ESTIMATION AND MEASUREMENT OF CROSS-TALK IN THE 32 MHZ BASEBAND SYSTEM

STUDENT PROJECT

By

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June 7, 2013 – June 25, 2013

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NCRA • TIFR

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June 2013

ACKNOWLEDGEMENT

We thank our guide Mr. Ajith Kumar of the digital backend group at GMRT for guiding, monitoring and encouraging us throughout the project. His dedication and thought provoking words were the key behind the smooth and successful completion of this project.

We also thank Mr. Navnath Shinde and Mr. Prakash Hande whose guidance and support helped in the completion of the project.

We also take this opportunity to express our thanks and appreciation to the GMRT Digital Backend team for their cooperation and help. We sincerely thank Mr. Hari and the entire lab for their help during our project work. We would like to thank the analog backend group for providing the hardware and necessary support for carrying out the tests required for this project.

We would also like to thank the staff members of GMRT who were always supportive and friendly.

Thank you.

June 2013

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ABSTRACT

The aim of this project is to measure the leakage signals from one channel to another in an baseband system due to various components involved in the system.

The baseband system includes analog signal processing circuits in the central electronics building of the GMRT, where signals from all the antennas are brought to a common location for further down conversion and bandshaping. We set out to verify the leakage measured practically in the system was in accordance with the calculated values we had obtained. We began the project by studying the block of the baseband system which included two units , the J41 unit and the new IF conversion unit. We then proceeded to make an analysis of the baseband system by theoretically calculating the values of the various figure of merits like gain , noise figure and 1 dB compression point for each unit separately and then both the units as a whole. The theoretical values were then verified practically. Then we analyzed the various possibility for cross talk in the baseband system and set out to calculate it theoretically. These theoretical calculations were then verified practically.

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1. GMRT

The Giant Metrewave Radio Telescope, located near Pune in India, is the world's largest array of radio telescopes at metre wavelengths. It is operated by the National Centre for Radio Astrophysics, a part of the Tata Institute of Fundamental Research, Mumbai.

1.1 About GMRT:

The Giant Metrewave Radio Telescope (GMRT) consists of thirty 45 m diameter antennas spread over a 25 km region. Half the antennas are in a compact, quasi randomly distributed array with a diameter of about 1 km. The remaining antennas are on 3 arms of length of about 14 km (North West, North East and South) with 5 or 6 antennas on each arm. The longest baseline is about 25 km and the shortest is about 100 m without foreshortening.

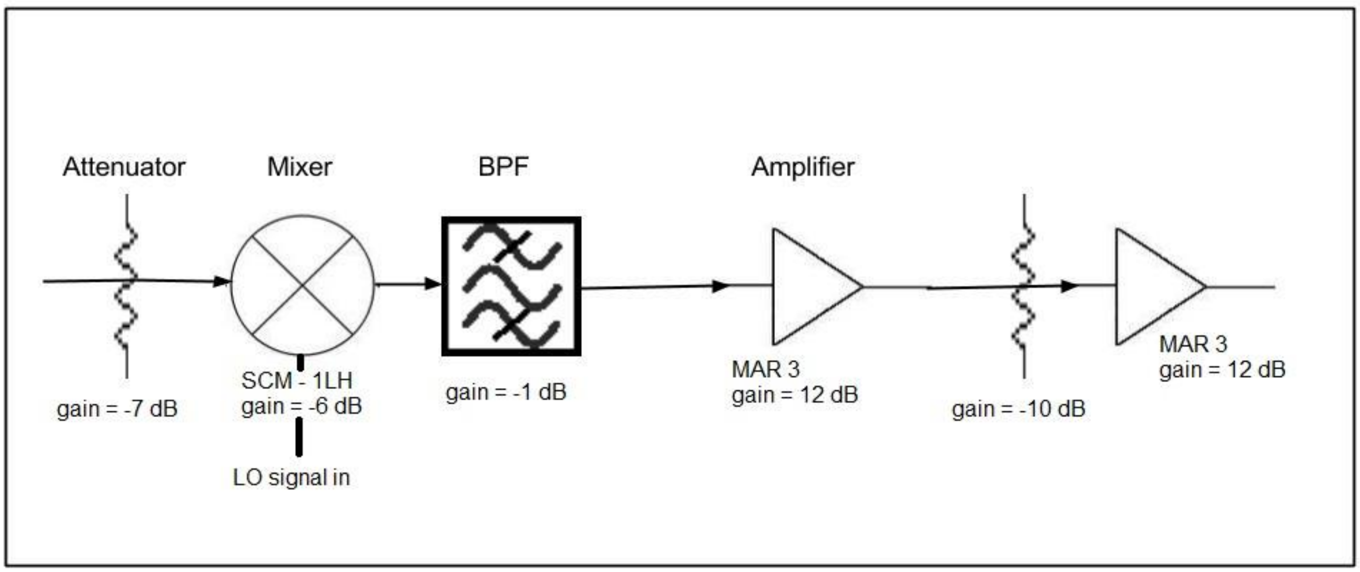
The GMRT contains 30 fully steerable telescopes, each 45 metres in diameter with the reflector made of wire rope stretched between metal struts in a parabolic configuration. This configuration works fine as the telescope operates at long wavelengths (21 cm and above). Every antenna has four different receivers mounted at the focus. Each individual receiver assembly can rotate, enabling the user to select any of them for the observation. GMRT antennas operate in five frequency bands centred at 153, 233, 327, 610, and 1420 Mhz.

Out of the 30 telescopes at GMRT, 14 telescopes are randomly arranged in the central square of 1 km by 1 km in size. Rest 16 telescopes are arranged in three arms of a nearly “Y"-shaped array each having a length of 14 km from the array centre. Therefore GMRT can act as an interferometer which uses a technique known as aperture synthesis to make images of radio sources. The maximum baseline in the array gives the telescope an angular resolution (the smallest angular scale that can be distinguished) of about 1 arc-second, at the frequency of neutral hydrogen.

2.BASEBAND SYSTEM

The baseband system contains two units, the J41 unit and the new IF conversion unit. Both these units are housed in a metallic box. The J41 and the new IF conversion units together make up the baseband system. The units are available for a channel of 175 MHz as well as 130 MHz. These units play the key role in the downconversion of the RF signal.

2.1 J41 UNIT



2.1.2 Working Principle :

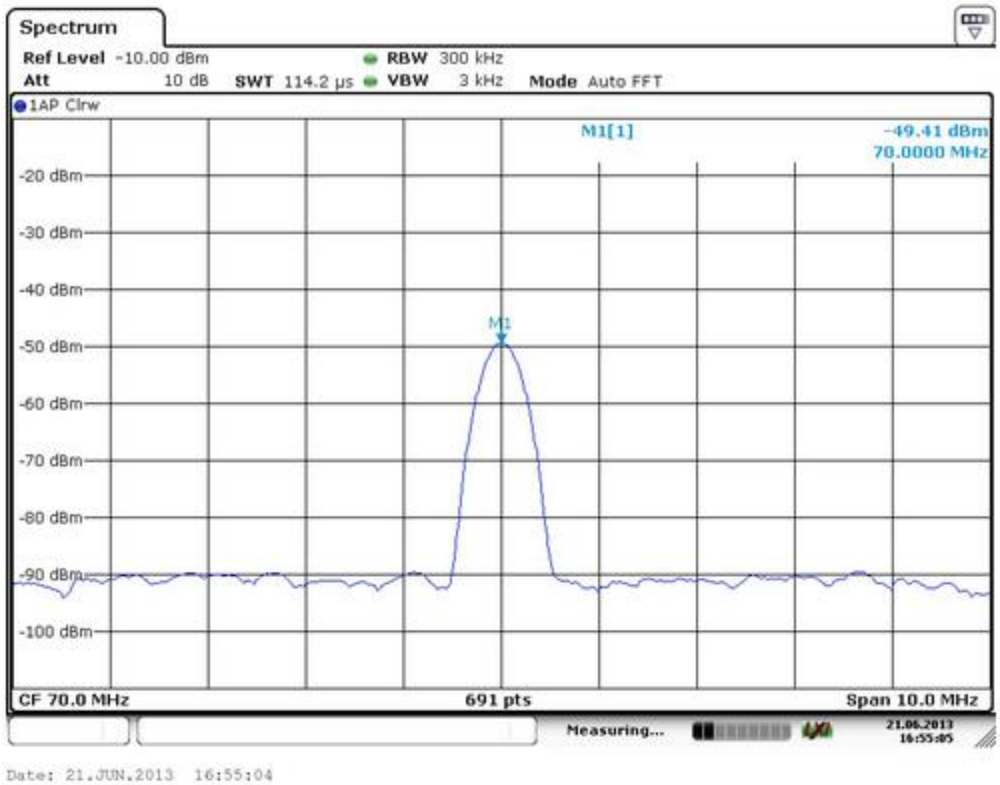
1.There are two inputs to the J41 unit

- RF INPUT(175/130 MHz)
- LO INPUT(105/200 MHz)

