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A PROJECT REPORT ON

BANDSHAPING CIRCUITS FOR THE NEW IF CONVERSION SYSTEM

BY

ASHOK BOSE
B.TECH, 3rd YEAR
INSTITUTE OF RADIO PHYSICS & ELECTRONICS
UNIVERSITY OF CALCUTTA

UNDER THE GUIDANCE OF

SHRI B.AJITHKUMAR

GIANT METREWAVE RADIO TELESCOPE
NATIONAL CENTRE FOR RADIO ASTROPHYSICS
TATA INSTITUTE OF FUNDAMENTAL RESEARCH

ACKNOWLEDGEMENT

To carry out a task successfully, it is very important to have a good vision and guidance. My guide provided me the same. I would like to extend my thanks to my guide Prof. B. Ajithkumar for his incessant interest in my regular day to day work whenever it was possible for him. I express my deep gratitude for the patience and step by step assistance he extended towards me in completing the project. Prof. T. L. Venkatsubramani also helped me in getting acquainted with the GMRT system.

My respect to Prof. Yashwant Gupta and Prof. Rajaram Nityananda for providing us all the support and entrusting his belief in assigning the project.

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My thanks to my Teachers at The Institute of Radio Physics & Electronics. Finally thanks to my friends Suman, Partha, Soumo and my senior Somak Bhattacharyya for assisting me in my project.

ASHOK BOSE

ABSTRACT

The Software based correlator of GMRT is currently being developed as an alternative to the existing hardware correlator system. In the new system being planned RF signals from the antenna received at the CEB at IF of 130MHz & 175MHz are converted to suitable frequencies and digitized using PC based ADCs. Now since the ADCs are capable of sampling an input analog signal of 66MSPS it is necessary for this input signal to have a maximum bandwidth of half the sampling rate. So to maintain this criterion it is important that the signal be adequately filtered before sampling & to produce the undistorted waveform of the wanted signal.

My project revolved around in the making of a bandpass filter. This bandpass filter should have a -6dB fall for a bandwidth of 32MHz centering around 130MHz & 175MHz respectively. Moreover it should fall to -40dB, 10MHz away from the 32MHz bandwidth region. The desired filter was designed using LC configuration and then simulated. Later it was practically designed and the response was obtained from the Network Analyzer.

CONTENTS

AN OVERVIEW OF GMRT

1.1 Introduction

Radio Astronomy is a part of observational Astronomy. It is the study of planets, stars, galaxies & other astronomical phenomena using radio waves they emit. Since wavelength of radio waves are longer than that of light, radio astronomy require large antennas.

Giant Meter-wave Radio Telescope (GMRT) is a project run by National Centre for Radio Astrophysics (NCRA) under Tata Institute of Fundamental Research (TIFR) which provides us with the knowledge of celestial process with the help of one of the largest & powerful telescope observatory operating at meter wavelength.

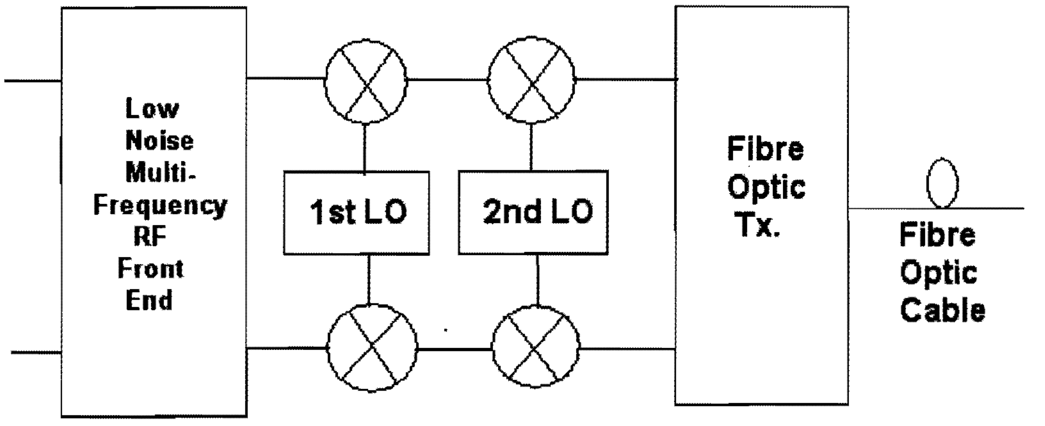
GMRT, situated at Khodad, 80 km north of Pune, has 30 parabolic antennas of 45 m diameter, spread over a distance of about 25 km. Operating frequency ranges from 50 MHz to 1450MHz. Antennas constructed using a technique named SMART & their reflecting surfaces consists of panels of wire mesh . The location of antennas & their number was optimized to meet the principal astrophysical objectives which require sensitivity at high angular resolution as well as ability to image radio emission from less fused extended regions. 14 of 30 dishes are located randomly in the central array within 1 sq km , remaining are spread out along 3 arms of a 'Y' shaped configuration ,each 1 km apart .

1.2 GMRT Receiver System

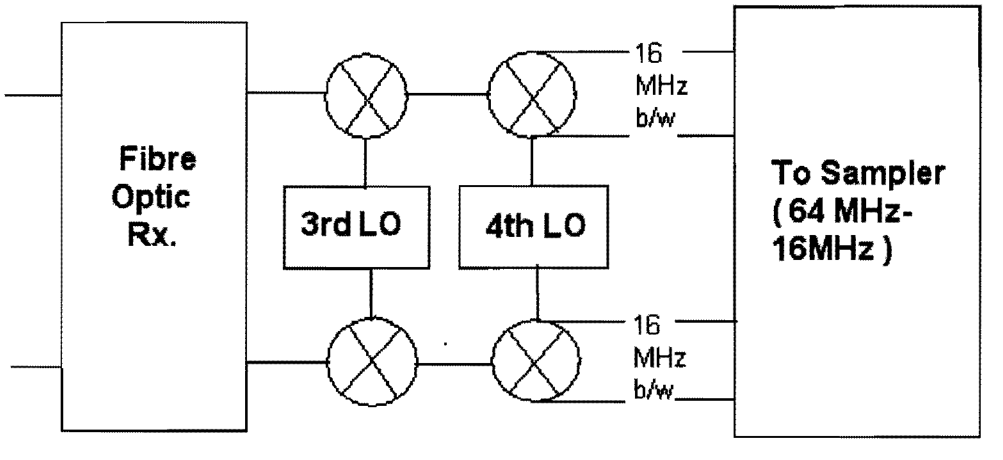
In present GMRT system for the antennas , the array will operate in six frequency bands centered around 153 , 233 , 325 , 610 , &1420 MHz with sub bands of 1060 , 1170 , 1280 , 1390 MHz in the L-band . All these feeds provide dual polarization outputs. Here the RF signals in two polarizations are passed through two channels each of maximum b/w 32 MHz . Then each of these signals are down converted to 70 MHz & then to 130 &175 MHz respectively in the IF & LO system. The signals are then combined & sent to Central Electronics Building (CEB) through optical fiber link. Signals are then processed in the Baseband system, where each polarization is converted into two sidebands of 16 MHz b/w. Thus a total of 4 baseband cannels are available from each antenna which are sampled & digitized with ADC with sampling frequency 32MSPS .

Now an upgraded GMRT receiver system is being planned with improved sensitivity, instantaneous b/w & more facilities. In the new scheme a wider b/w signal will be digitized with less electronics at the remote antenna sites. The current idea is to send the entire RF spectrum received at the feeds to the CEB & further processing is to be done there. The possibility of direct down conversion and digitization of the signals is being investigated with the concept of Bandpass Sampling. These signals are further processed using hardware based correlator.

Current Receiver System



Antenna Electronics



CEB Electronics

SOFTWARE BACKEND

2.1 Description of the Software Backend

A Software based correlator is currently being developed as an alternative to the existing hardware correlator. In the new system being planned RF signals from antenna received at the CEB at IF of 130 & 175 MHz are converted to suitable frequencies and then digitized using PC based ADCs. The digitized data is further processed in powerful computers using software.

The software backend at GMRT used for interferometric observation is a 32 MHz wide FX correlator which can go upto 100 MHz.

Each antenna is dual polarized and each polarized signal (which has a maximum b/w of 32 MHz) is digitized in a separate ADC channel. So 30 antennas needs 30*2=60 ADC inputs of 32 MHz b/w with a sampling frequency of 66MSPS. So there is no need to split the received signal into USB and LSB in the current receiver.

PCI based quad input ADCs are used for digitizing the data. In the software backend 16 such ADC cards are used, 8 for each polarization channel. Each ADC card is installed in a separate PC, then a total of 16 PCs are used for data acquisition and another 32 PCs are used for data processing.

The 8 data acquisition PCs are to be provided with some clock pulse signal & should be triggered together.

2.2 Advantages of the Software Backend

The software backend system is better than the hardware system because of the following reasons:

