)$ entering through the side lobes .A reference antenna is connected to the second channel (see Chapter 3) which receives the interference $i_r(n)$.

The interference in the reference channel $i_r(n)$ is correlated with the interference in the primary channel $i_p(n)$.*The* adaptive filter estimates the correlation of the signals .The LMS adaptive algorithm used determines and updates the filter coefficients or the weights .The adaptive filter uses the co-efficients to alter the reference signal, $i_r(n)$ to produce y(n), which resembles the interference in the primary channel $i_p(n)$.The output $\varepsilon(n)$ will be

$$\varepsilon(n) = s(n) + i_p(n) - y(n) \tag{2.21}$$

The output power of the system is given by

$$\varepsilon^2(n) = s^2(n) + [i_p(n) - y(n)]^2 + 2s(n)[i_p(n) - y(n)] \tag{2.22}$$

The expected value of the system output will be now,

$$E[\varepsilon^2(n)] = E[s^2(n)] + E[(i_p(n) - y(n))^2] \tag{2.23}$$

As the LMS $E[\varepsilon^2]$ algorithm minimizes by adapting the filter coefficients ,the term reaches the minimum power:

$$E_{\min}[(\varepsilon^2(n)] = E[s^2(n)] + E_{\min}[(i_p(n) - y(n))^2] \tag{2.24}$$

Since the astronomical signal is constant, minimizing the total system output power minimizes the output interference power.

2.4.1 Performance expectations of the RFI suppression scheme

Any interference suppression scheme must be able to recover the astronomical signal without any distortion. Since the filtering of signals occurs in the reference channel, the astronomical signal coming through the primary channel is not distorted by the interference canceler. The performance of this system is measured by the attenuation of RFI .

The attenuation of RFI is the ratio of the interference power spectra in the system output to the primary input. The attenuation of RFI can be written as a function of the interference-to-noise ratios in the reference channel IN_r.

Typically the rejection achieved in sensitive observations is about 20 dB. This is not adequate for sensitive radio astronomy observations. This motivates for an improvement in the rejection of RFI.

CHAPTER 3

AN EXPERIMENTAL SETUP FOR RESEARCH IN RFI MITIGATION

3.1 INTRODUCTION

The experimental setup for research in RFI mitigation requires two separate channels to receive signals from a telescope and RFI from a reference antenna .We used the 1.4 GHz receiver of the 10.4-m telescope as the primary channel. A reference antenna and receiver system have been built near 1.42 GHz frequency which forms the reference channel. The overall block diagram of the experimental setup is shown in the Fig.3.1.

The two channels are connected to the IF system of the 10.4-m telescope and the digitized voltages are acquired using a data acquisition system. This chapter describes the construction of a reference antenna and receiver system and the data acquisition system used for recording the data

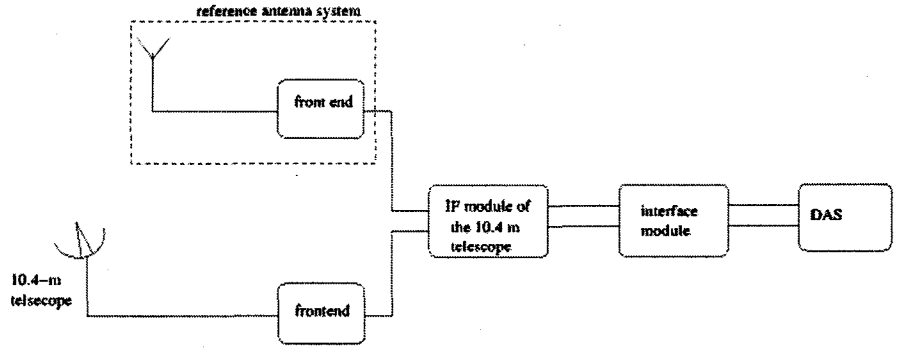

Fig 3.1: Block diagram of the 10.4-m telescope and reference antenna system.

3.2 THE 10.4 M TELESCOPE AND 1.4 GHZ RECEIVER SYSTEM

The 10.4 m diameter telescope at Raman Research Institute, Bangalore is currently used for centimeter wave observations. The telescope has an elevation limit of 15 deg and covers a declination range of –60 deg to 85 deg.The telescope is equipped with a range of spectrometers and is operated at two frequencies ,viz.,(i) at 6.7 GHz with a right circularly polarized helical feed and,(ii) at 1.42 GHz with a dual polarization (V,H) with a conical feed. Both the feeds are mounted at the prime focus.

The 1.4 GHz conical feed is connected to the front end unit comprising low-noise amplifiers for amplifying the low power RF signals . After further amplification in the IF module ,the receiver output is connected to the backend through an interface circuit.

3.3 DESIGN AND CONSTRUCTION OF REFERENCE ANTENNA

The receiver system is required to operate at frequencies near 1.4205 GHz. In radio astronomy, power spectral density is represented in terms of temperature. The thermal noise power available at the terminals of a resistor at physical temperature T K is given by $P = kTB$ Watts, where B is the bandwidth over which power is measured and k is the Boltzmann constant. This implies that a power spectral density, P/B can be represented by an equivalent temperature T.The antenna temperature T_o represents a power per Hertz of value kT_a that is available at the antenna output terminals. The amplifier used for amplifying the sky signal collected by the antenna contributes its own noise. This noise power can be equivalently represented by receiver temperature T_R.Hence, receiver temperature of an amplifier is the noise power(in temperature equiv) that must be added at the input terminal of an ideal noiseless amplifier with the same gain so as to produce the same noise power at the output. The system temperature T_{sys} of any receiver is the sum of the receiver temperature T_R and the antenna temperature T_o.The gain of the reference antenna is determined from the estimated T_{sys} and the power required at the input of the backend used for the experiment. The output power of a reference antenna is then given by,

$$P = GkT_{sys}\Delta v \qquad (3.1)$$

where Δv is the bandwidth of the receiver system and G is the power gain. Considering $T_{amp} = 30K$, $T_{sky} \approx 1K$, $T_{CMRR} \approx 2.7K$, $T_{ground} \approx 10K$, we get $T_{sys} \approx 50K$. For 50 K and a bandwidth of 1 MHz, the input power is -122 dBm. The average noise power required at the input of the backend is -57 dBm. Thus, the gain of the reference antenna is , $G = -57 - (-122) = 65dB$.The other requirement is that the bandwidth of the receiver should be restricted near the front-end to avoid saturation of the IF stage due to RFI. From practical considerations of the filter design near 1.4 GHz , we restricted the bandwidth to 25 MHz.

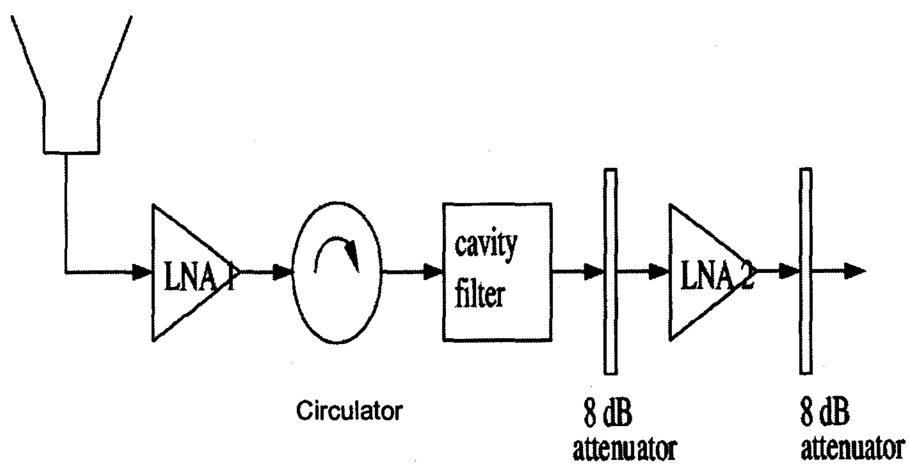

Fig 3.2: Block diagram of the reference antenna and receiver system.