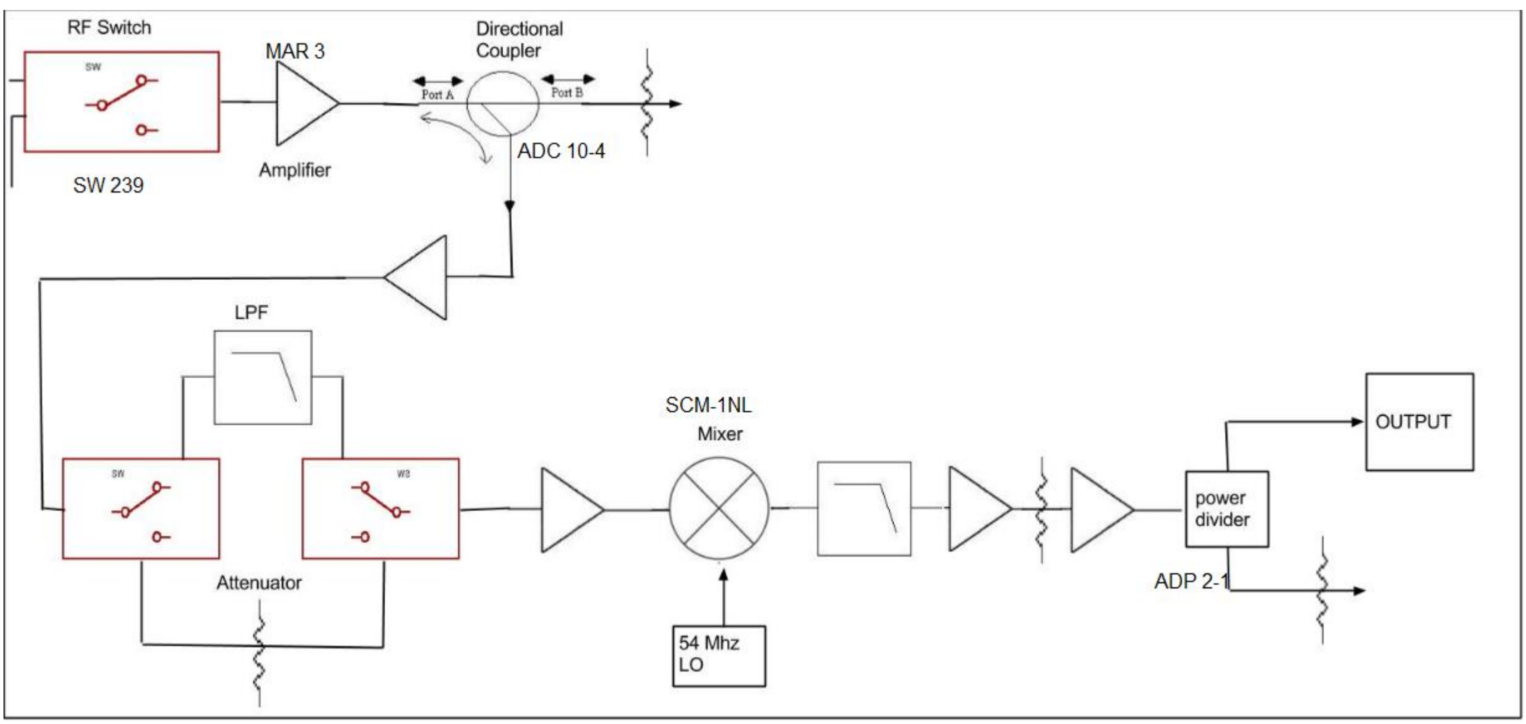
2.The mixer gives the sum and the difference of the two input signals

3.The 100 MHz band pass filter removes the higher frequency component

4.The frequency thus obtained at the output of the J41 unit is 70 MHz



2.2 NEW IF CONVERSION UNIT:



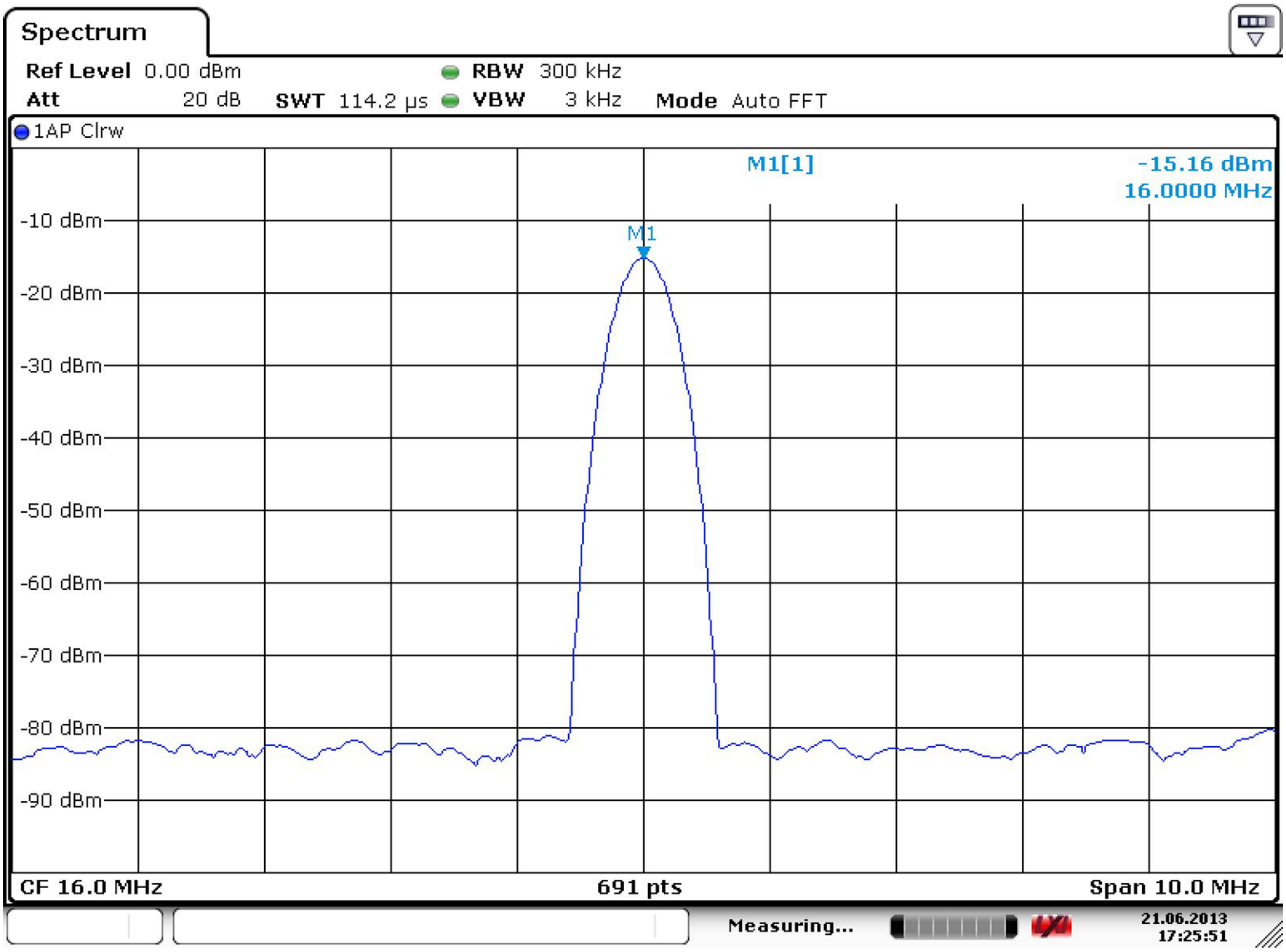
2.2.2 Working principle :

1.The 70 MHz signal from the J41 is given as input to the new IF conversion unit

2.The 70 MHz signal undergoes several stages of amplification and attenuation because of the various components in the unit.

3.An LO frequency of 54 MHz is introduced at the mixer stage which produces the two frequencies 124 MHz and 16 MHz

4.A 100 MHz filter is used and the higher frequencies are rejected and thus 16 MHz is seen at the output.