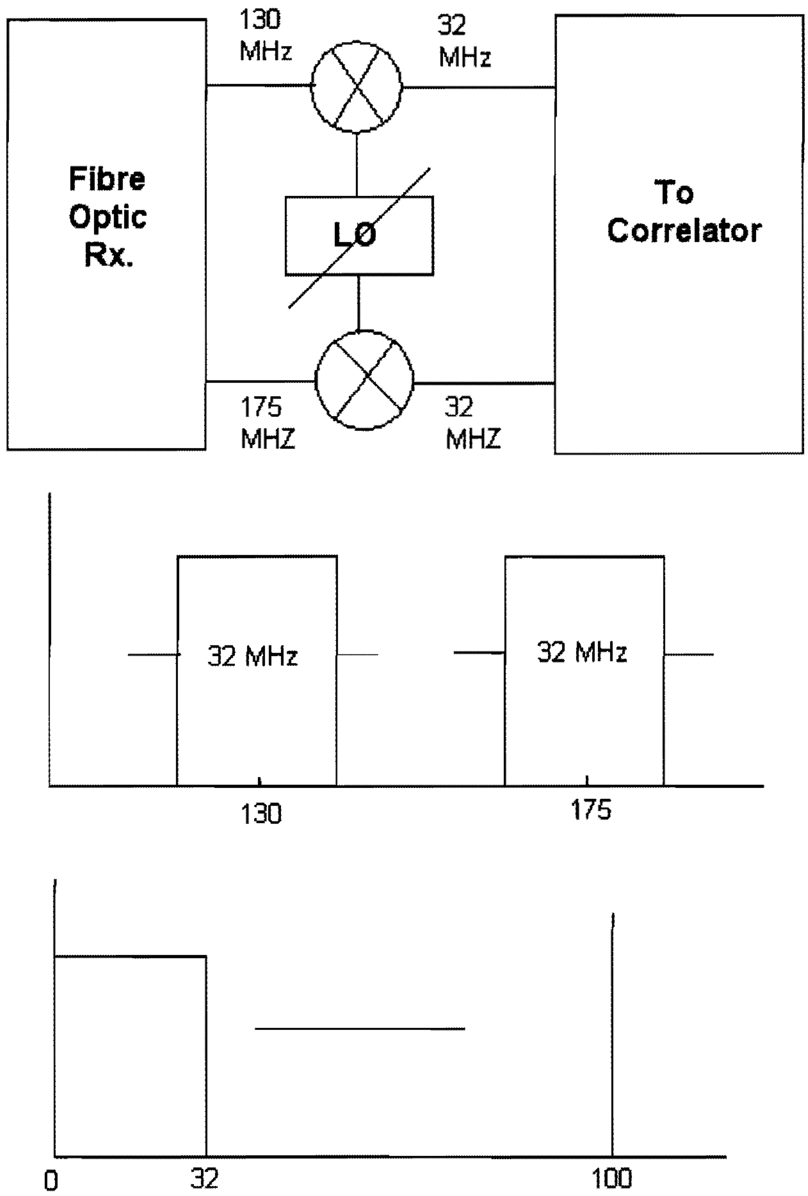
1. Bandwidth of the software system can reach upto a maximum of 100 MHz but at present upto 32 MHz is used.
2. Reconfiguring of the system is easier than that of the hardware portion.
3. Standard components are used & only changes in the program are to be made for any discrepancies rather than discrete hardware components.

CONVERSION OF IF SIGNALS

3.1 New IF Conversion

The IF signals obtained from the output of the optical fiber receiver have a b/w of 33 MHz at -1dB respectively at 130 and 175 MHz centre frequency . Now this 33 MHz b/w signal at -1dB is converted to 32 MHz b/w at -6dB signal. This 32MHz bandwidth signal can range from 0-32MHz, 32-64MHz & 64-96MHz with a variable LO which are suitable for ADCs in the software correlator & for the requirement of bandpass sampling explained later.

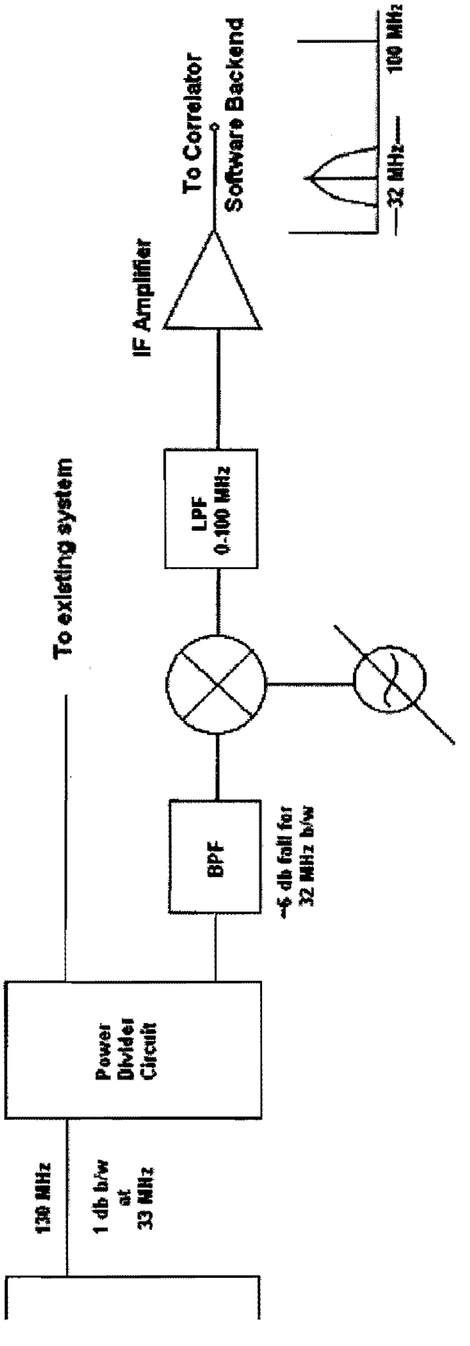
New IF Conversion



3.2 The New System

The new system will be effective & work simultaneously with the existing one as shown in the diagram below. The output from the OF Rx. is passed through a power divider circuit, one for the present system & other for the new one. A BPF is incorporated with required specification. The signal is then mixed with a variable LO & finally passed through a LPF & IF Amplifier to the correlator.

Block Diagram Of New System



3.3 Bandpass Sampling

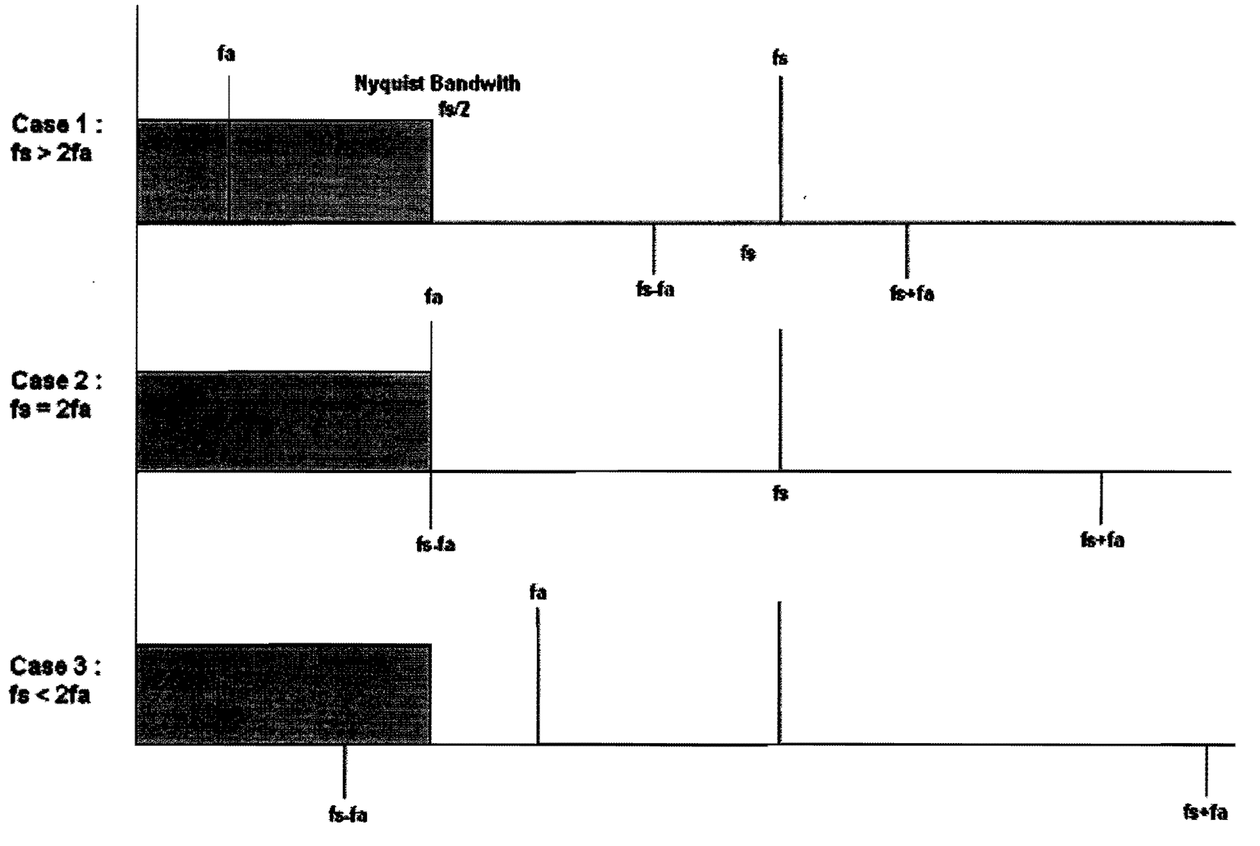
Before we get to know the importance of 32 MHz BPF we must be familiar with the technique of Bandpass Sampling. A continuous analog signal to be sampled at discrete intervals must be carefully chosen to insure accurate representation of the original signal. This leads us to Shanon's information theorem & Nyquist criterion:

1) An analog signal with a bandwidth of fa must be sampled at a rate of fs > 2fa in order to avoid the loss of information.
2) The signal bandwidth may extend from DC to fa (Baseband Sampling) or from f1 to f2 where fa = f2-f1 (Under sampling, Bandpass Sampling, Harmonic Sampling, Super-Nyquist)
3) If fs < 2fa, then a phenomena called Aliasing will occur.
4) Aliasing is used to advantage in under sampling applications.

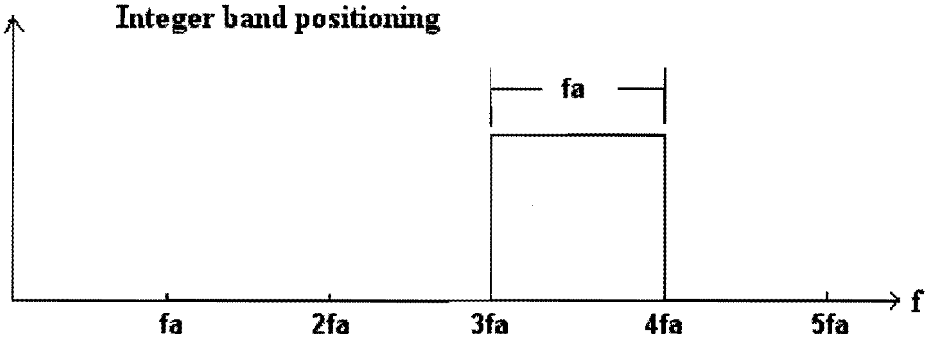
3.4 Frequency Domain Effects of Sampling

Sampling of analog signal fa at a sampling rate fs produces two alias frequency components fs+fa & fs-fa. The effects of sampling will cause either the actual signal or an aliased component to fall within the Nyquist b/w between DC & fs/2. Therefore any signal which fall outside the b/w of interest, whether they be spurious tones or random noise, must be adequately filtered before sampling. If unfiltered, the sampling process will alias them back with the Nyquist b/w where they can corrupt the wanted signal.