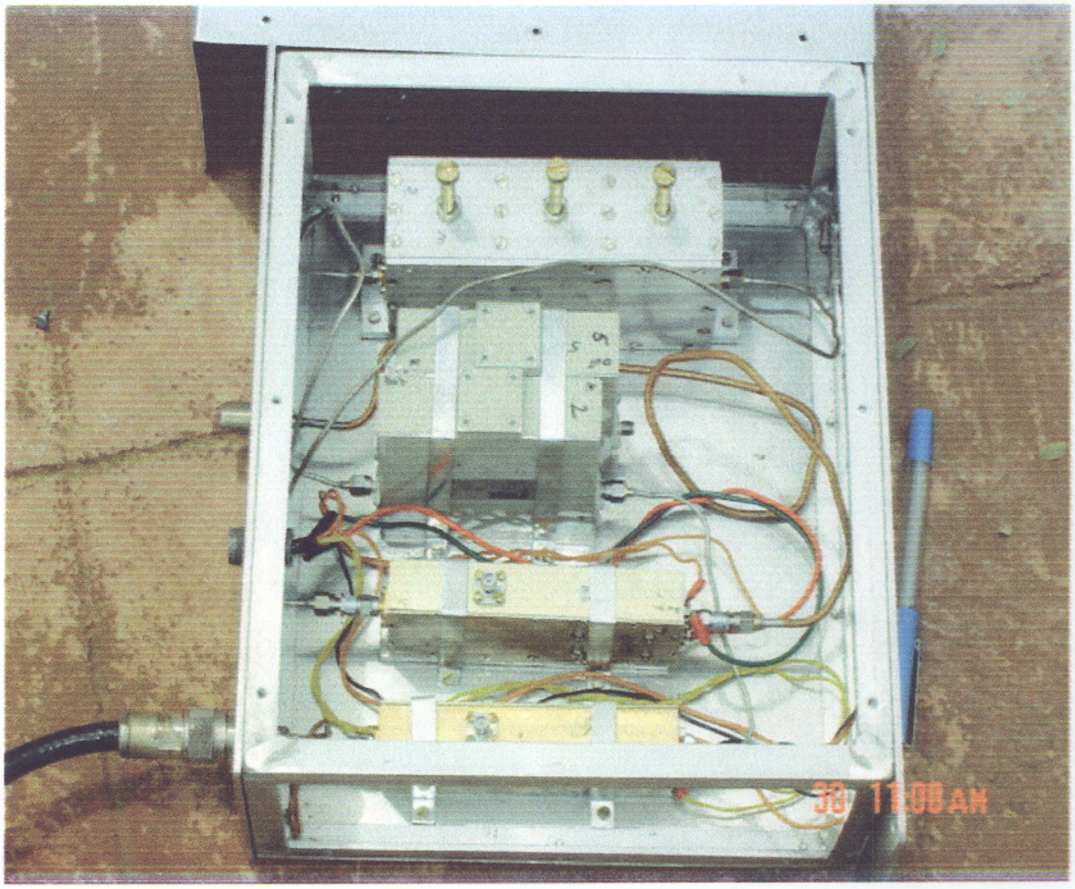

Fig 3.3: Reference Antenna System

The block diagram of the reference antenna system is shown in Fig.3.2. and the photograph is shown in figure 3.3 . A horn antenna is used to receive the RFI signals which is then amplified by a Low Noise Amplifier(LNA).A single linear polarized probe is used to couple the signal to the LNA. This LNA forms the first stage of the receiver system. The LNA has a gain of about 30 dB and a noise temperature of 25 to 30 K over a frequency band of about 500 MHz centered at 1.42 GHz .It is a three stage HEMT amplifier (A.Raghunathan ,2000). The amplified RF signals are passed on to a circulator to prevent the reflection from successive stages from affecting the first stage amplifier. A band pass filter of bandwidth 25 MHz and centered at 1.42 GHz is used to restrict the bandwidth of the signal. The band-pass filters are tunable cavity filters and has been tuned to the required center frequency of 1.4205 GHz and bandwidth .

The output of the filter is further amplified and connected to the IF module of the 10.4-m telescope. To avoid saturation of the IF module we have introduced about 16 dB of attenuation. Attenuators are used to reduce the power levels of a signal by a fixed amount without any distortion and the amount of attenuation is expressed in decibels(dB).The output signal is attenuated relative to the input signal maintaining the input and output impedance close to 50 Ω.Two 8 dB attenuators of Minicircuits, which has a range of operation from DC to 12.5 GHz, is used just after the cavity filter and the final amplifier.

3.3.1 Design and construction of the interface module

The amplified signals after passing through the IF module of the 10.4-m telescope is passed through an interface module(Fig.3.4.)The reference antenna signal and the amplified signal(see section 3.3.2) from the telescope are down-converted to 50 MHz .This is achieved using a frequency mixer. Frequency mixers are devices used to translate the input RF signal to an output IF signal. ZFM -2000 are the mixers used which can be used upto 2 GHz of RF input and has a conversion loss of 9.5 dB. The LO applied is 1370.5 MHz signal with a power of +7 dBm from a Hewlett-Packard 8648C signal generator. The bandwidth of the 50 MHz IF signal obtained is bandlimited to 10 MHz using band-pass filters designed (see section 3.3.3).

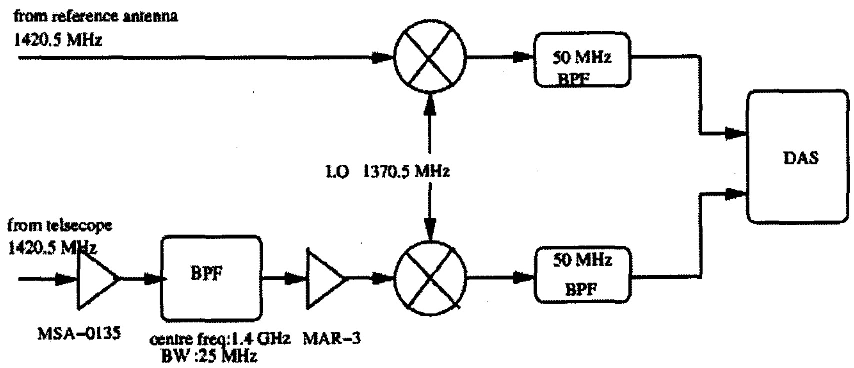

Fig 3.4: Block diagram of the interface module.