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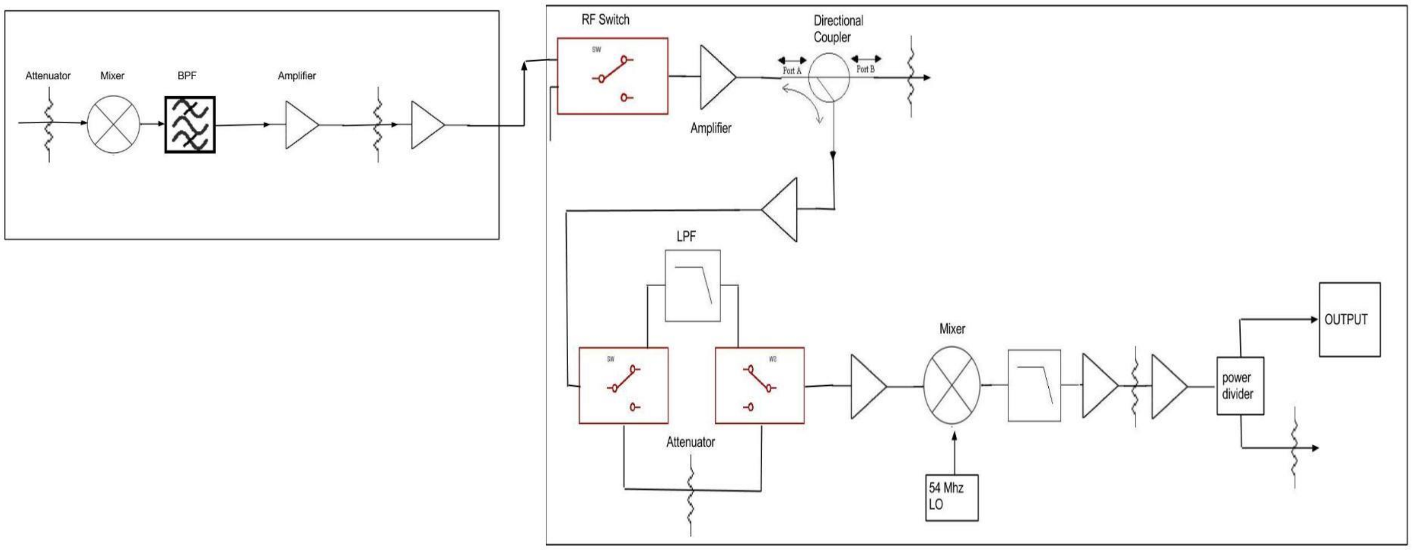
2.3 INTEGRATION OF J41 AND NEW IF CONVERSION UNITS



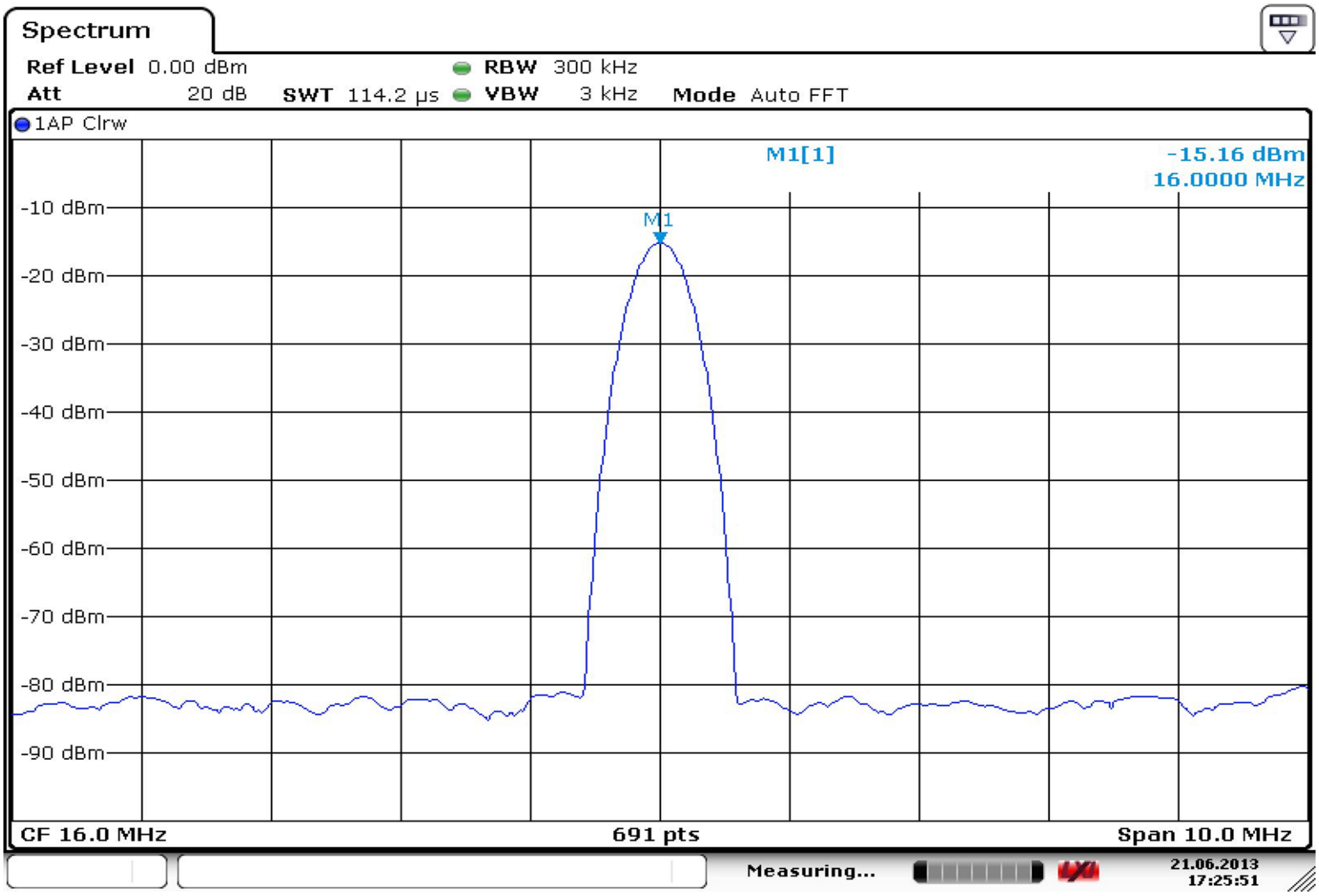
The J41 and New IF conversion basically work on the super heterodyne principle. The RF and LO signals given to the J41 as inputs. The mixer in the J41 converts these two signals ,of frequency *f*rf and *f*lo , into the two new heterodyne frequencies *f*rf + *f*lo and *f*rf - *f*lo. The mixer may inadvertently produce additional frequencies such as third- and higher-order intermodulation products. Ideally, the IF bandpass filter removes all but the desired IF signal at f_{IF}.

The frequency of the local oscillator f_{LO} is set so the desired reception radio frequency f_{RF} mixes to f_{IF}. There are two choices for the local oscillator frequency because the dominant mixer products are at $f_{RF} \pm f_{LO}$. If the local oscillator frequency is less than the desired reception frequency, it is called **low-side injection** ($f_{IF} = f_{RF} - f_{LO}$); if the local oscillator is higher, then it is called **high-side injection** ($f_{IF} = f_{LO} - f_{RF}$).

The IF stage includes a filter and/or multiple tuned circuits in order to achieve the desired selectivity. This filtering must therefore have a band pass equal to or less than the frequency spacing between adjacent broadcast channels. Ideally a filter would have a high attenuation to adjacent channels, but maintain a flat response across the desired signal spectrum in order to retain the quality of the received signal.

Therefore a selected frequency of 70 MHz is filtered out and amplified and passed on to the new IF conversion unit. A central frequency 16 MHz signal in a 32 MHz bandwidth is the final output of the system.

The output expected is a 16 MHz signal when a 175/130 MHz signal is given as RF input.



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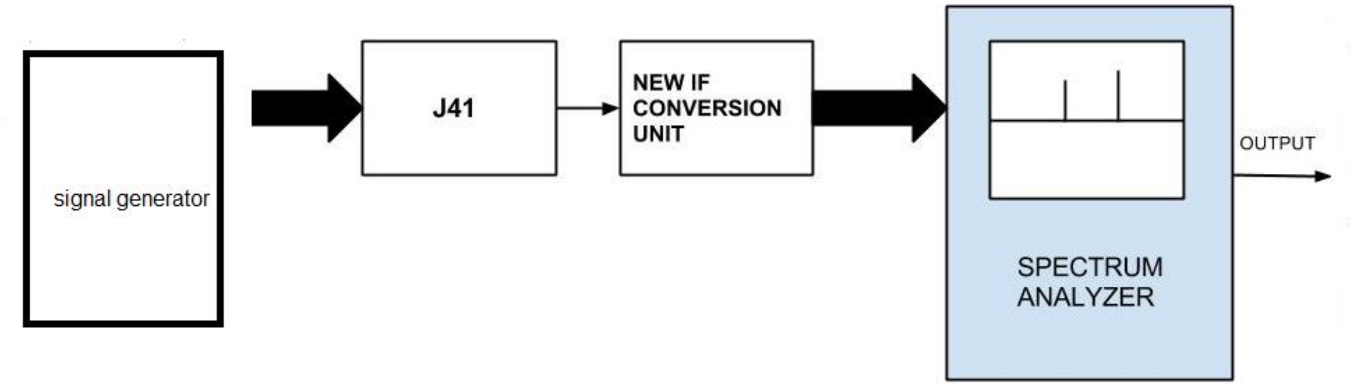
3.FIGURES OF MERIT

In the course of this project we have determined the theoretical values of the figures of merit and have verified it practically. A point to be noted is that the values used in the formula are all linear values and not in dB.

A program in the language C++ was also written to help make the theoretical calculations. The program can be used to determine the values for gain , 1 dB compression point and noise figure for many units in a cascaded configuration. The program takes it input through a file, does the necessary calculations and then writes the output back to a file and displays the output on the screen as well. This program can be modified to include other parameters and helps simplify the processing of calculating the theoretical values.