Effects Of Bandpass Sampling



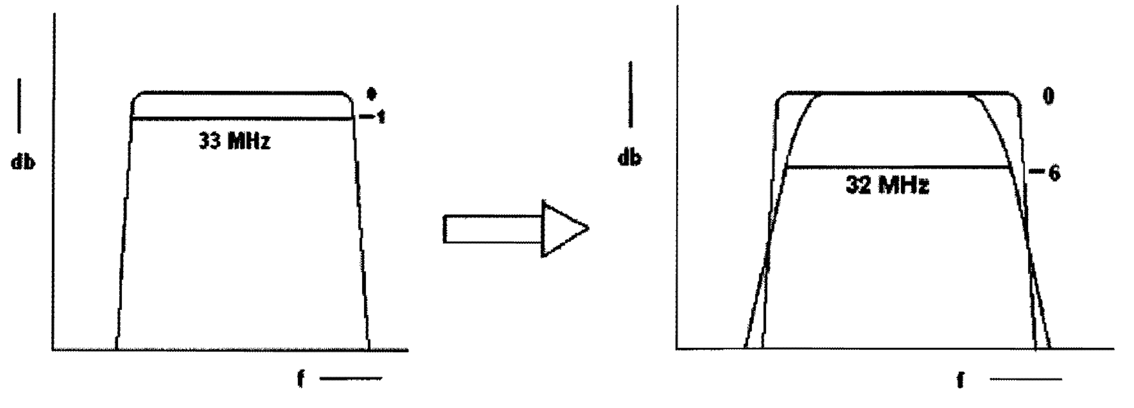
Another important aspect of this technique is ' band position ' which refers to the fractional number of bandwidths from the origin at which the lower band edge resides. A special case is 'integer band positioning' , which holds when the band is located at an integral number of bandwidths from the origin ,i.e. ,fl = c(fu-fl) , c=0,+1,+2,....The theoretical minimum rate fs = 2fa is seen to apply for integer band positioning .



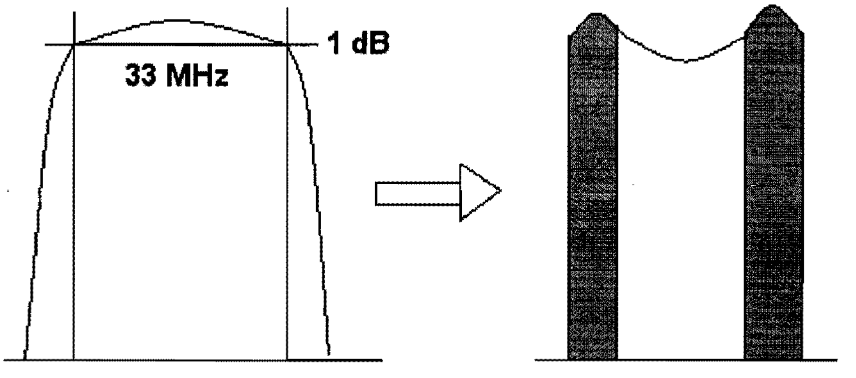
3.5 Need for Band shaping Filter

As stated earlier for sampling a particular signal Nyquist criterion is to be maintained. Hence a signal at -1 dB with 33 MHz b/w is converted down to 32 MHz b/w at -6 dB since the ADCs can handle a clock frequency of
66 MSPS. Moreover it is essential to maintain the required band position. So this 32 MHz b/w signal with centre frequency of 130MHz &175MHz respectively is mixed & passed through a Low Pass filter such that the ultimate signal satisfies Nyquist criterion & lie in the range which are multiples of the bandwidth. This is necessary to avoid signals which fall beyond the Nyquist bandwidth which may fall back and corrupt the wanted signal.

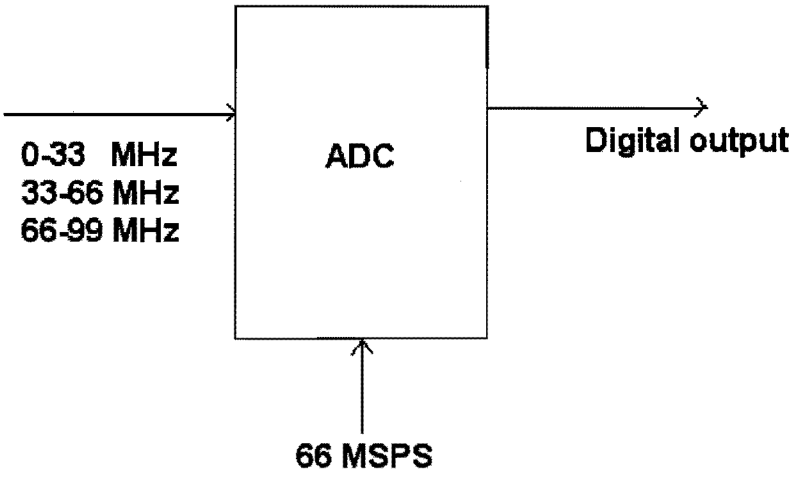
Need for Band shaping Filter



Distortion produced if Nyquist criterion is not maintained



Digitization of required Bandpass signal



FILTER

4.1 Filter Basics

Real world signals contain both wanted & unwanted signals. Therefore it is essential to separate the two. The process of separation of these two signals is called ' Filtering ' and the circuit used for this purpose is a 'Filter'.

4.2 Filter Classification

Filters can be classified on the basis of the type of components used, the band of frequency they select and the nature of the transfer function which tries to approximate the ideal characteristics.

Depending upon the passband & stop band location filters are basically divided into four types:

1. Low pass
2. High pass
3. Band pass
4. Band stop

All the above filters can be constructed using different methods:

1. **Butterworth** - These filters are characterized by the fact that it has no ripple in the passband or stop band and has monotonically decreasing passband. These filters are all pole filters.

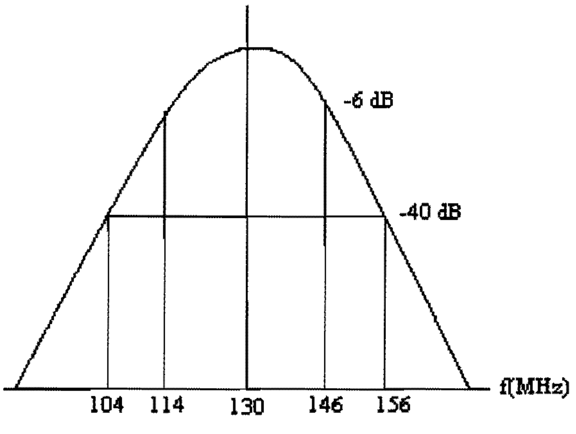
2. **Chebyshev** - The Chebyshev response is characterized by the presence of ripple in the passband and no ripple in the stop band. The ripple can be controlled and is directly proportionally to the SWR and Reflection coefficient. The cutoff frequency is specified at an attenuation equal to pass band ripple. The Chebyshev response is more selective than Butterworth response at the expense of insertion loss and greater group delay.

3. **Elliptic** - The Elliptic response is characterized by the presence of ripple in both passband & stop band. This response is more selective than the above two but exhibit more group delay variation in the passband.

BANDPASS FILTER DESIGN

5.1 Bandpass Filter specification with centre frequency of 130 MHz

1. Minimum inductor elliptic
2. Cauer-Chebyshev Normal
3. -6dB fall for a b/w of 32MHz
4. -40dB fall for a b/w of 52MHz
5. 50Ω input and output impedance.



5.2 Order Estimation

Geometric Centre Frequency, $f_0 = \sqrt{(f_l f_u)}$
$= \sqrt{(114*146*10^{12})}$
$= 129.01$ MHz

Corresponding geometric freq. for each stop band freq. is calculated from ,$f_1 f_2 = f_0^2$

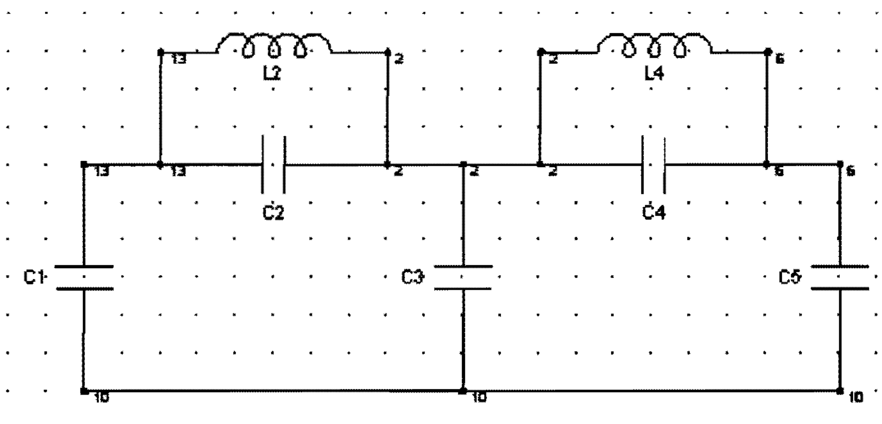
f1	f2	f2-f1
104	160.03	56.03
106.69	156	49.31

$f_0 = 129.01$ MHz, $BW_{6dB} = 32$ MHz, $BW_{40dB} = 49.31$
Bandpass steepness factor , $A_s = BW_{40dB}/BW_{6dB} = 49.31/32$
$= 1.54$