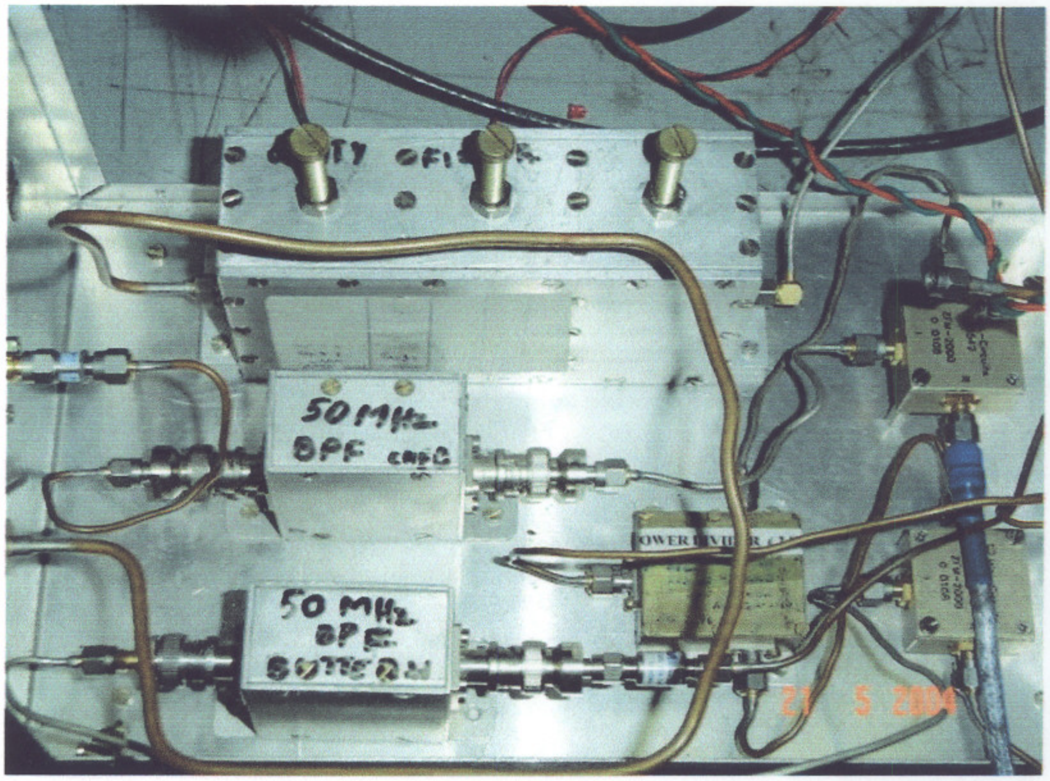

Fig 3.5: Interface Module

3.3.2 Design and construction of the MAR-3 amplifier

The MAR-3 amplifier is a commercially available module from Minicircuits. It is selected since it has a gain of 8 to 10 dB in the desired frequency range. The MAR-3 amplifier operates over a frequency range of DC to 2 GHz with a typical gain of 10.5 dB and bias current I_b of 35 mA. The device voltage V_d at the collector end of the transistor inside the amplifier is +5V .For a supply voltage $V_{cc}$of +15V, the biasing resistance is calculated by

$$R_b = \frac{(V_{cc} - V_d)}{I_d} = 285\Omega \qquad (3.2)$$

The amplifier was realized on a PCB ,with input and output dc coupling capacitors of 100 kpF(surface-mount).The amplifier was shielded using an aluminum box .The measured gain of the amplifier was 9 dB. The figure 3.6 showns the circuit of MAR-3 amplifier and Fig 3.7 shows the photograph.

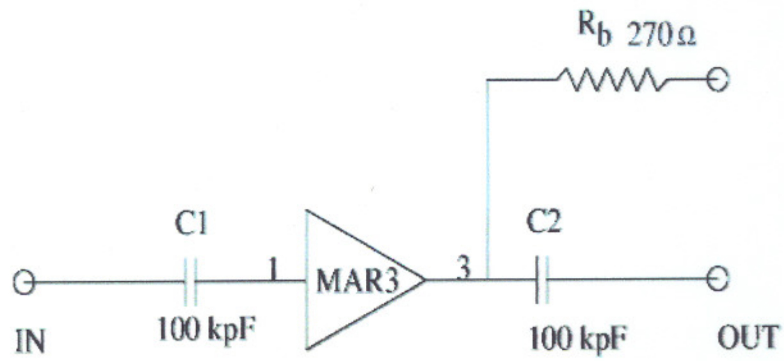

Fig 3.6: Circuit diagram of MAR-3 amplifier

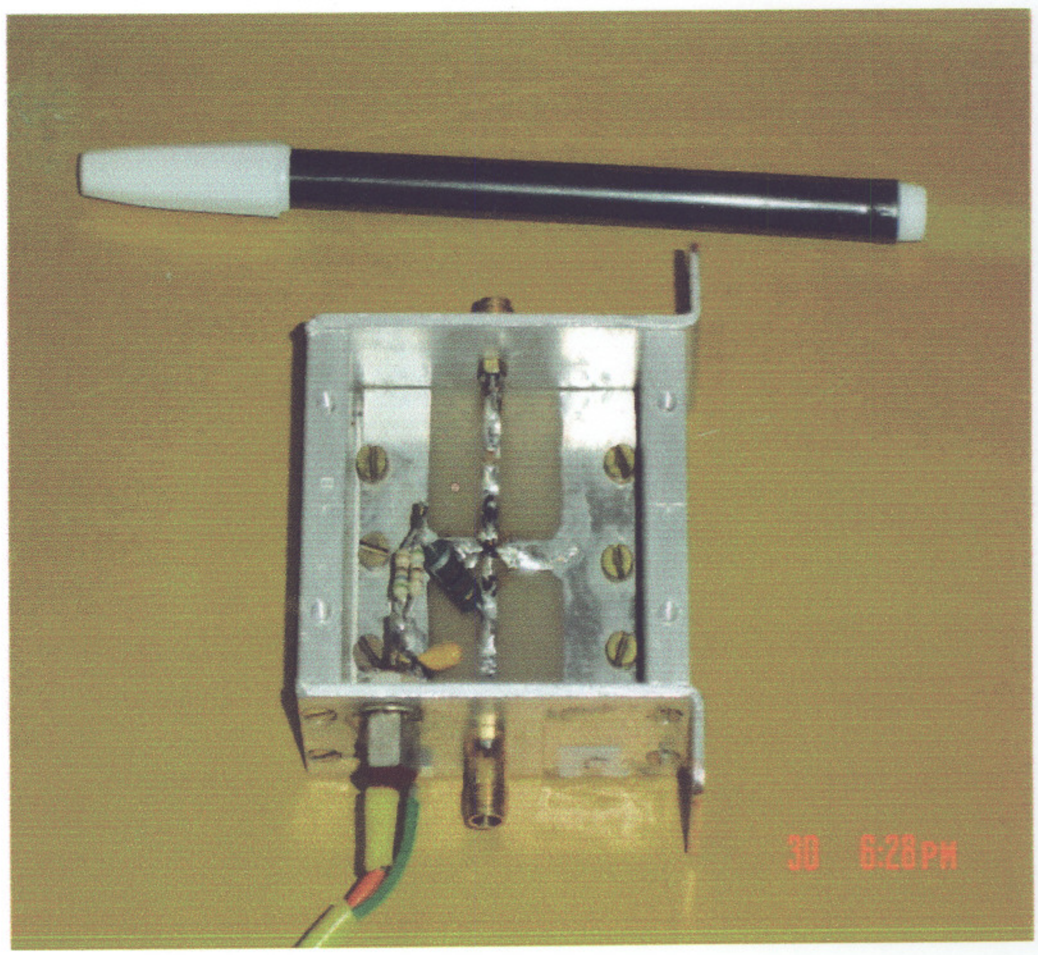

Fig 3.7: MAR-3 amplifier circuit

3.3.3 Design and construction of 50 MHz analog Bandpass filter

The design requirements of the narrow band-pass filter are

1. a centre frequency of 50MHz
2. 3 dB bandwidth of 10 MHz
3. 20 dB minimum at 30 MHz and 70 MHz
4. pass-band ripple < 0.1 dB.