A **figure of merit** is a quantity used to characterize the performance of a device, system or method, relative to its alternatives. In engineering, figures of merit are often defined for particular materials or devices in order to determine their relative utility for an application.

- **linear value = 10^(dB value/10)**



- **SETUP FOR CALCULATION OF GAIN, 1 dB COMPRESSION POINT AND NOISE FIGURE**

3.1 OVERALL GAIN

Gain : The gain of an amplifier is the ratio of output to input power or amplitude, and is usually measured in decibels

formula: $G = G1 + G2 + G3 + \ldots\ldots\ldots + Gn$

- G: gain of the cascaded system.
- G1: gain of the first component.
- G2: gain of the second component
- Gn: gain of the nth component.

THEORETICAL CALCULATION :

GAIN OF J41 UNIT

Amplifier	Mixer	Band pass filter	Attenuators 1&2
gain +12	Gain -6	Gain -1	Gain -17

(in dB)

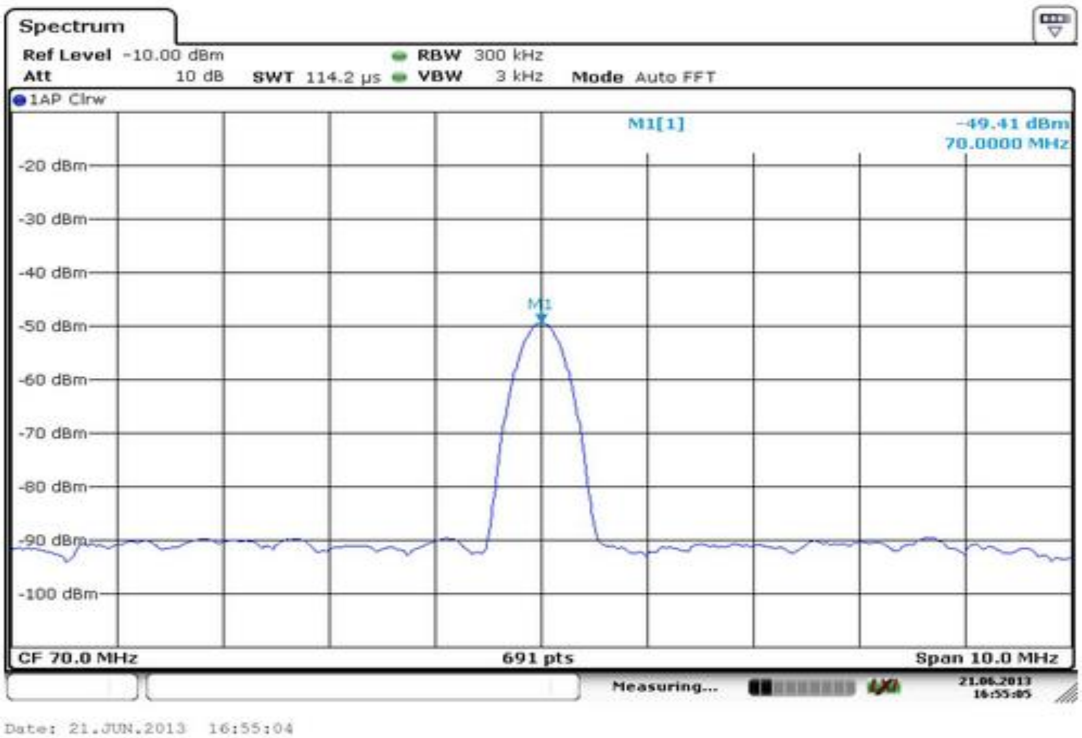
The theoretical calculation involved obtaining the individual gain of each component in the system and applying it in the above formula to obtain the gain for a cascaded system. The gain of the individual components can be obtained from the datasheet of the device used provided the device model number is known. The Gain can be calculated by linearly adding in dB and no conversion to linear values is required.

- The gain of J41 unit is G1= 0 dB
- The gain of new IF conversion unit is calculated as G2= 34 dB
- The overall gain of the baseband system is calculated as G = G1 + G2
- Thus G = 34 dB *(theoretical)*

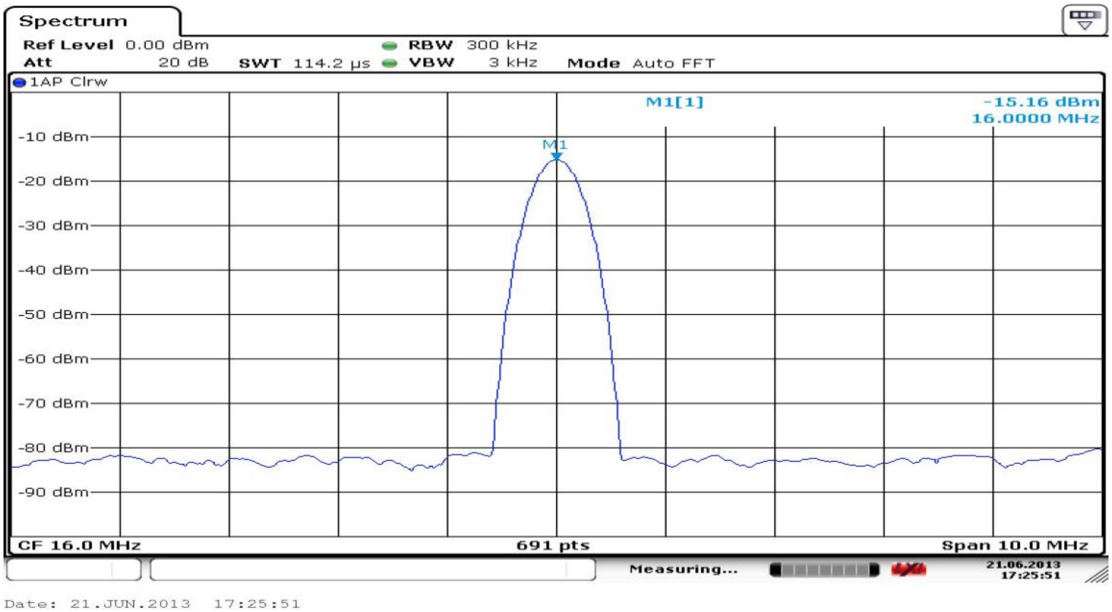
We verified the theoretical value by practically checking the gain of the J41 and the new IF conversion unit and both the systems together.

PRACTICAL MEASUREMENT :

The setup was made as per figure. Then a signal of 175 Mhz,-50dBm and a LO signal of 105 Mhz, 7dBm was given to the J41 unit alone. The output from the J41 was given to a spectrum analyzer. A peak at 70 MHz was observed at a power level of -50dBm.



The same procedure was carried out for the new IF conversion unit but with inputs of 70 MHz, -50 dBm and LO of 54 Mhz,7dBm. Similar peak was observed at 16 MHz and at a power level of 15dBm.



Then the J41 unit and the new IF conversion unit were connected together to model an actual antenna unit and an input of 175 mhz,-50dBm and a LO signal of 105 Mhz,7dBm and another LO signal of 51 Mhz,10dBm were given to the J41 unit and new IF conversion unit respectively.

CONCLUSION **:**

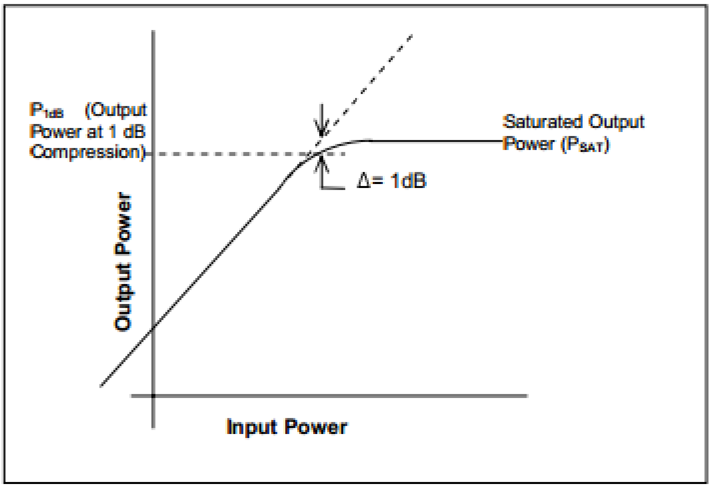
1. Gain of the J41 Unit = 0 dB
2. Gain of the New IF conversion unit = 34 dB
3. Gain of the entire unit = 34 dB
4. practically A peak at 16 Mhz and at a power level of -15 dBm was observed

Therefore the theoretically obtained values were in accordance with the practically measure values.

3.2 1 dB compression Point

1 dB compression point :1 dB compression point defines the output level at which the amplifier's gain is 1 dB less than the small signal gain, or is compressed by 1 dB (P1dB).