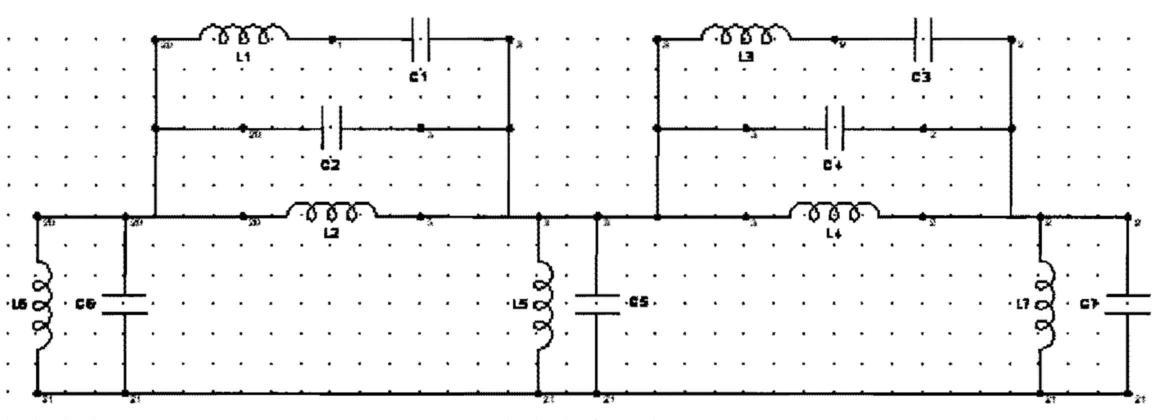
5.3 Design Procedure

Let us choose ρ of 20% which corresponds to a ripple of 0.18 dB & A_ρ of 13.9dB *
(since $A_\rho = 20\log |1/\rho|$)
$A_{min} + A_\rho = (40 + 13.9)$ dB $= 53.9$ dB
$\therefore \Omega_s^* = 1.54$
Hence, order = 5

Hence for the above specification, from table* C0520
$\Omega_s = 1.5243$, $\Theta = 41.0$ $A_{min} = 46.88$
C1 = 1.18411
C2 = 0.14706
C3 = 1.77455
C4 = 0.41894
C5 = 0.97798
L2 = 1.20241
L4 = 0.95213

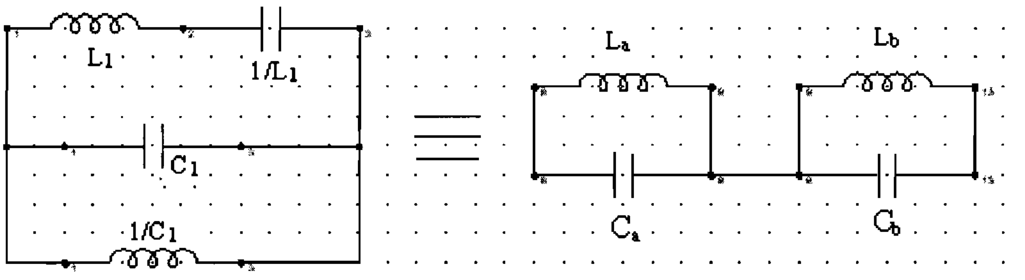


Bandpass $Q_{bp} = f_0/BW = 4.03$
After this we multiply all inductances and capacitances by Q_{bp}, then transform the network into a BPF centered at $\omega_0 = 1$ by resonating each cap. With a parallel inductor & each ind. with a series cap. The resonating elements introduced are the reciprocal values.



*Reference Arthur B.Williams & Fred J.Taylor,'Electronics Filter Design Handbook'

Now the above circuit can be represented in the form as shown below

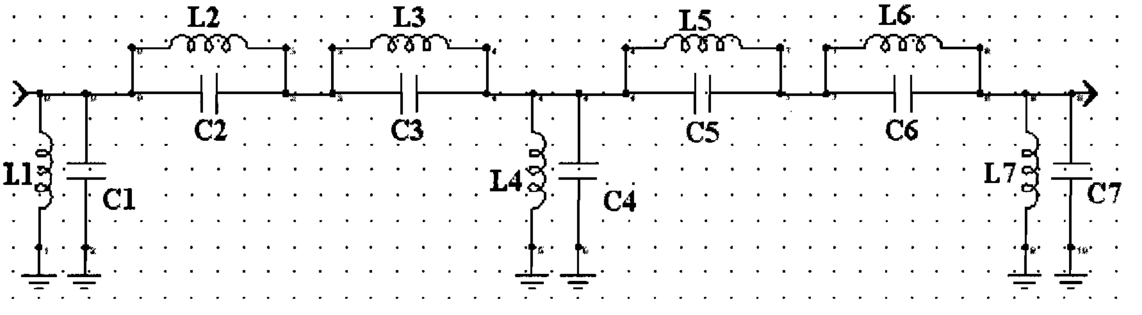


$$\beta = 1 + \frac{1}{2L_1C_1} + \sqrt{\left(\frac{1}{4L_1^2C_1^2} + \frac{1}{L_1C_1}\right)}$$

$$L_a = \frac{1}{C_1(\beta+1)} \;,\; L_b = \beta L_a \;,\; C_a = \frac{1}{L_b} \;,\; C_b = \frac{1}{L_a}$$

After this the BPF is scaled to the required centre frequency and impedance level by multiplying all ind. by $\frac{Z}{FSF}$ & dividing all cap by $Z * FSF$ where $Z = 50$ and $FSF = 2\Pi f = 2\Pi f_0$

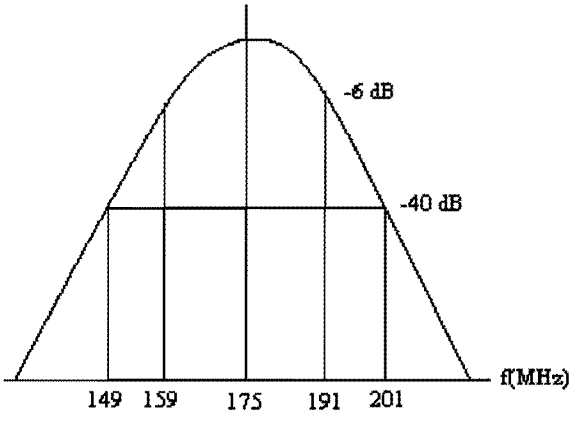
The final circuit is shown below:



This final was simulated using "Eagleware" software and after tuning the component values to meet the required specification the final response was obtained, shown in the next chapter.

5.4 Bandpass Filter specification with centre frequency of 175 MHz

1. Minimum inductor elliptic
2. Cauer-Chebyshev Normal
3. -6dB fall for a b/w of 32MHz
4. -40dB fall for a b/w of 52MHz
5. 50Ω input and output impedance.



5.5 Order Estimation

Geometric Centre Frequency, $f_0 = \sqrt{(f_l f_u)}$

$= \sqrt{(159*191*10^{12})}$

$= 174.26$ MHz

Corresponding geometric freq. for each stop band freq. is calculated from ,$f_1 f_2 = f_0^2$

f1	f2	f2-f1
149	203.8	54.82
151.08	201	49.91

$f_0 = 174.26$ MHz, $BW_{6dB} = 32$ MHz, $BW_{40dB} = 49.91$

Bandpass steepness factor, $A_s = BW_{40dB}/BW_{6dB} = 49.91/32$

$= 1.559$

5.6 Design Procedure

Let us choose ρ of 20% which corresponds to a ripple of 0.18 dB & A_ρ of 13.9dB *
(since $A_\rho = 20\log |1/\rho|$)

$A_{min} + A_\rho = (40 + 13.9)$ dB $= 53.9$ dB

$\therefore \Omega_s* = 1.54$

Hence, order = 5

Hence for the above specification, from table* C0520

$\Omega_s = 1.5243$, $\Theta = 40^0$ $A_{min} = 35.85$

C1 = 0.6580

C2 = 0.1456

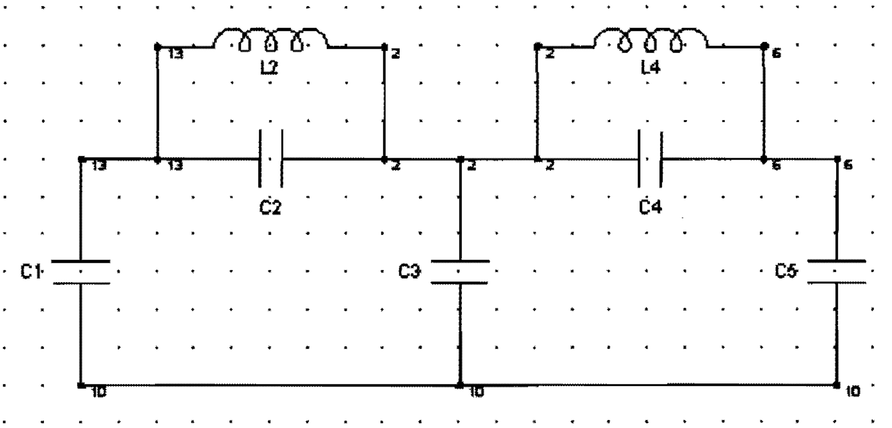
C3 = 1.3282

C4 = 0.459

C5 = 0.4426

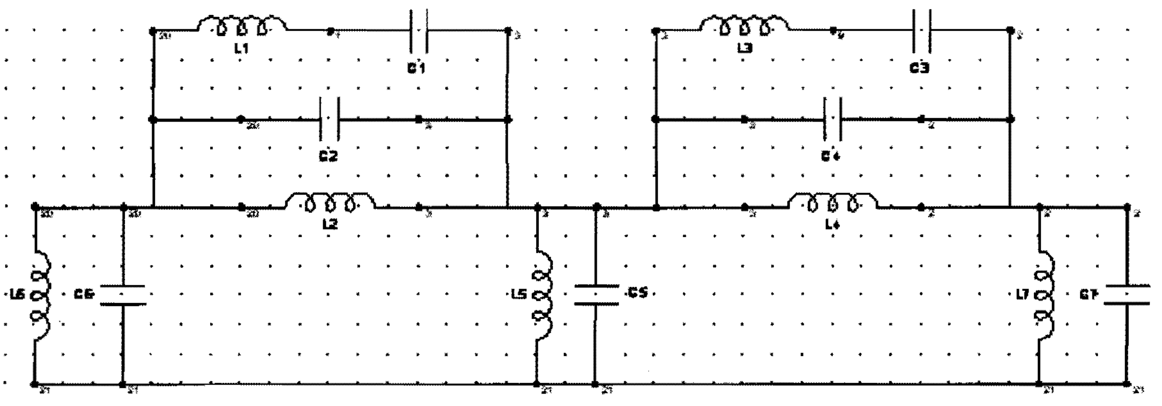
L2 = 1.1556

L4 = 0.8331



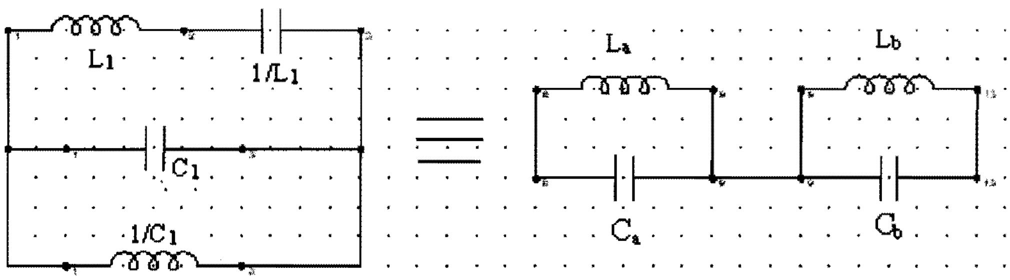
Bandpass $Q_{bp} = f_0/BW = 5.446$

After this we multiply all inductances and capacitances by Q_{bp} , then transform the network into a BPF centered at $\omega_0 = 1$ by resonating each cap. With a parallel inductor & each ind. with a series cap. The resonating elements introduced are the reciprocal values.