Design : The band-pass filter requirements (Fig.3.5) were first transformed into a normalized low-pass specification. A satisfactory order of the low-pass filter is selected from the characteristics curves given by Anatol I.Zverev (1967) .The geometric mean frequency of the filter for the design is computed as

$$\sqrt{f_l . f_u} = \sqrt{45 \text{ x } 55} = 49.75\text{MHz}$$

The frequencies corresponding to the lower and upper cutoff frequencies computed from this centre frequency are as follows:

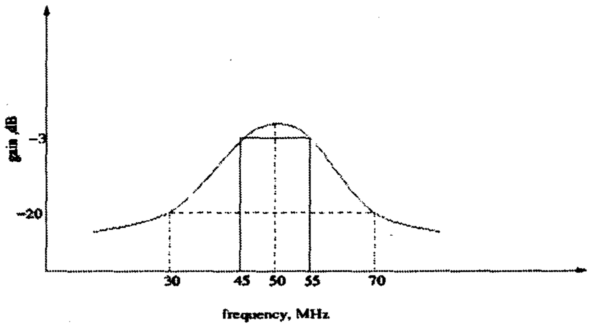

Fig 3.8: The design specification for band-pass filter.

Table 1: Frequency specification for the filter

f_l MHz	f_u MHz	f_u-f_l MHz
30	82.5	52.5
35.35	70	34.65

The second pair of data was more stringent for our specification and so we selected the centre frequency, f_o = 49.75 MHz,BW_{3dB} = 10 MHz,.BW_{30dB} =34.65 MHz. The band-pass steepness factor was determined as

$$A_s = \frac{stopbandwidth}{passbandwidth} = 34.65/10 = 3.46$$

From the attenuation characteristics (Anatol P.Zverev ,1967), the Chebyshev filters (0.1 dB ripple) of order 2 satisfied the requirements. The parameters for the normalized LC low-pass filter are : order of the filter, n = 2, R_s = 1.3554, L_1 = 1.6382 , C_2 = 1.2087 . The low-pass filter is denormalized using an impedance of 50Ω and a frequency scaling factor (fsf) of $2\pi BW_{3dB}$ where BW_{3dB} is 10 MHz. The denormalized values of the filter were determined as:

$$L_1^{'} = \frac{L_2 \times 50}{f_s f} = 1303nH$$

$$C_2^{'} = \frac{C_1}{f_s f \times 50} = 384.7pF$$

The band-pass transformation was done by resonating the capacitor with a parallel inductor and the inductor with a series capacitor, using the centre frequency and its values are:

$$C_1^{'} = \frac{1}{\omega^2 C_2} = 7.75pF$$

$$L_2^{'} = \frac{1}{\omega^2 C_2} = 26.35nH$$

The filter values are optimized by simulating the band-pass filter using Genesys software and the optimized values were determined as $L_1^{'}$ =

1342nH, $C_1 = 7.5pF$, $C_2 = 470pF$, and $L_2 = 25nH$.

Construction: After designing the filter, it was realized on a PCB with the optimum values of the filter. The capacitances of the specified values with a low Q were selected and used. The low value of inductance *,25nH,* was wound with an air core and the higher inductance, 1342nH was wound over a black toroidal core to provide better permittivity and compactness. The inductances were checked for the optimum values using Smith chart in Vector Network Analyser. With the components fabricated, BNC connectors were used for the terminals which can go upto a frequency range of 1 GHz. The filter was shielded in an aluminum box for protection.

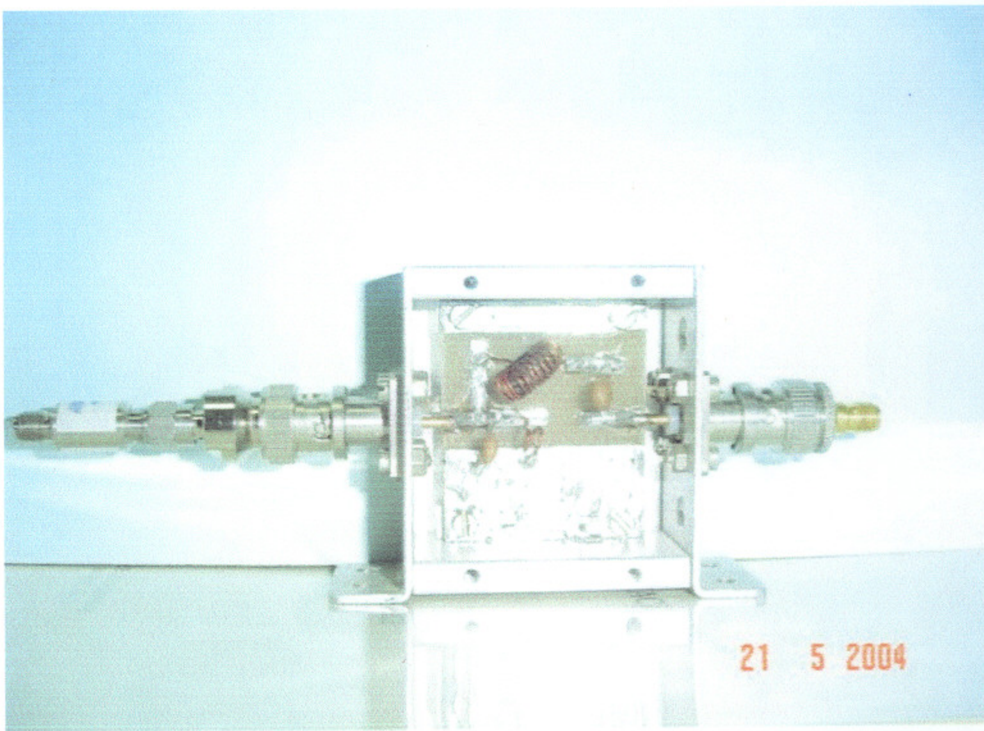

Fig 3.9: Second order Chebyshev bandpass filter

3.4 DATA ACQUISITION SYSTEM

We used an existing Data Acquisition System (Deshpande et al.,2004) to acquire data from the telescope and the reference antenna. This DAS is capable of recording digitized voltages from two channels. The signals from the reference antenna and the 10.4 m telescope after heterodyned to an IF of 50 MHz are fed into the two channels of the DAS. The DAS comprises of four basic units,viz., the IF section, sampler and digitizer, bit-packing logic and the PC add-on card. The block schematic of the DAS is shown in Fig.3.10.

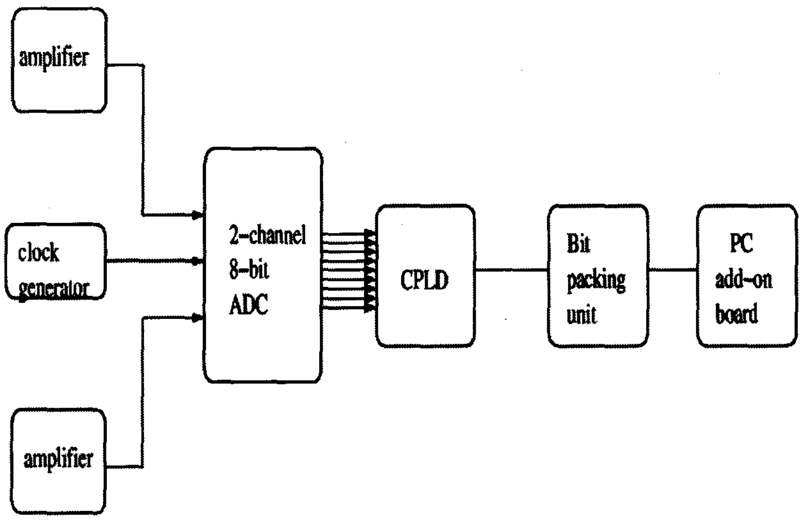

Fig 3.10: Block diagram of the Data Acquisition System.