At higher input levels, the amplifier goes into saturation and its gain decreases. The 1 dB compression point (P1dB) indicates the power level that causes the gain to drop by 1 dB from its small signal value.



$$p1dB_N = \frac{1}{\frac{1}{p1dB_{N-1} * g_N} + \frac{1}{p1dB_N}} \quad \text{where} \left| \begin{array}{l} N = \text{current component output value} \\ N-1 = \text{previous stage output value} \end{array} \right.$$

$$P1dB_N = 10 * \log(p1dB_N)$$

OUTPUT 1 dB COMPRESSION POINT OF J41 UNIT

Amplifier	**Mixer**	**Band pass filter**	**Attenuators 1&2**
10	100	100	100

OUTPUT 1 dB COMPRESSION POINT OF NEW IF CONVERSION UNIT

RF switch	**Amplifier**	**Directional coupler**	**Low pass filter**	**Mixer**	**Power divider**
20	10	100	100	100	100

(in dB)

1. A point to be noted in calculating the one dB compression point is that only two stages at a time can be applied for while using the formula.
2. Care should also be taken while applying the formula for either the input one dB compression point or the output one dB compression point.

The theoretical value of the 1 dB compression point was calculated by taking two components at a time for the entire system.

J41 unit was calculated to have an output 1dB compression point as 7.87 dB
New IF conversion unit was calculated to have an output 1 dB compression point as 6.13 Db
The overall output 1dB compression point of the basesband system is 6 dB

- Input 1dB comp. point = output 1dB comp. point - gain
  = 6 – 34
  = -28 dB

PRACTICAL MEASUREMENT :

The practical value of 1 dB compression point was obtained by connecting the antenna unit , that is the J41 unit and the new IF conversion unit together, to a signal generator. A suitable LO of 105 MHz was also applied along with an input of 175 MHz, -50dBm. The output of the J41 is given as input to the new IF conversion and a LO signal of 54 Mhz,10dBm was also given. The output of the new IF conversion unit was given to the spectrum analyzer. Then the input power of the signal given was increased slowly until the gain of the system dropped by 1 dB from the actual expected gain. The input power at which the gain drops by 1 dB is known as the input 1dB compression point.

Observation**:**
The following readings were noted to see the gain dropping by 1 dB.

input power (in dBm)	***output power (in dBm)***	***gain (in dBm)***
-50	-13.9	36.1
-41	-5	36
-32	3.6	35.6
-31	4.36	35.36
-30	5	35

CONCLUSION**:**

1. An output 1 dB compression point of **-29 dB** was obtained by the theoretical calculations.
2. The system was found to practically compress at an input power of **-30dBm**.

Therefore the theoretical calculation and practical measurement were equal.

3.3 Noise Figure

Noise Figure (F) : a measure of the signal - to - noise ratio between input and output of the component .

$$F = \frac{S_i / N_i}{S_o / N_o} \geq 1$$

Where, S_i : input signal power S_o : output signal power

N_i : input noise power N_0 : output noise power

Friis's formula is used to calculate the total noise factor of a cascade of stages, each with its own noise factor and gain. The total noise factor can then be used to calculate the total noise figure. The total noise factor is given as :

$$F_{total} = F_1 + \frac{F_2 - 1}{G_1} + \frac{F_3 - 1}{G_1 G_2} + \frac{F_4 - 1}{G_1 G_2 G_3} + ... + \frac{F_n - 1}{G_1 G_2 ... G_{n-1}}$$

where F_n and G_n are the noise factor and available power gain, respectively, of the n-th stage. Note that both magnitudes are expressed as ratios, not in decibels.
An important consequence of this formula is that the overall noise figure of a radio receiver is primarily established by the noise figure of its first amplifying stage. Subsequent stages have a diminishing effect on signal-to-noise ratio. For this reason, the first stage amplifier in a receiver is often called the low-noise amplifier (LNA).

The theoretical value was calculated with help of the formula. The noise figure and gain of each component was determined and substituted in the above formula to obtain the theoretical value of the noise figure.
Then from the noise figure the respective noise temperature was calculated.

- noise temperature, $\boldsymbol{T = 290 * \{10^{\frac{dB}{10}} - 1\}}$

From the above equation the corresponding noise temperature was calculated. Subsequently the noise power was calculated from the noise temperature with the help of the formula ,

- noise power, **N=KTB** where,

- K is the Boltzmann constant = 1.38 x 10-23 J/K
- T is absolute temperature
- B is the 3dB noise bandwidth

PRACTICAL MEASUREMENT :

The practical measurement involved in terminating the RF input to a J41 unit and measuring the value of the noise floor with the help of an spectrum analyzer.
The practical and theoretical values were noted down and tabulated.

CONCLUSION**:**

Theoretical value of noise figure for the J41 unit	18.64 dB
Theoretical value of noise figure for the new IF conversion unit	3.90 dB
Theoretical value of noise figure for J41 unit and new IF conversion unit combined	18.72 dB
Noise Temperature	21607.2272 K
Noise Power	-119.22 dB
Practically measured value of the noise floor using a spectrum analyzer	-110 dBm

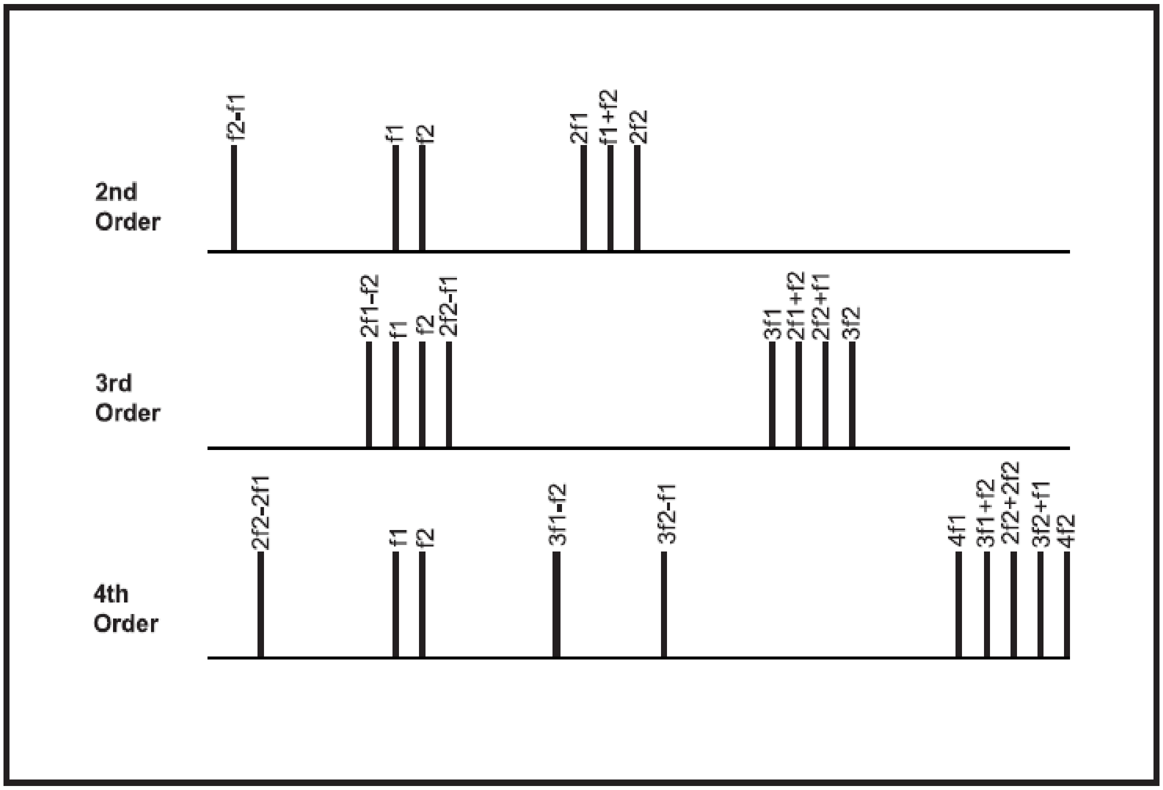
4 INTERMODULATION DISTORTION

Intermodulation distortion is the result of two or more signals interacting in a non linear device to produce additional unwanted signals. These additional signals (intermodulation products) occur mainly in devices such as amplifiers and mixers, but to a lesser extent they also occur in passive devices such as those found in many transmission systems.

Two interacting signals will produce intermodulation products at the sum and difference of integer multiples of the original frequency. For two input signals, the output frequency components can be expressed as :

mf1+nf2 or **mf1-nf2** ,where m and n are integers.

The order of the intermodulation product is the sum of the integers m+n. The ‘two tone’ third order components, (2*f1-f2 and2*f2-f1) are particularly important because unlike 2nd order distortion, i.e. harmonic distortion at 2*f1 or 2*f2, they can occur at frequencies close to the desired/interfering signals and so cannot be easily filtered. Higher order intermodulation products are generally less important because they have lower amplitudes and are more widely spaced. The remaining third order products, 2f1+f2 and 2f2+f1, do not generally present a problem. The distribution of harmonics and third order products are shown in figure.



Example.