*Reference Arthur B.Williams & Fred J.Taylor,'Electronics Filter Design Handbook'

Now the above circuit can be represented in the form as shown below

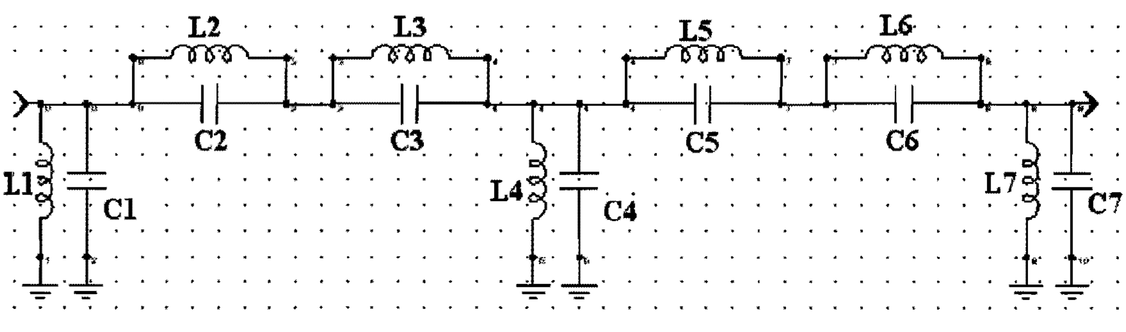


$$\beta = 1 + \frac{1}{2L_1C_1} + \sqrt{\left(\frac{1}{4L_1^2C_1^2} + \frac{1}{L_1C_1}\right)}$$

$$L_a = \frac{1}{C_1(\beta+1)}\ ,\ L_b = \beta L_a\ ,\ C_a = \frac{1}{L_b}\ ,\ C_b = \frac{1}{L_a}$$

After this the BPF is scaled to the required centre frequency and impedance level by multiplying all ind. by $\dfrac{Z}{FSF}$ & dividing all cap by $Z * FSF$ where $Z = 50$ and $FSF = 2\Pi f = 2\Pi f_0$

The final circuit is shown below:

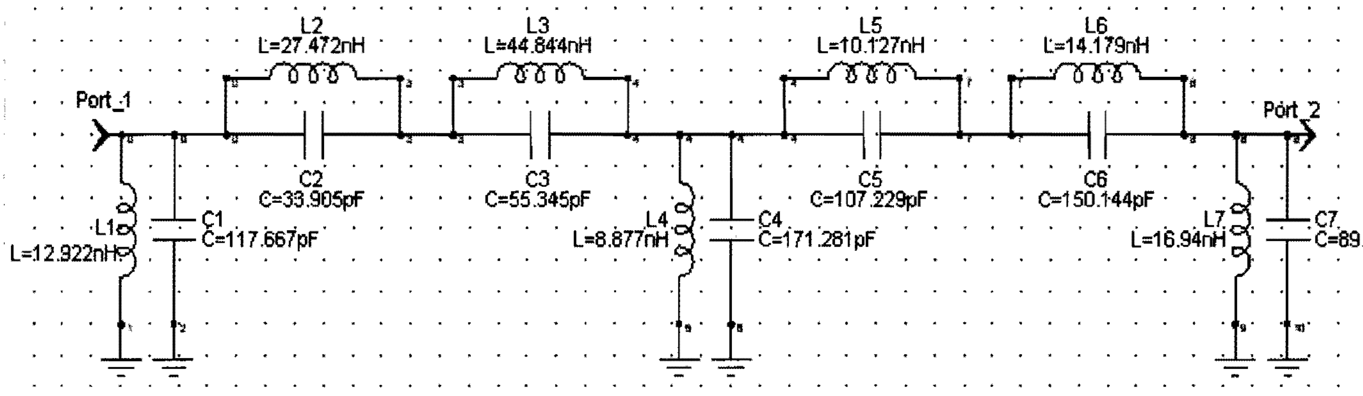


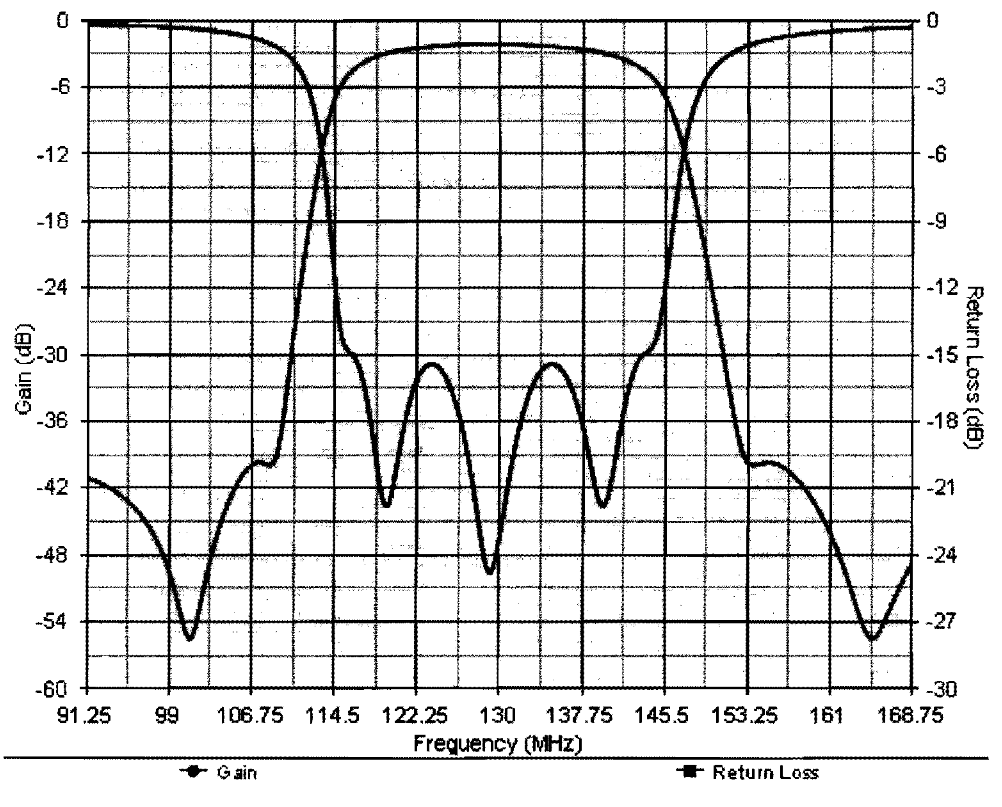
This final was simulated using "Eagleware" software and after tuning the component values to meet the required specification the final response was obtained, shown in the next chapter.

SIMULATION & EXPERIMENTAL RESULTS

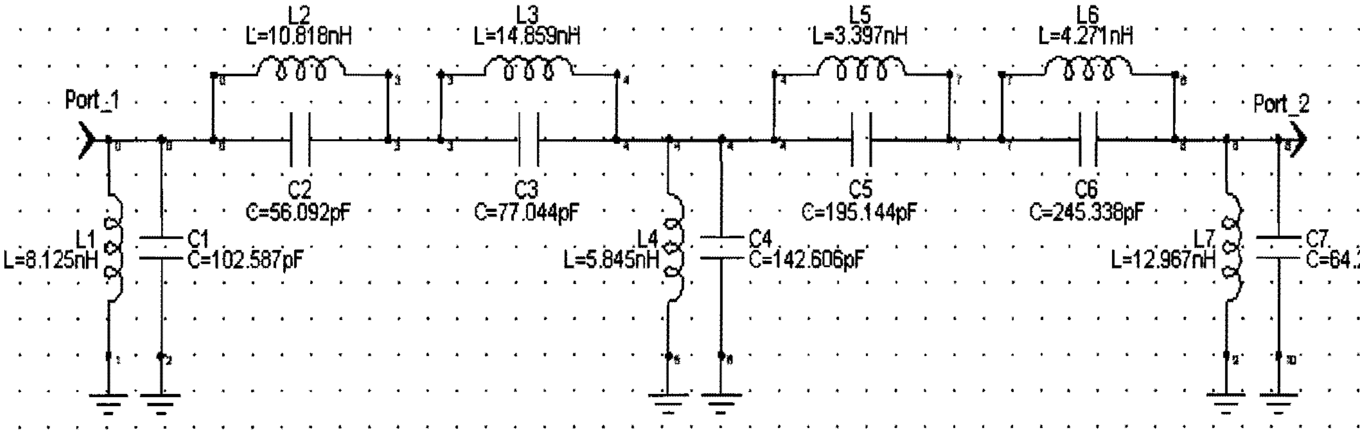
6.1 Filter responses from Genesys Software