3.4.1 Block diagram and description of DAS

IF section The signals in the two channels at 50 MHz, with a bandwidth of 10 MHz ,after adequate amplification ,is down-converted to an IF of 10.7 MHz using a Local Oscillator (LO) frequency of 39.7 MHz generated internally in the DAS .The LO frequency is computed internal to the DAS when the RF inputs are specified. After appropriate amplification, the signals are filtered to obtain a bandwidth of 1.3 MHz centered at 10.7

MHz. The signals are further amplified and passed on to the ADC.

Sampler There are two ADC's packed into a single board, which generates an eight-bit data of the signals. Due to limitation of data acquiring rate to the PC, the number of quantization level is reduced to two bits. The conversion of 8-bit to 2-bit is achieved using the digital comparators which are implemented in a CPLD(XC9572XL).The input signal is band--pass sampled using a sampling frequency, f_s of 2.99 MHz. The fourth harmonic of this frequency falls around 11.97 MHz. This harmonic sampling allows us to sample the desired signal band from 10.05 MHz to 11.35 MHz.

Bit-packing A bit packing logic packs 8 consecutive 2 bit samples into 16 words to optimize the data transfer to the PC, since the bus used for data transfer supports a 16 bit data transfer. The bit packing logic is also done to have efficient storage of data in the PC. The bits after being packed into a 16-bit word is multiplexed with another 16-bit of 'marker' data. The marker has its 8 most significant bits (MSB) as zeros with the least significant bits(LSB) acting as counter. For every 8K words of data ,a 'marker' is inserted and the LSB counter is incremented. The counter keeps a track of the 8K blocks of data transferred. The counts are used for checking for slips in the data acquisition.

PC interface card This card functions as an interface between the high speed data acquisition system and a PC. It is provided with two data storage buffers, which are First In First Out (FIFO) memory devices. The data are written into FIFO, and are acquired by the acquisition program.

Post -processing of data Recorded data is searched for the marker to identify any data loss .Consecutive 8K data set are written to an output file with markers set to zero If there is a loss in the data the number of consecutive 8K blocks are written as the marker value.

3.5 PERFORMANCE OF THE REFERENCE RECEIVER SYSTEM

The performance of the receiver system is determined by its dynamic range. The dynamic range of the receiver is the range of signal levels over which it can over which it can operate. An experiment was performed on the reference receiver constructed to determine its dynamic range.

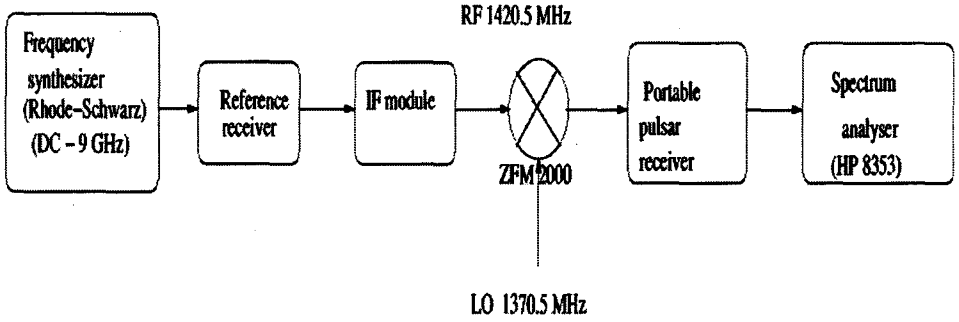

Fig. 3.11: Block diagram of the setup to measure the dynamic range of the reference receiver

The block diagram of the experimental set up to determine the dynamic range is shown in Fig.3.11. An RF input signal of frequency 1420.5 MHz , at increasing power levels from -130 dBm was connected to the input of the LNA in the reference system using Rohde-Schwarz frequency synthesizer. After amplification by the frontend and the IF module, the RF signal was down -converted to 50 MHz using the mixer ZFM-2000 with a LO frequency of 1370.5 MHz provided by the Hewlett-Packard (HP) 8648C signal generator. The output IF power from the input of the sampler was measured using HP 8593E spectrum analyser.

The input power level of the signals was increased from -130 dBm and the corresponding output power level was measured with the following parameters set in the spectrum analyser :

1. Center frequency : 10.689 MHz
2. Resolution bandwidth : 100 Hz
3. Video BW : 100 Hz
4. Reference level : +30 dBm

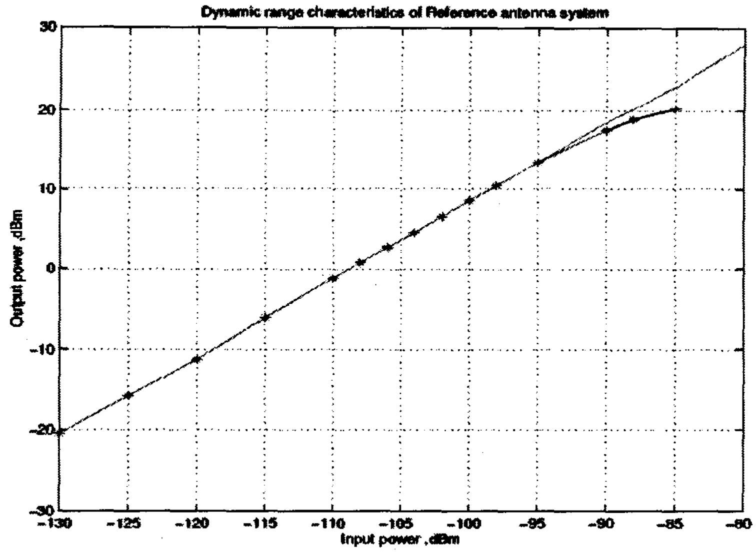

Fig 3.12: Dynamic range characteristics of reference receiver

The plot (Fig.3.12) shows the linear range of operation of the reference receiver .The 1 dB compression point from the plot is -90 dBm of input power. For normal operation the IF power is set about 0 dBm. Thus the receiver system has sufficient dynamic range to withstand strong RFI.

CHAPTER 4

RFI SUPPRESSION IN ADAPTIVE FILTERS

The RFI rejection achieved in conventional adaptive filters has to be improved further for sensitive radio astronomical observations. In this chapter, we study two aspects that affect RFI suppression in a practical, conventional adaptive filter. They are

1. the relative difference between the bandwidth of RFI and that of the primary channel signal.
2. the number of bits used for quantization of signals in the primary and reference channels.

We present the results of this study done on simulated data using MATLAB and on real astronomical data taken with the experimental setup.

4.1 EFFECT OF BANDWIDTH RESTRICTION OF REFERENCE CHANNEL ON RFI SUPPRESSION

We consider the case where the bandwidth of the RFI signal is smaller than the primary channel bandwidth. Such situation is often encountered in Radio astronomy where the sensitivity of continuum observation can be improved by increasing the bandwidth but the RFI occupies a smaller frequency range. The modified block diagram of the adaptive interference cancellation is shown in the Fig.4.1.