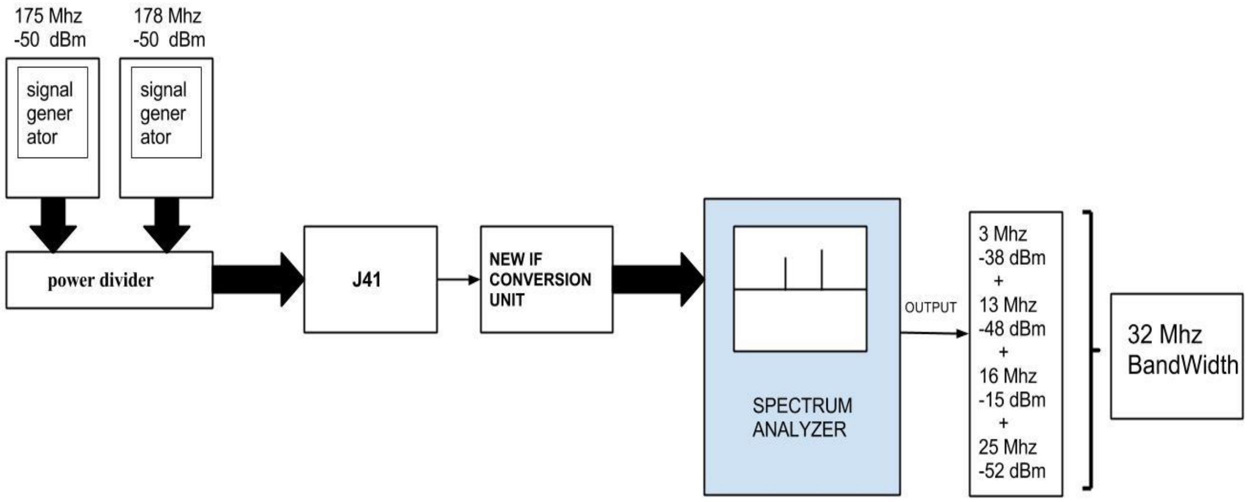
If two signals, f1 and f2, at 90 MHz and 95 MHz respectively are applied to an amplifier, any non linearity of the device will result in :

- Second order intermodulation products at 5 MHz (95 – 90) and 185 MHz (95 + 90)
- two tone third order intermodulation products at 85 MHz ((2 x 90) - 95)) and 100 MHz ((2 x 95) - 90),
- two further third order intermodulation products at 275 MHz ((2 x 90) + 95)) and 280 MHz ((2 x 95) + 90),
- 2nd order harmonics at180 MHz (2 x 90) and 190 MHz (2 x 95)
- additional 3rd order harmonics at 270 MHz and 285 MHz.
- Higher order intermodulation products are also produced in the similar manner.

The magnitude of intermodulation products cannot be predicted easily but it is known that their amplitude diminishes with order.

4.1 MEASUREMENT OF INTERMODULATION INDEX

The set-up for the measurement of intermodulation index is as shown in the figure below :



PROCEDURE:

The two signal generators were set at 175 MHz, -50dBm and 178 MHz, -50dBm respectively.
(Two frequencies with minor differences in magnitude were selected so as to enable the observation of intermodulation products within the 32 MHz bandwidth.)

The J41 unit and the New if conversion unit is connected in series thus having a total gain of around 35dBm

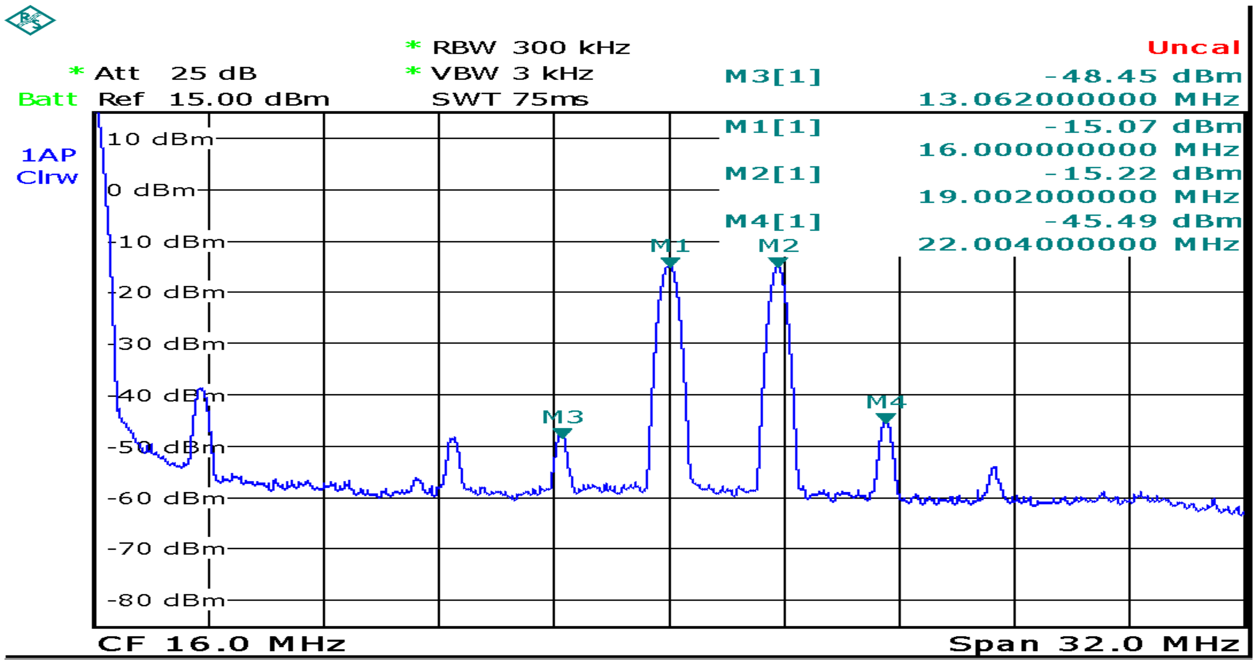
OBSERVATION:

1) Two central frequencies were obtained at 16 MHz and 19 MHz with magnitudes of -15dBm
2) Five other peaks apart from the central frequencies were also observed in the 32 MHz bandwidth.

2.1)13 MHz((2 x 16) – 19) and 22 MHz((2 x 19) – 16) are the third order intermodulation products.

2.2) 3 MHz(19 – 16) is the second order intermodulation product
(the other second order intermodulation product is 35 MHz(19 + 16) and hence could not be observed in the 32 MHz bandwidth.)

2.3) 25 MHz((3 x 19) – (2 x 16)) is the fifth order intermodulation product
(the other component of the fifth order intermodulation product also falls outside the 32 MHz bandwidth)



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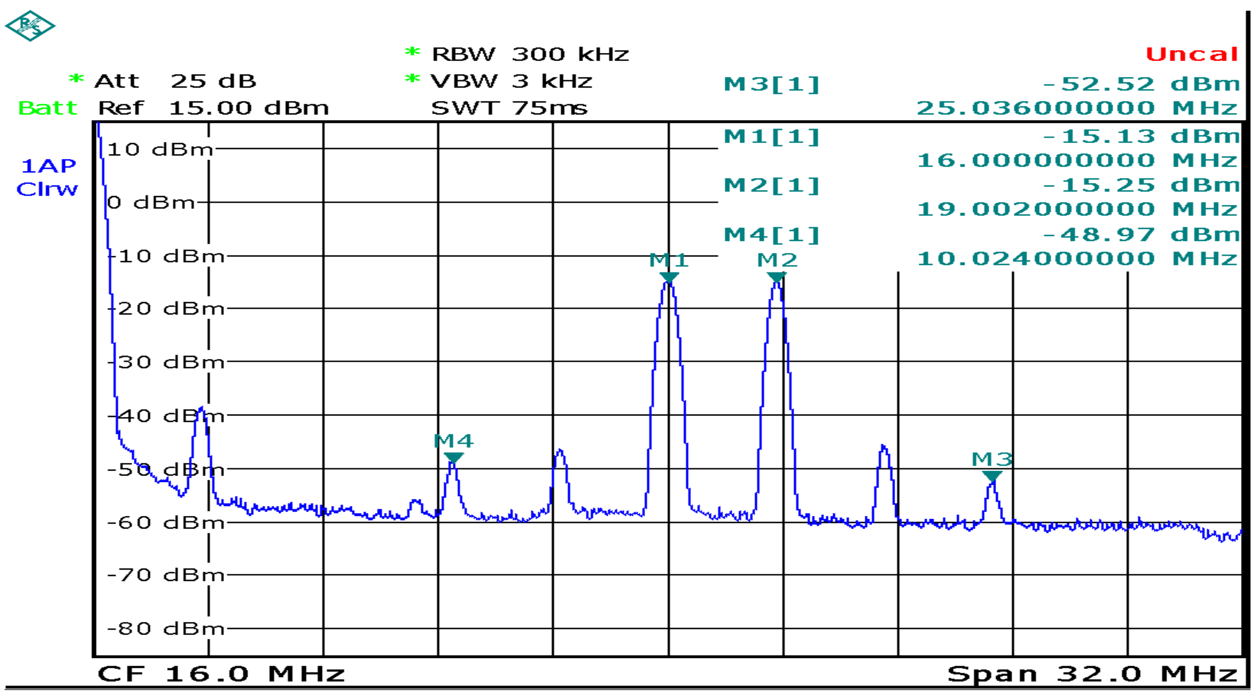