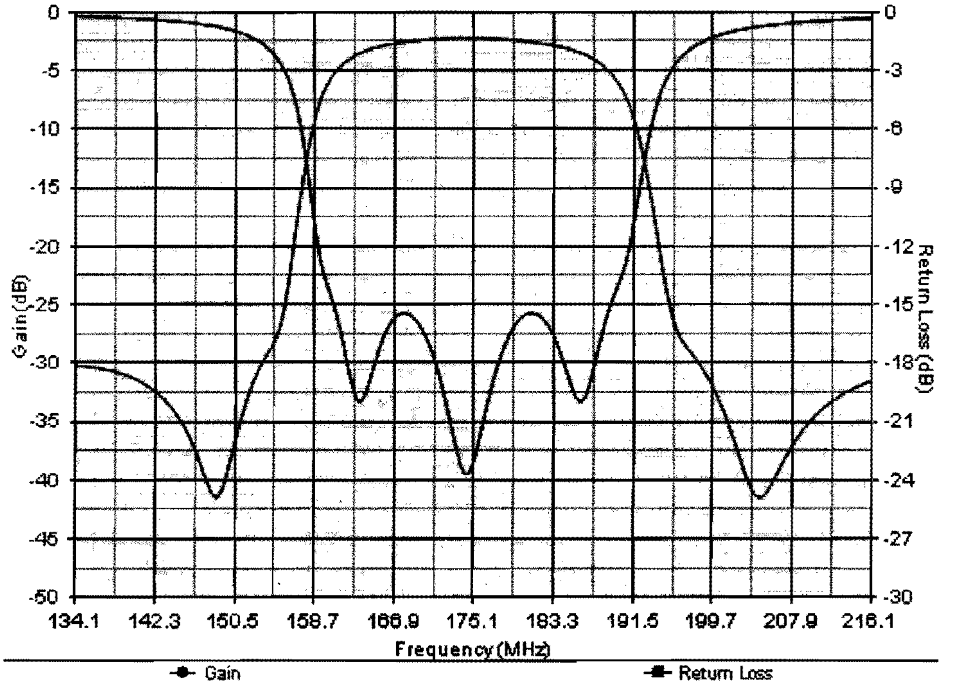
Bandpass filter response with centre frequency of 130 MHz



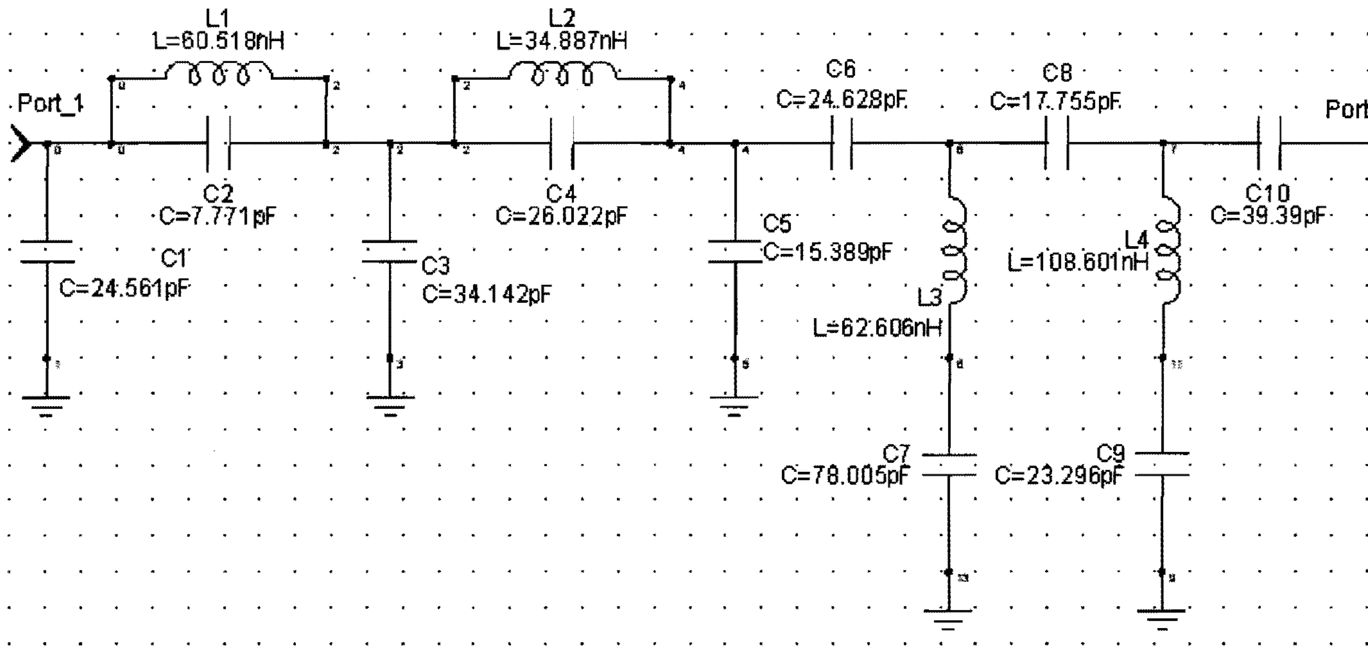


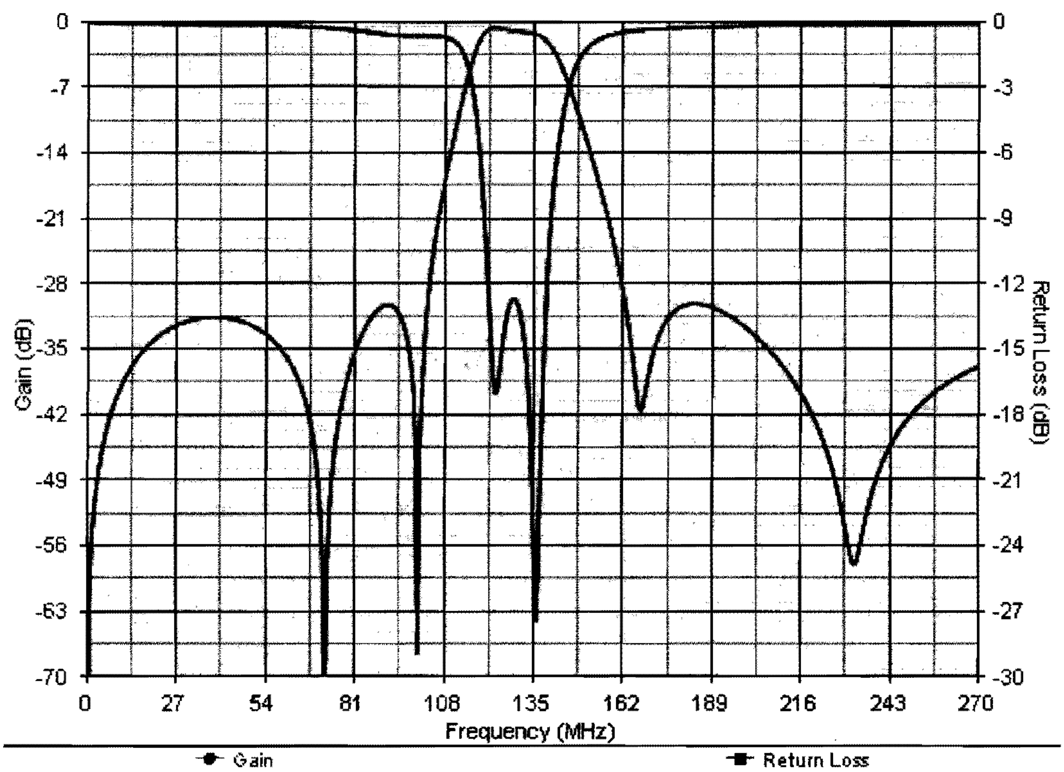
Bandpass filter response with centre frequency of 175MHz