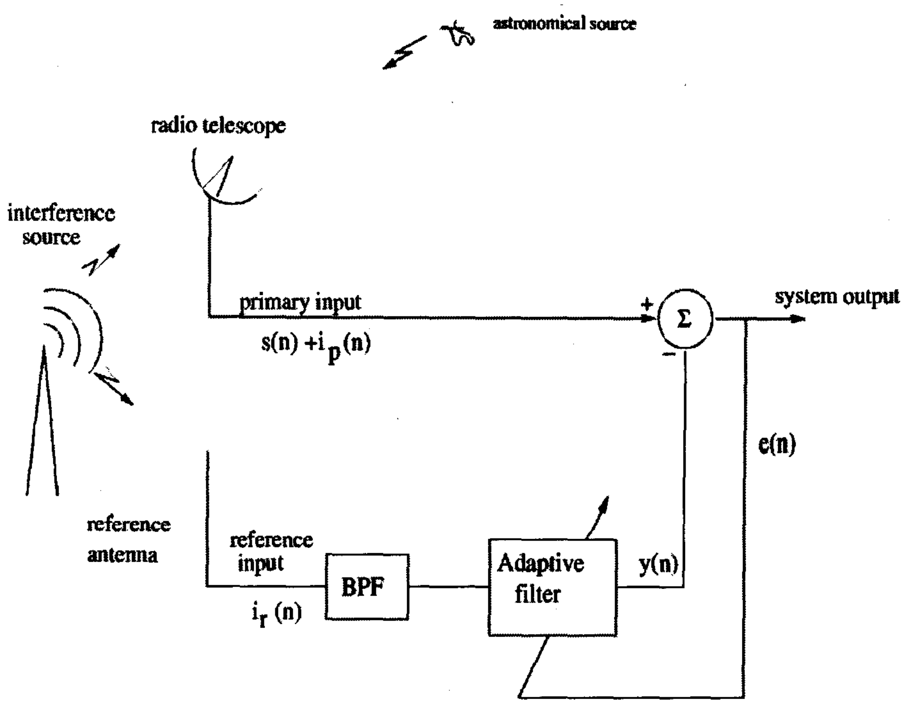

Fig 4.1: Block diagram of the modified adaptive interference canceler.

4.1.1 Simulation

The block diagram of the simulation is shown in Fig. 4.2.The two signals for primary and reference channels are generated as uncorrelated noisy sinusoidal signal. The error signal is obtained as the difference of the primary and reference signals after passing through the adaptive FIR filter. The reference signal is filtered by a band-pass filter thus, restricting the band-width of the signal cantered at the frequency of the sine wave. The adaptive filter is implemented using LMS algorithm . After filtering , the power spectra of the error and input signals are obtained. The RFI rejection is measured by taking the ratio of the power at the channel where the signal (sine wave) is present in the error signal spectrum to that in the input spectrum. This attenuation of RFI is measured as a function of the interference-to-noise ratio in the reference channel. The interference-to-noise ratio(INR) is the ratio of the power of sine wave in the reference signal to the noise power in the reference channel. Since only the reference signal is filtered there is no distortion of the astronomical signal (primary signal).

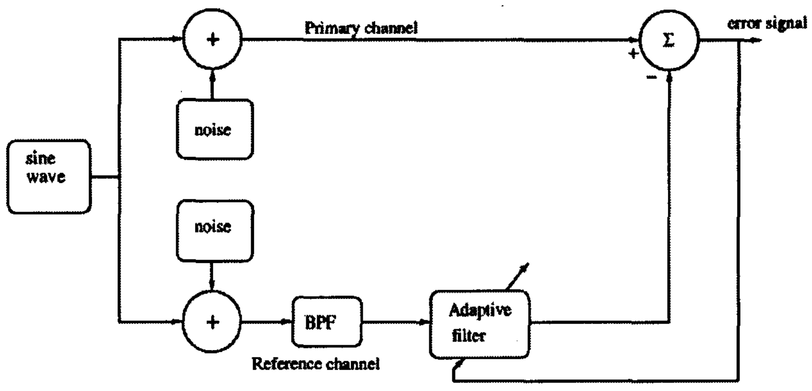

Fig 4.2: Block diagram of the modified adaptive interference canceler for the simulation

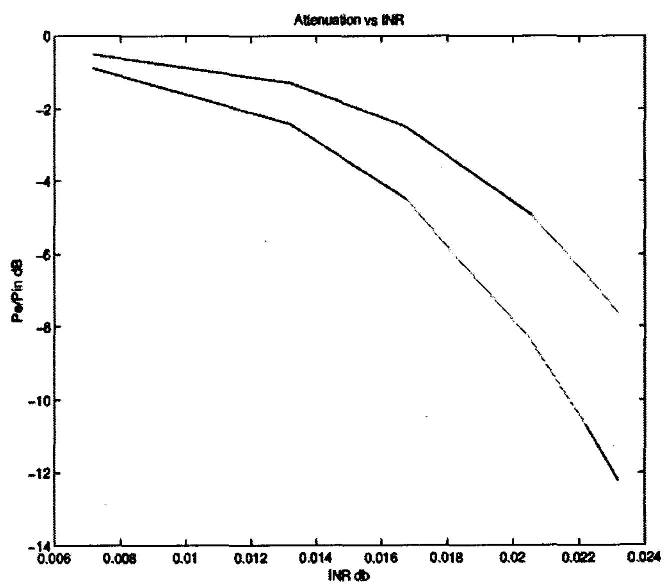

Fig 4.3: Plot of the attenuation of RFI Vs INR in the reference channel with and without the band-pass filter.

The attenuation of RFI is plotted as a function of the interference-to-noise ratio. For comparison we have plotted the attenuation of RFI in the absence of band-pass filter (Fig4.3).The rejection of RFI with the band-pass filter is approximately 5 dB better than the case when no band-pass filter is present. The improvement in the rejection also depends on the INR.

4.1.2 Results from real data

Using the experimental setup we obtained a data set by transmitting RF signals from a frequency synthesizer connected to a log periodic antenna. The block diagram to obtain the astronomical data is shown in Fig.4.4.We used Rohde-Schwarz frequency synthesizer for the experiment and set a frequency of 1420.5 MHz, with a power level of -60 dBm. The 10.4-m radio telescope and the reference antenna formed the two channels for recording the data in the DAS.4000 blocks of 8 kB of data were acquired. The acquired data was then checked for any loss of data .

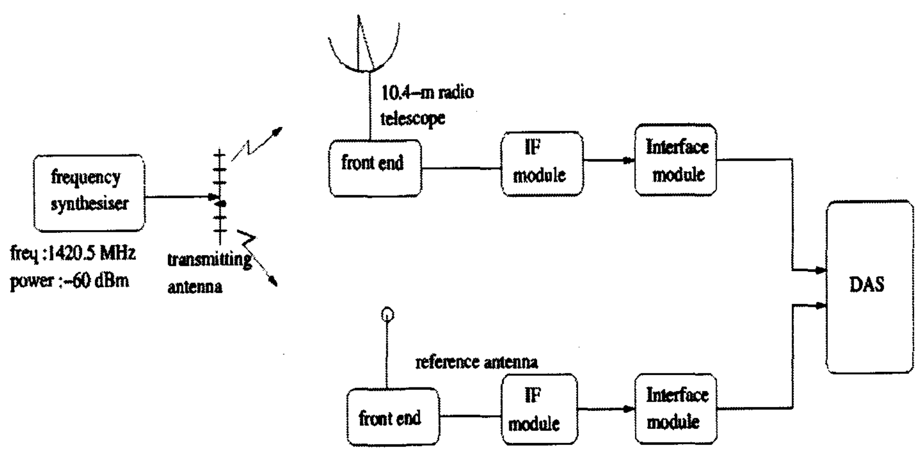

Fig 4.4: The experimental setup for acquiring astronomical data.

In MATLAB, a software program was developed to convert the data acquired into equivalent voltages .Over 3988 blocks of consecutive data were adaptively filtered in MATLAB using LMS algorithm. The adaptive filter gain set to 0.01 and the length of the FIR filter was taken as 10.After adaptive filtering, the power spectra of the error signal and input signal averaged over 1024 data points were taken. The RFI rejection was obtained from these power spectra. The rejection obtained was about 1.4 dB.

To compare the rejection when restricting the bandwidth of the reference channel, we used a band-pass FIR filter of length 20 and pass-band around the RFI signal frequency of 1420.5 MHz and filtered the reference signal by this band-pass filter. The output of this band-pass filter is used as the reference signal for the same adaptive filter used earlier. The RFI rejection is obtained as described earlier. We found that the RFI rejection has improved by about 3 dB. The plots of the power spectra obtained for the astronomical data with and without the bandpass filter are shown in Fig.4.5 to Fig.4.8.