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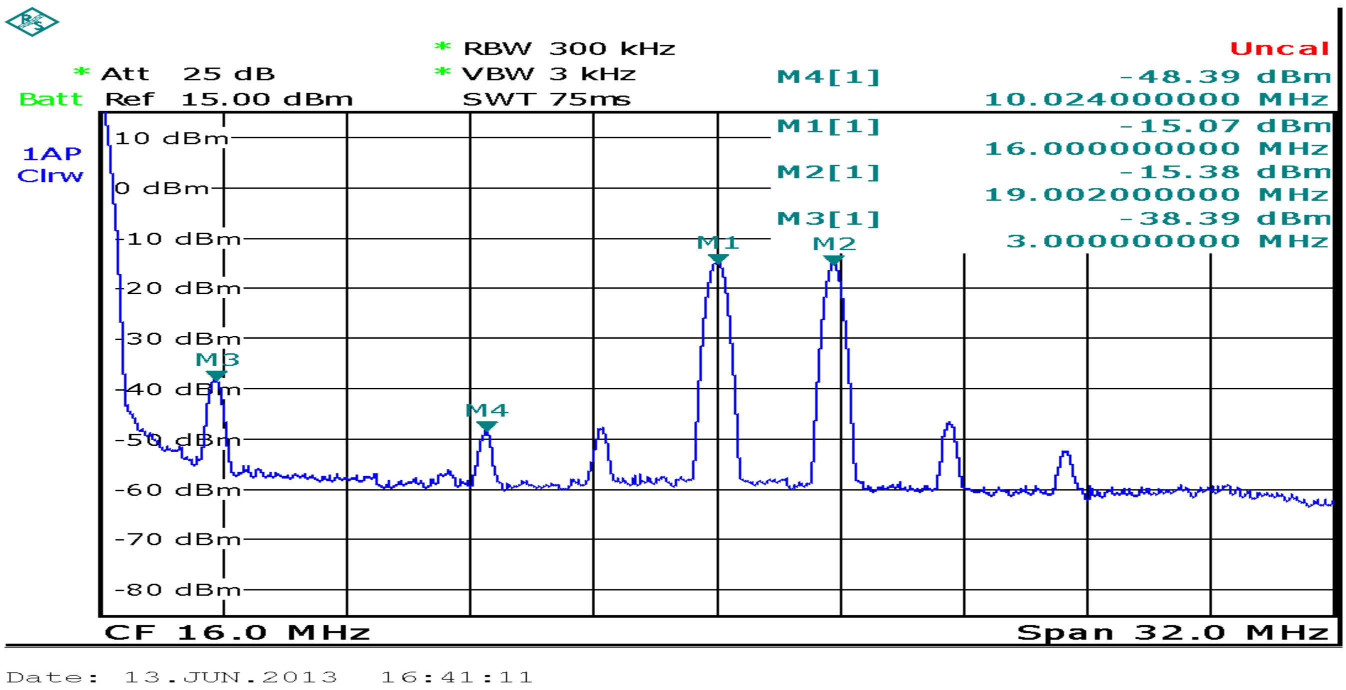


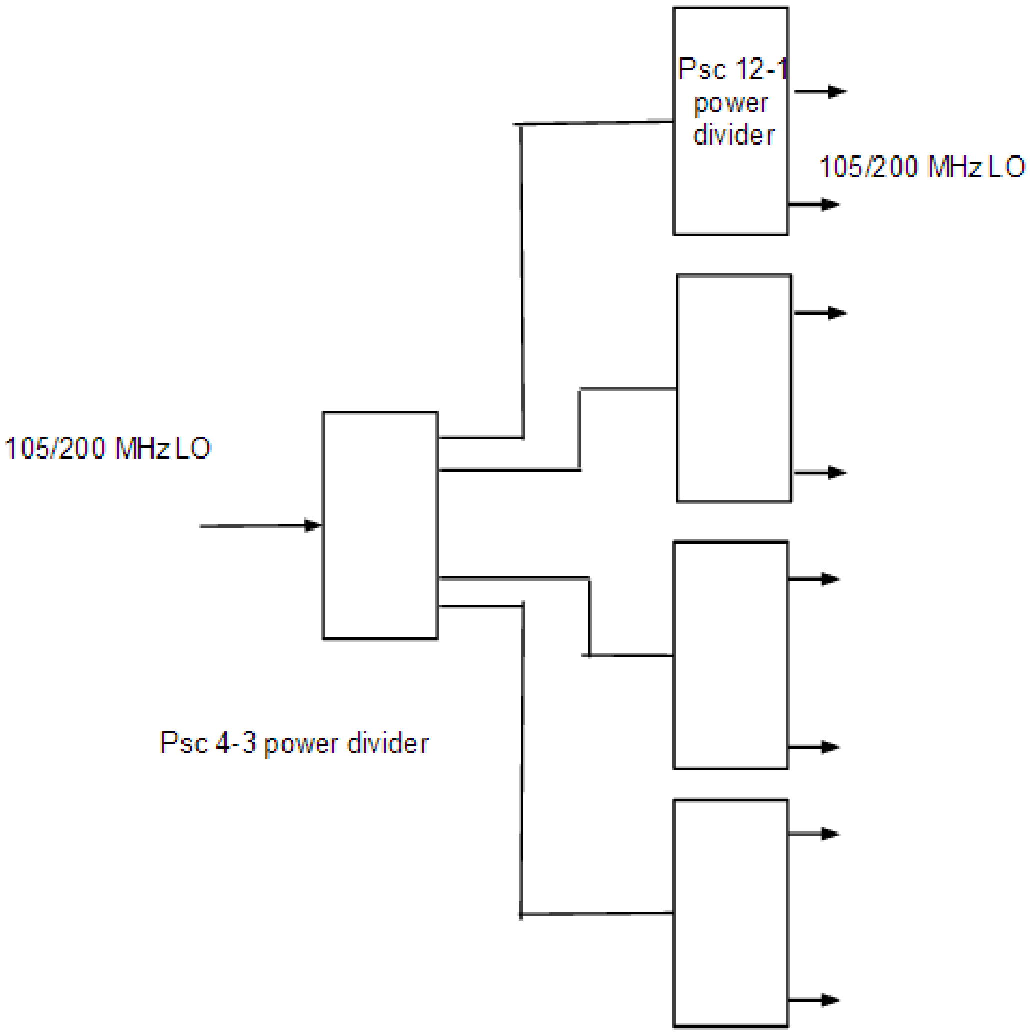
5.CROSS TALK

- What is cross talk ?
- In electronics, crosstalk (XT) is any phenomenon by which a signal transmitted on one circuit or channel of a transmission system creates an undesired effect in another circuit or channel.

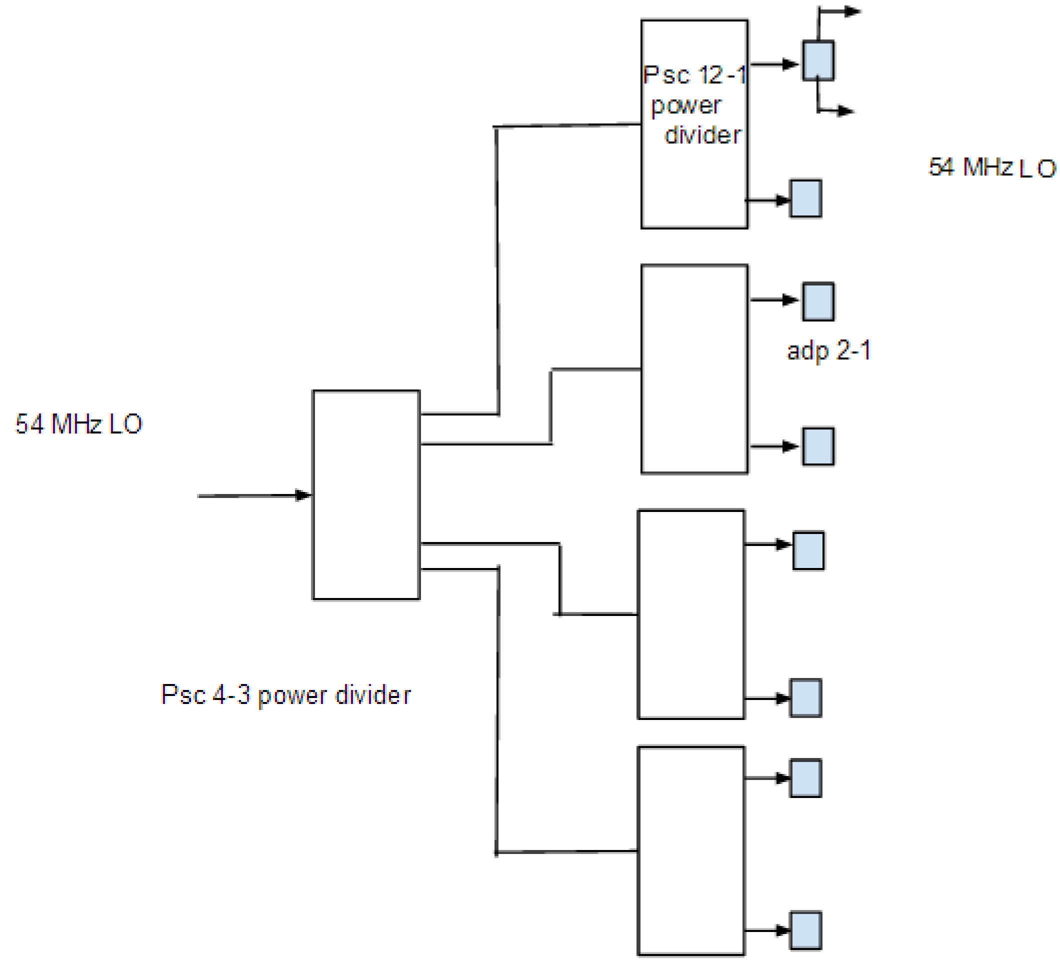
No Transmitter - Receiver system is perfect, Hence there will be a leakage of certain signals at certain frequencies from one channel to another or even from one unit to another. These types of signal leakages could have been due to the power dividers used , the various components used or even could also be attributed to the body of the antenna system. In this type of antenna system with which we worked with , the main area of focus was the leakage of the the signals due to the power dividers used to distribute the LO signals.