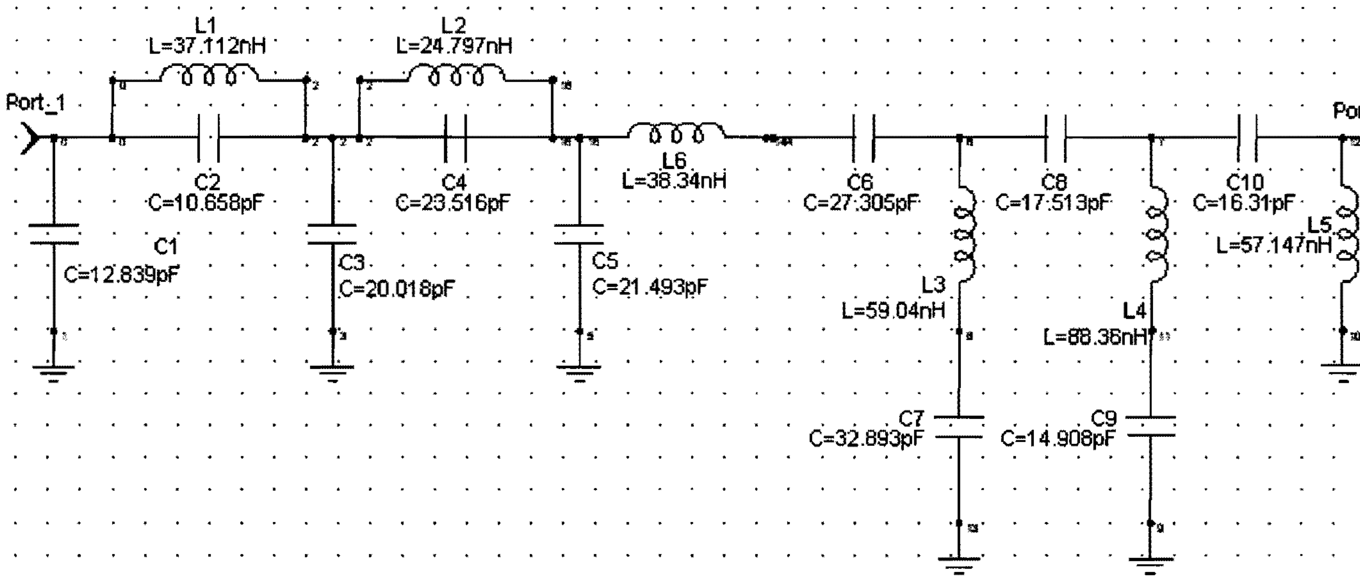


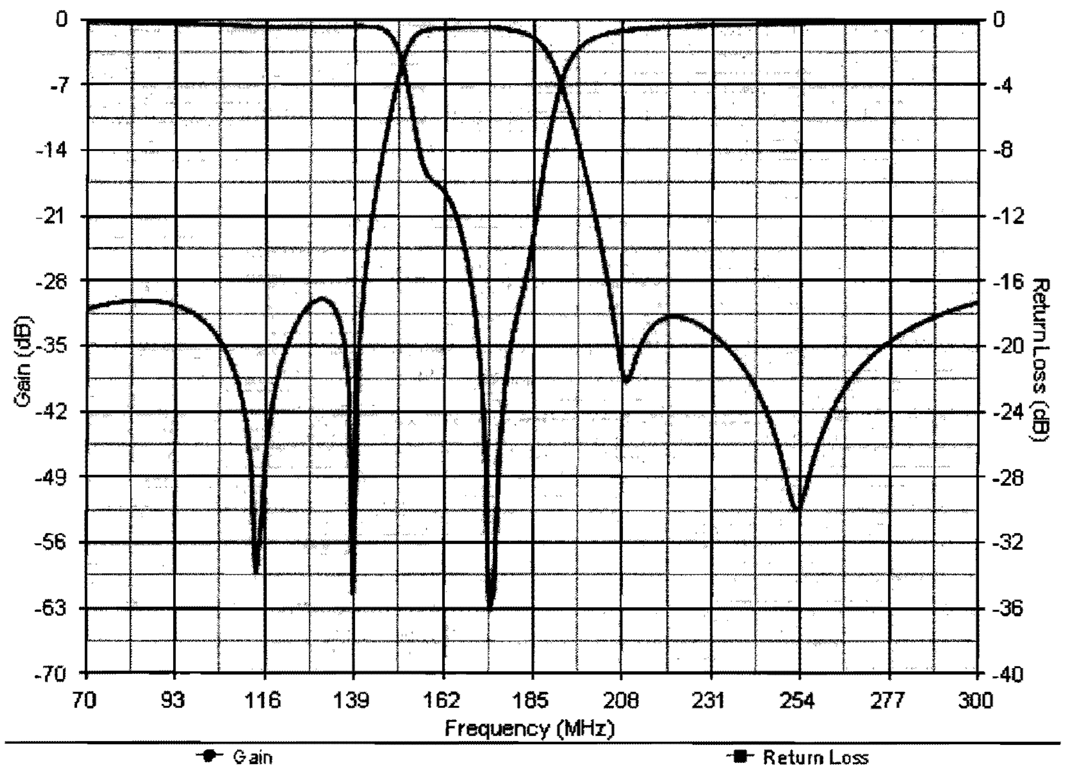
Lowpass-Highpass combination circuit & response with centre frequency of 130 MHz



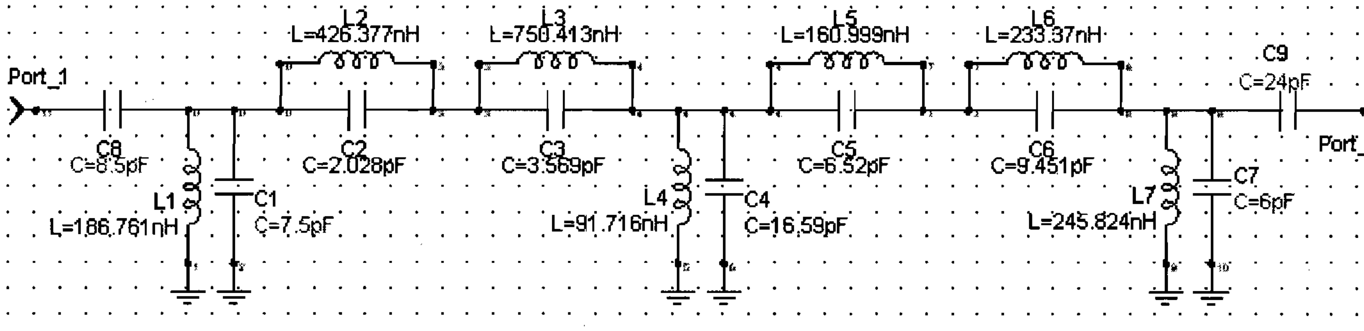


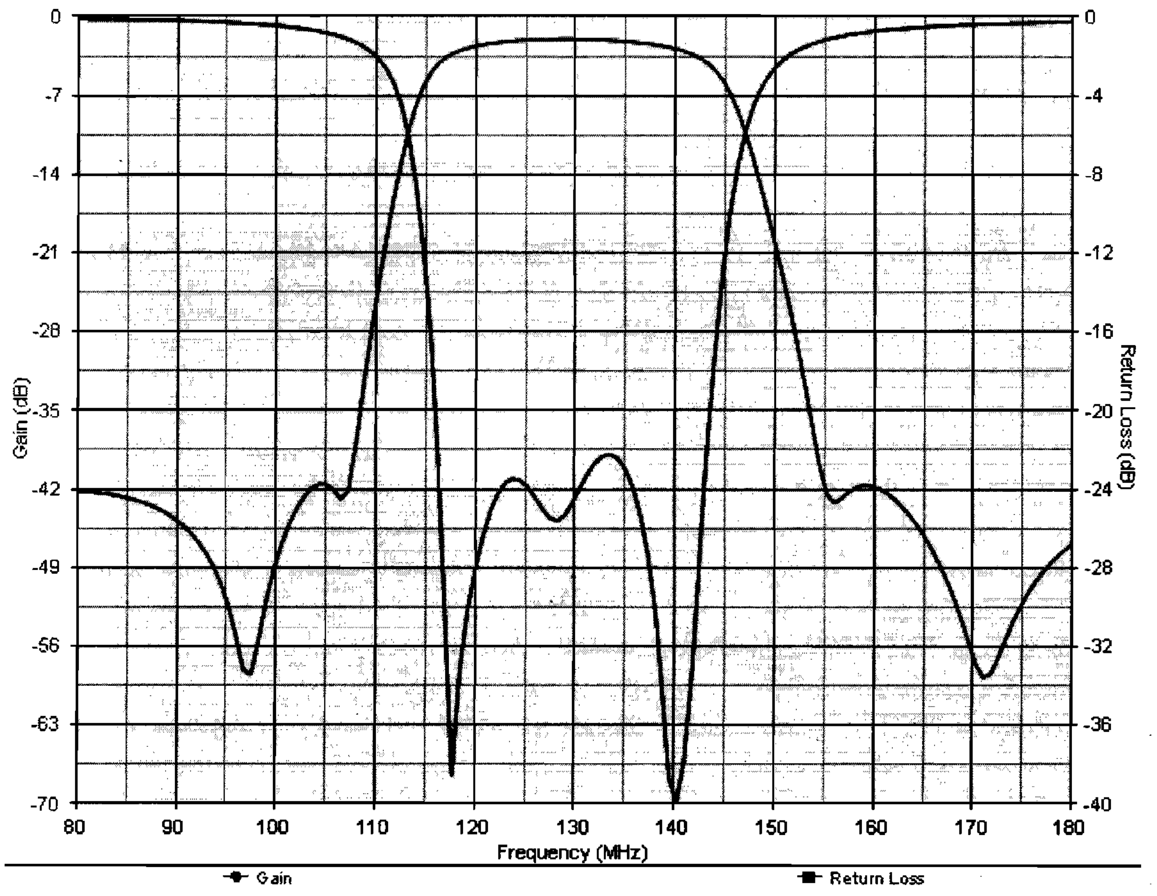
Lowpass-Highpass combination circuit & response with centre frequency of 175MHz





Bandpass filter circuit & response at 130MHz with impedance transformation





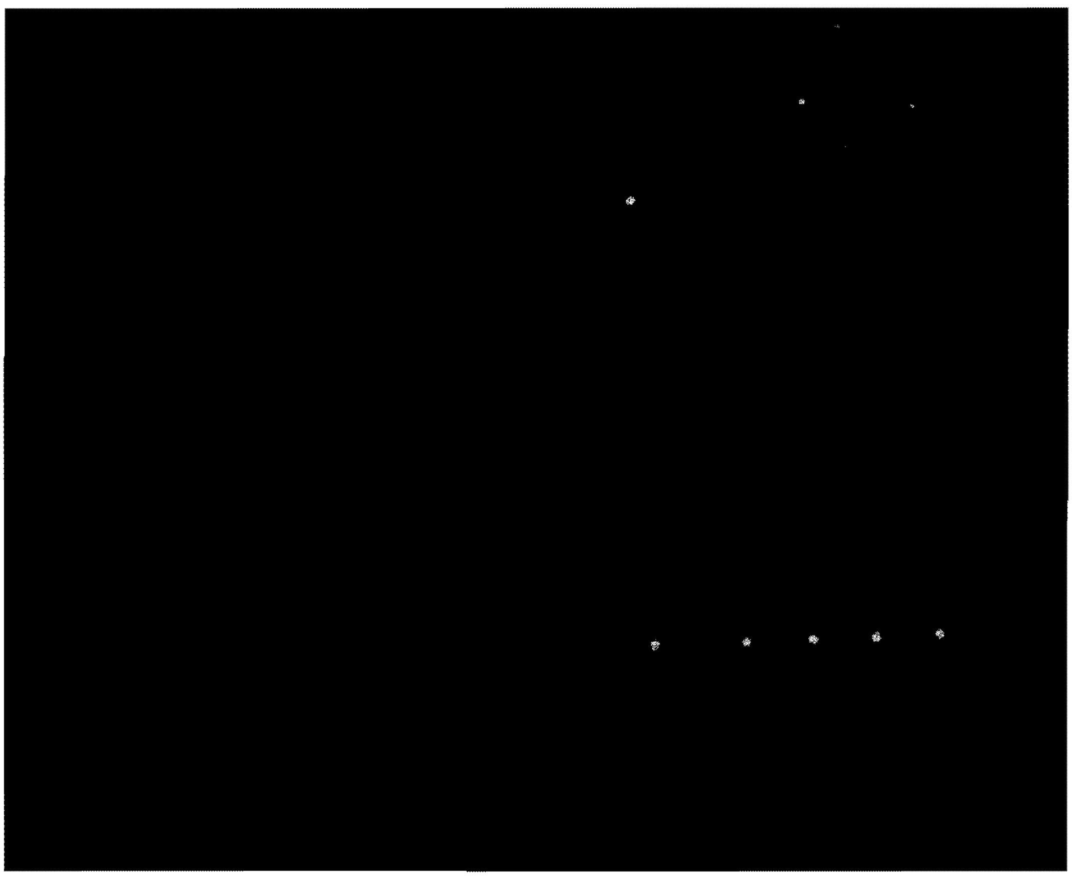
6.2 Practical Difficulties

Although the simulation of the Bandpass filter gave us a response as required, there are some practical difficulties while making it on a circuit board with discrete components.