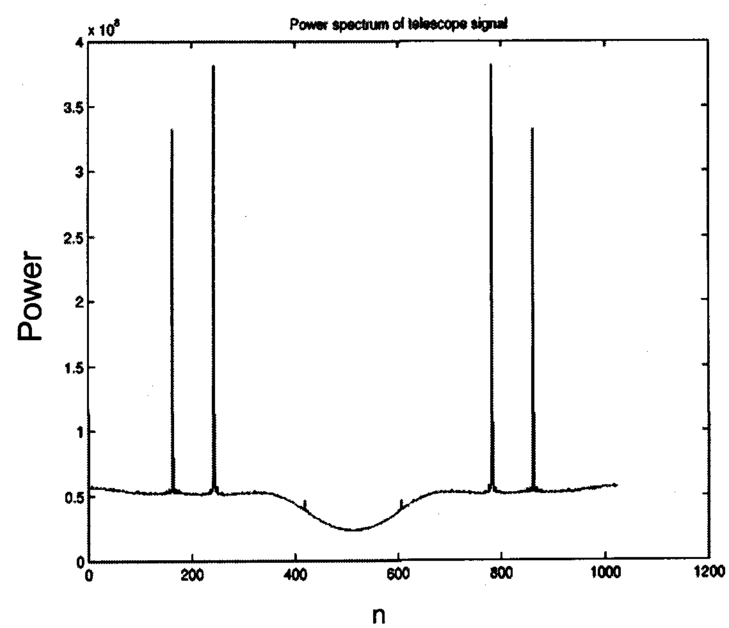

Fig 4.5: The power spectrum of the telescope signal.

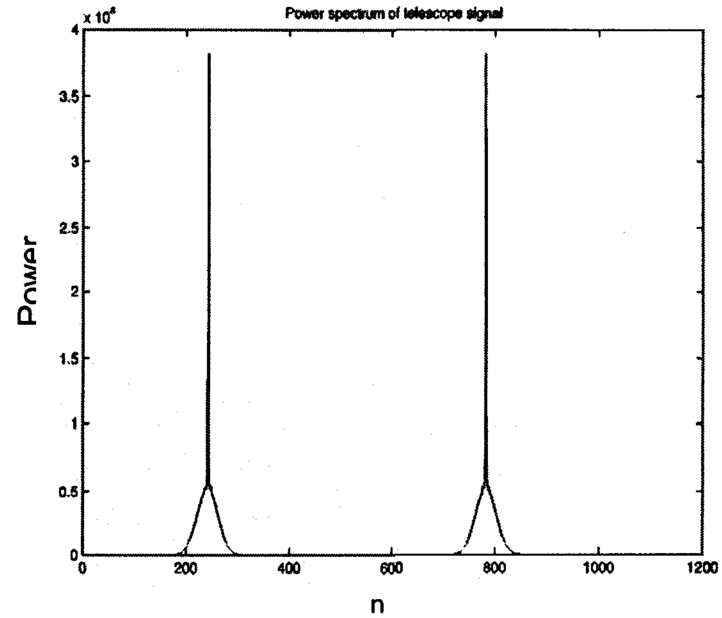

Fig 4.6: The power spectrum of band-pass filtered telescope signal.

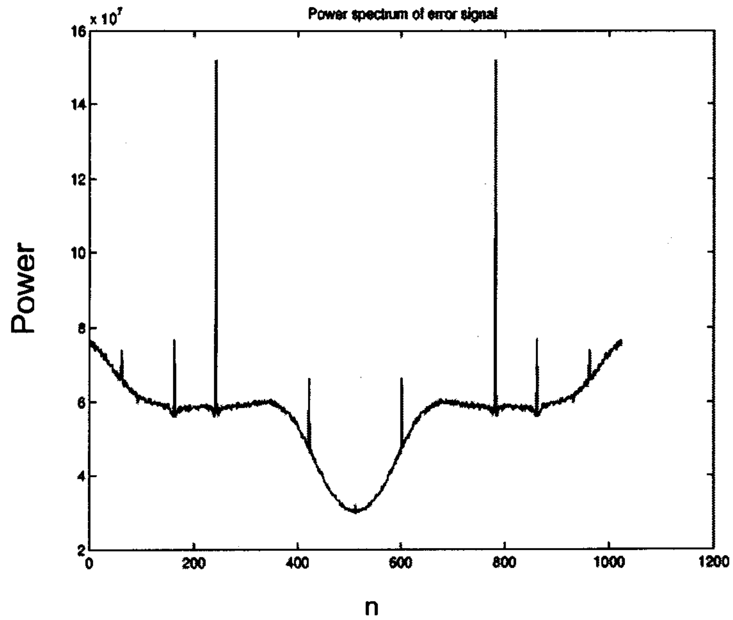

Fig 4.7: The power spectrum of the error signal without bandwidth restriction.

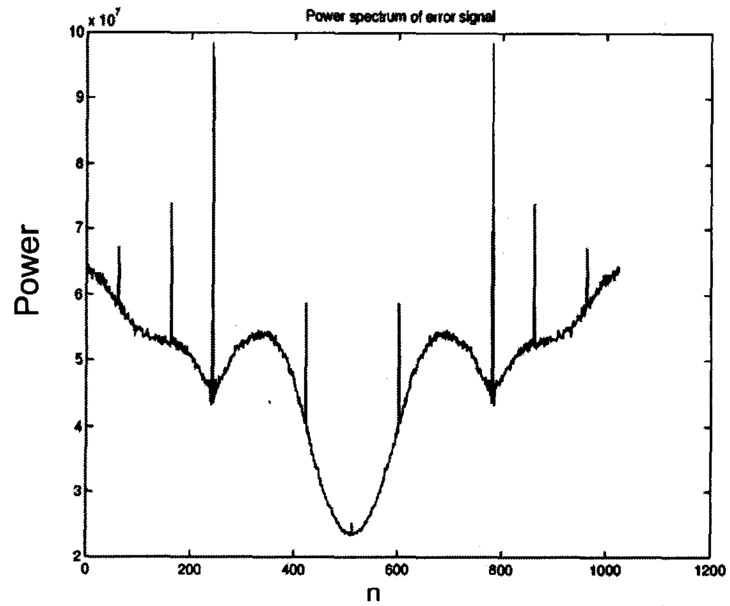

Fig 4.8: The power spectrum of the error signal with bandwidth reduction.

4.2 EFFECT OF THE NUMBER OF QUANTIZATION LEVELS ON RFI SUPPRESSION

4.2.1 Simulation

The block diagram of the simulation is shown in the Fig.4.9. We generated two uncorrelated noise plus a common sinusoidal signals as the primary and reference inputs. These signals are then passed through a quantizer of 'k' bits which has 2^k quantizing levels. The quantized signals were passed through a LMS adaptive filter. The power spectra of the error signal and primary signals were obtained. The attenuation of RFI was measured by changing k from 4 to 1.