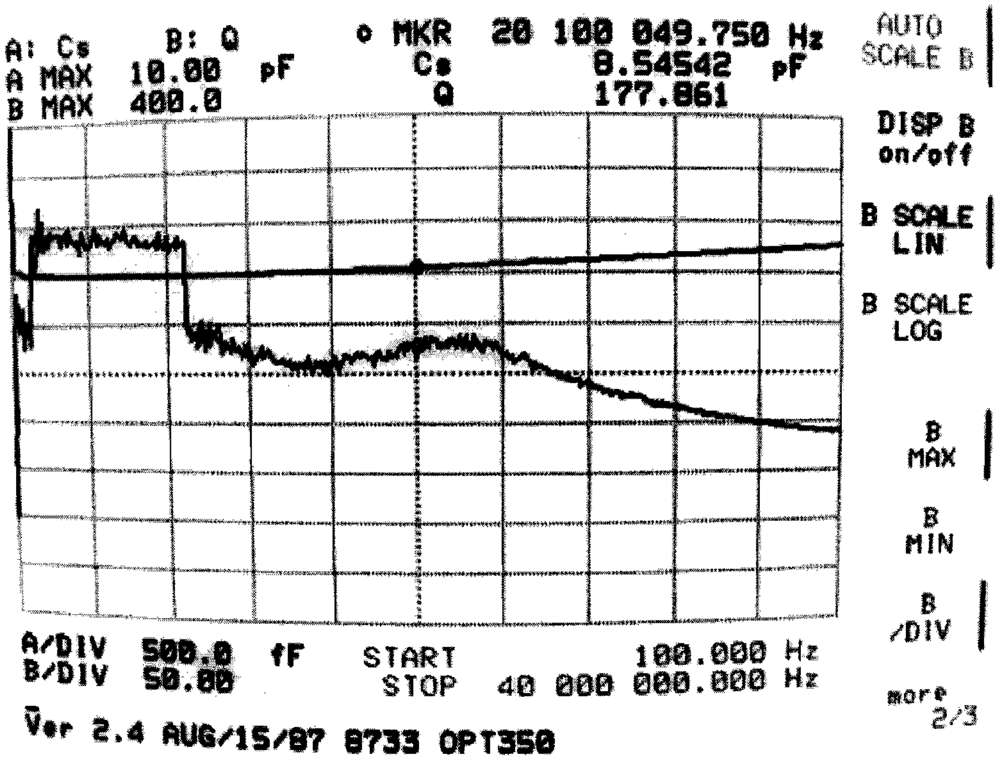
Firstly the BPF requires very small values of inductance going down to the order of 10 nH. Making such small values inductances is not possible with available copper wires. If we increase the ind. values & try to tune the value of the cap. to get the desired response , the cap. values goes down .Next we try to optimize the values by increasing the input & output impedance . So we had to redesign the circuit such that the impedance matches with the probe of 50Ω impedance. This too did not work out and we got merely noise instead of actual response while testing on a network analyzer. Initially there was some problem with the measurement of inductances & capacitance with the impedance analyzer. The Q values came out to be too small. So at last we tried out the design of the bandpass filter with a combination of a high pass & low pass configuration. The circuit was made & tested on a network analyzer & ultimately gave a response as desired.

6.3 Experimental Results

Impedance Analyzer: Instrument for measuring L , C & Q

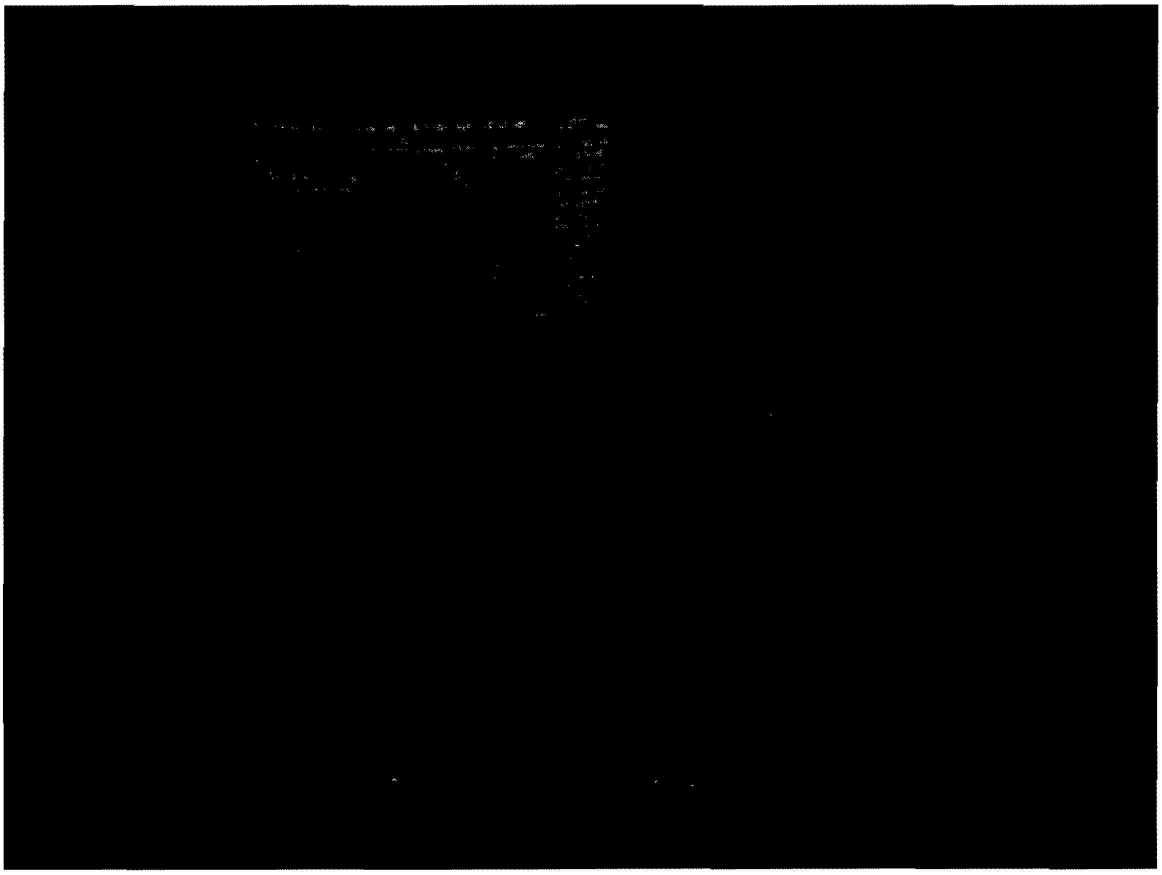


Reading from the Impedance Analyzer

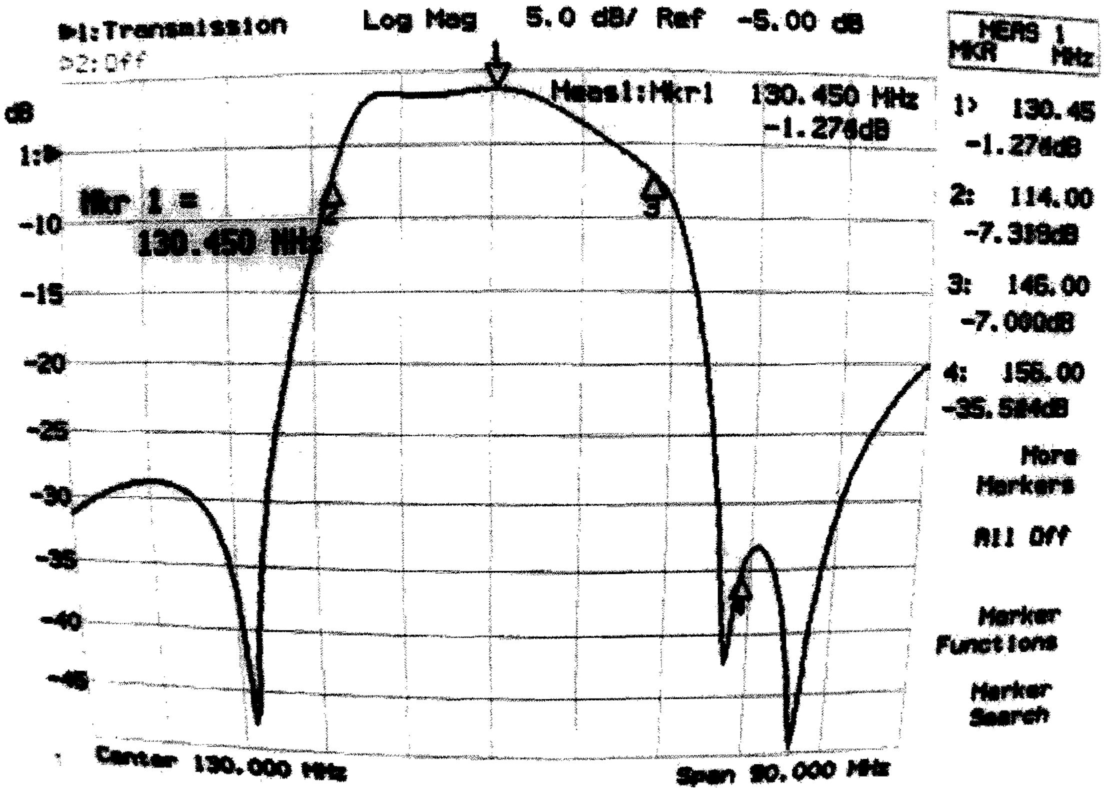


Lowpass-Highpass combination circuit

Experimental verification using a Network Analyzer



Response obtained from the Network Analyzer



CONCLUSION

The final experiment was performed using a low pass high pass combination circuit and the response was achieved as desired. From the response it is seen that a fall of 6 dB was achieved for a bandwidth of 32 MHz. Again almost a fall of 35 dB was obtained 10MHz away from the 32 MHz bandwidth region, which is actually 5dB less than that expected. Nevertheless the fall is large enough so that sampling procedure produces results with negligible distortion.

Now that the BPF with required specification is achieved it can be implemented on the new system that is being designed. Also the filter is designed using lumped components. They can be implemented with micro strip lines since in case of micro strip line low values of capacitances & inductances can be realized easily without any transformation or LP-HP combination.

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