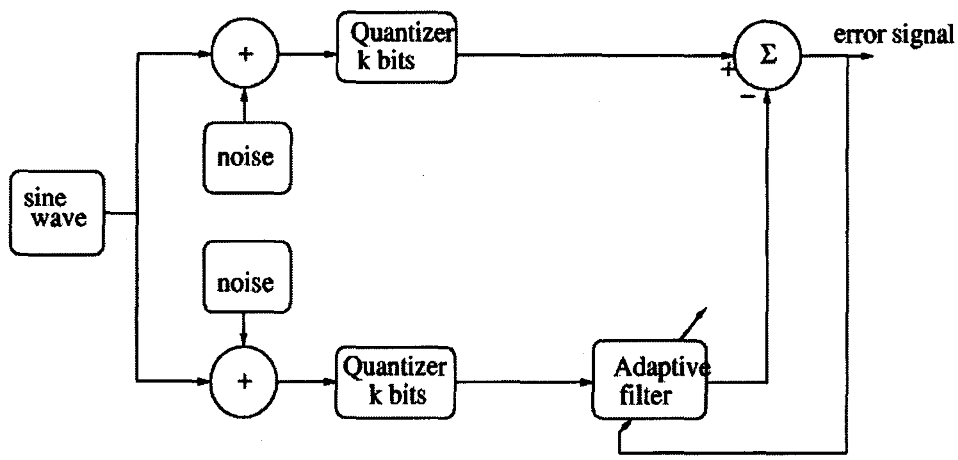

Fig 4.9: Block diagram for the study of effect quantization of bits in RFI suppression.

We have considered two cases for the study.

1. RFI attenuation is measured by changing the number of bits used for quantizing the reference signal. For this case the number of bits used for the primary channel is fixed to 3 bits.

2. The number of bits for the primary channel is changed and that for the reference signal is kept constant at 3.

The results of the simulations for both the cases are shown in Fig.4.10 and Fig.4.11.The simulations show that (i)the attenuation is better for larger number of bits, (ii)the reduction in number of bits of the reference signal affects the RFI attenuation more than that of the primary channel.

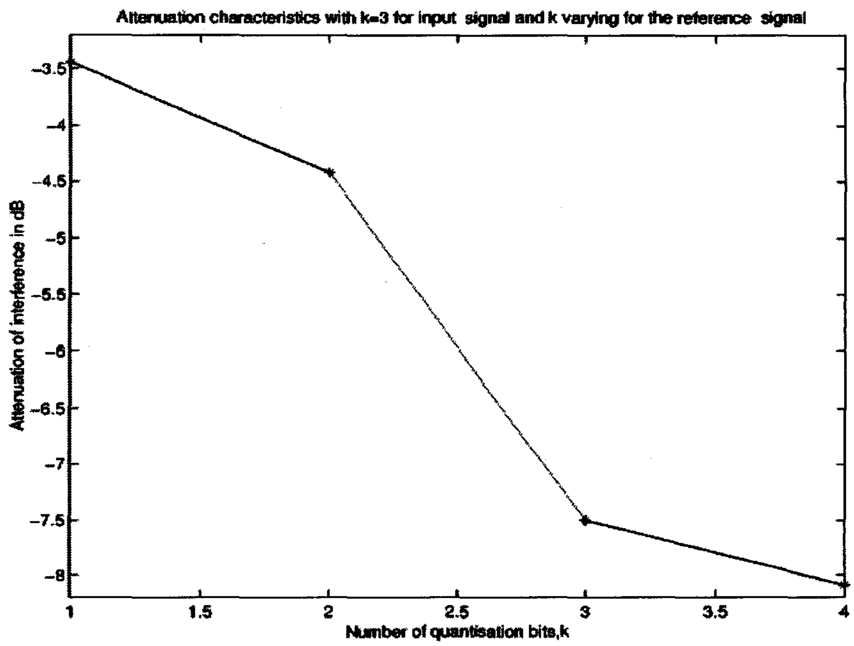

Fig 4.10: Attenuation Vs k characteristics with the primary signal quantized to 3 bits ,and varying k for the reference signal.

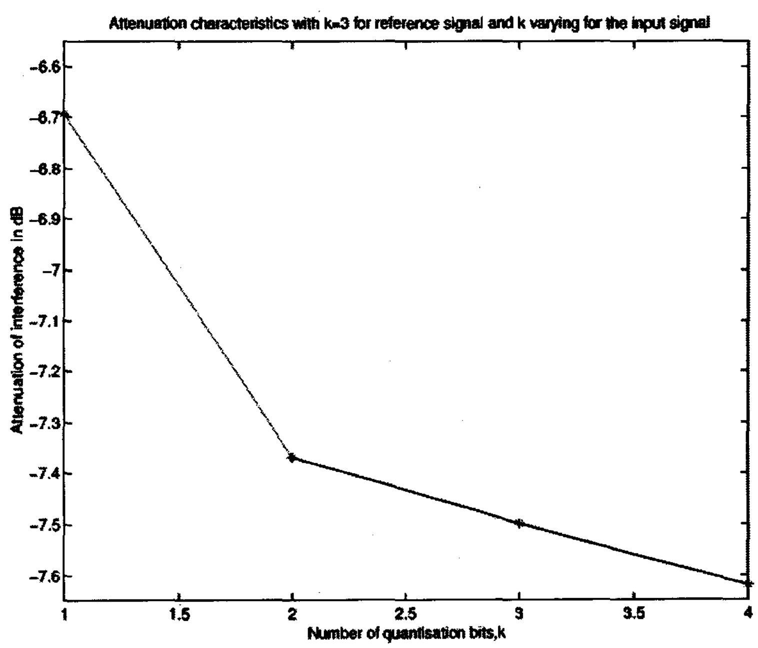

Fig 4.11: Attenuation Vs k characteristics with the reference signal is quantized to 3 bits ,and varying k for the primary signal.

4.2.2 Results from real data

The data acquired using the experimental setup is two bit quantized. To study the effect of the number of bits we reduced the signal quantization to one-bit and passed the data through the adaptive filter .We found that the attenuation achieved for 2 bit quantization of signals is better than that achieved for 1 bit quantization.

CHAPTER 5

CONCLUSION

For sensitive radio astronomy observations the RFI suppression achieved in conventional mitigation techniques are not adequate.This limitation demands the development of new techniques which give better RFI suppression. Adaptive interference cancellation which uses a reference channel to pickup RFI and subtract it from a contaminated signal in a primary channel is successful to some extent. To develop methods to improve RFI suppression in adaptive RFI cancellation technique, we have designed and built an experimental setup comprising of a reference and a primary channel. The 1.4 GHz receiver system of the 10.4-m telescope was used as the primary channel. A reference antenna and receiver system operating near the frequency 1.42 GHz were built. An interface module to connect the reference and telescope outputs to an existing DAS was also built. After constructing the reference antenna and receiver system, we measured the dynamic range of the system and the found that the 1 dB compression point corresponds to an input power of -90 dBm. This dynamic range is adequate to withstand strong RFI.

Two aspects that affect RFI suppression in adaptive interference cancellation are the relative bandwidth reduction of the RFI signal compared to the primary channel and the number of bits used for quantizing the reference and primary channel signals. These aspects have been studied through simulations using MATLAB and using real data obtained using the experimental setup. The simulation results have shown an improvement in suppression of about 5 dB when the reference signal is reduced in bandwidth compared to the case when the full bandwidth was used. With the real data, the improvement in RFI rejection achieved was about 3 dB.

The simulation to study the effect of the number of bits used for quantization have shown that higher the number of bits used for quantization, the better is the suppression. In the case of the quantization of reference signal the RFI suppression is improved by 4.5 dB when the number of bits were increased from 1 to 4.The tests on real data have also showed that RFI suppression is better with a two bit quantization over the one-bit quantization.

Future scope : Till now we used LMS adaptive filter. The relative gain between the primary and reference channels is modeled using a FIR filter. We would like to use an IIR filter for better modeling of the gain. Further extension of adaptive filters such as Kalman filters can be tried out to improve RFI suppression. Also the astronomical data obtained using the present experimental setup is two bit quantized. The RFI suppression can be improved by further increasing the quantization levels for the real data.

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