5.1 LO DISTRIBUTION

The LO signals of either 105 MHz or 200 MHz required for each unit was tapped from a single source which first went through a 4 way power divider. Therefore, one input signal of LO gave 4 output signals. Then each output of the 4 way divider became an input for the four 12 way power dividers. Each output of each 12 way divider was connected to an antenna system of J41 and new IF conversion system



For LO signal of 54 MHz , The output of each 12 way divider goes through another 2 way power divider before being applied to the mixer of the new IF conversion unit.



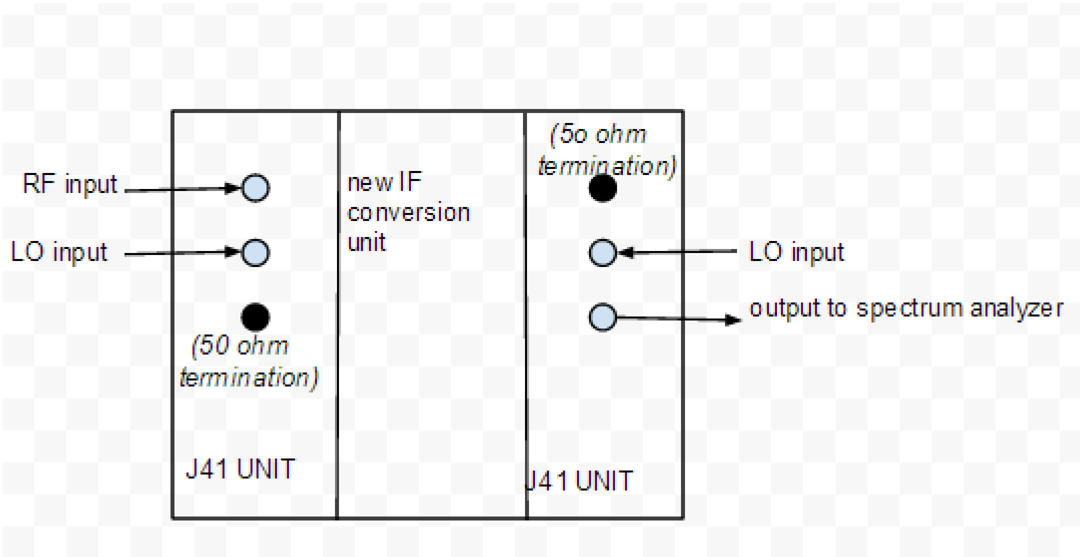
We assumed the main leakage of the signals through the LO distributors , that is through the power dividers. A signal of very small power can creep through the 12 way power divider from the antenna units and go into any of the 11 remaining ports thereby introducing this leakage in another unit. The theoretical calculations involved in assuming the worst case scenarios.

The mixers, which mix the RF and LO signals , have an isolation losses between each port which creates new signals of very low power levels which are introduced into other units via the power dividers.The isolation for the mixers were obtained from the datasheet of the mixers used in each unit. The isolation between each port for the 12 way power dividers was also obtained from the datasheet of the power dividers used in the antenna racks. The isolation between each port of the mixer or the power divider depends on the types of the mixers or power dividers used and frequency of the signal passing through it.
The Theoretical calculation involved retrieving the data from the datasheet and analyzing the path through which the leakage or the isolation was maximum, as only the worst case scenarios were considered. The leakage through the 12 way power divider and the 2 way power dividers were

calculated for LO signals of 105 Mhz , 200 Mhz and 54 Mhz. These calculated values were then verified against the practically determined values.

5.2 PRACTICAL MEASUREMENT

5.2.2 J41-J41 LEAKAGE



THEORETICAL CALCULATION :

- FOR 105 MHz LO signal.

Power input to RF port : -50 dBm
LO-RF isolation : 51.31 dB
RF leakage to LO port : (-50-51.37) = -101.31 dBm
12 way power divider isolation(for 105 MHz LO) =29.57dB
Power leakage to adjacent J41 unit(for 105 MHz LO) = (-101.31 – 29.57) = -130.88

- For 200 MHz LO

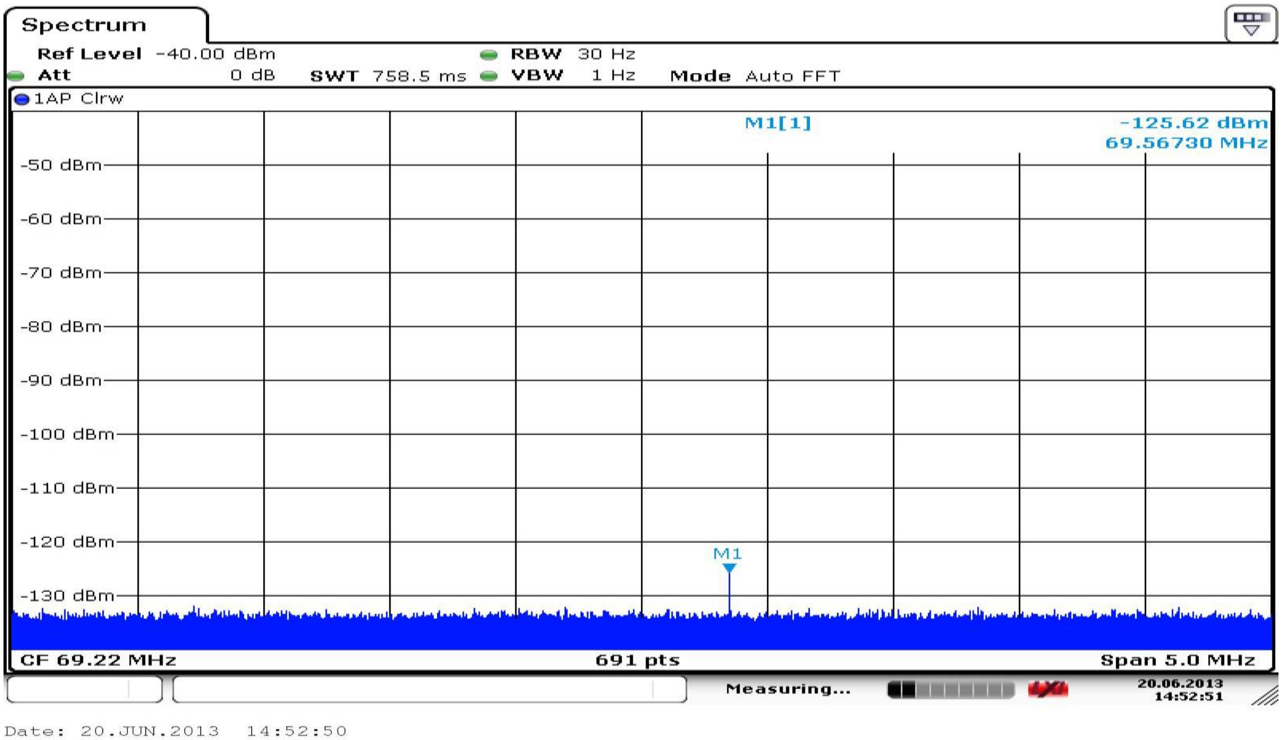
Power input to RF port : -50 dBm
LO-RF isolation : 45.74 dB
RF leakage to LO port : (-50-45.74) = -95.74 dBm
12 way power divider isolation(for 200 MHz LO) =28.33dB
Power leakage to adjacent J41 unit(for 200 MHz LO) = (-95.74 – 28.33) = -124.11

PRACTICAL MEASUREMENT :

A RF signal of 175/130 MHz,-50 dBm through a signal generator and a LO signal of 105/200 Mhz that is taken through the power divider from the tapping was given as an input to the J41 unit. The output of the J41 is terminated using a 50 ohm resistance. The RF input of the adjacent J41 unit is also terminated using a 50 ohm resistance. The LO input port is given a signal of 105/200 MHz and the output of the adjacent J41 unit is connected to the spectrum analyzer. The output seen on the spectrum analyzer

OBSERVATIONS :

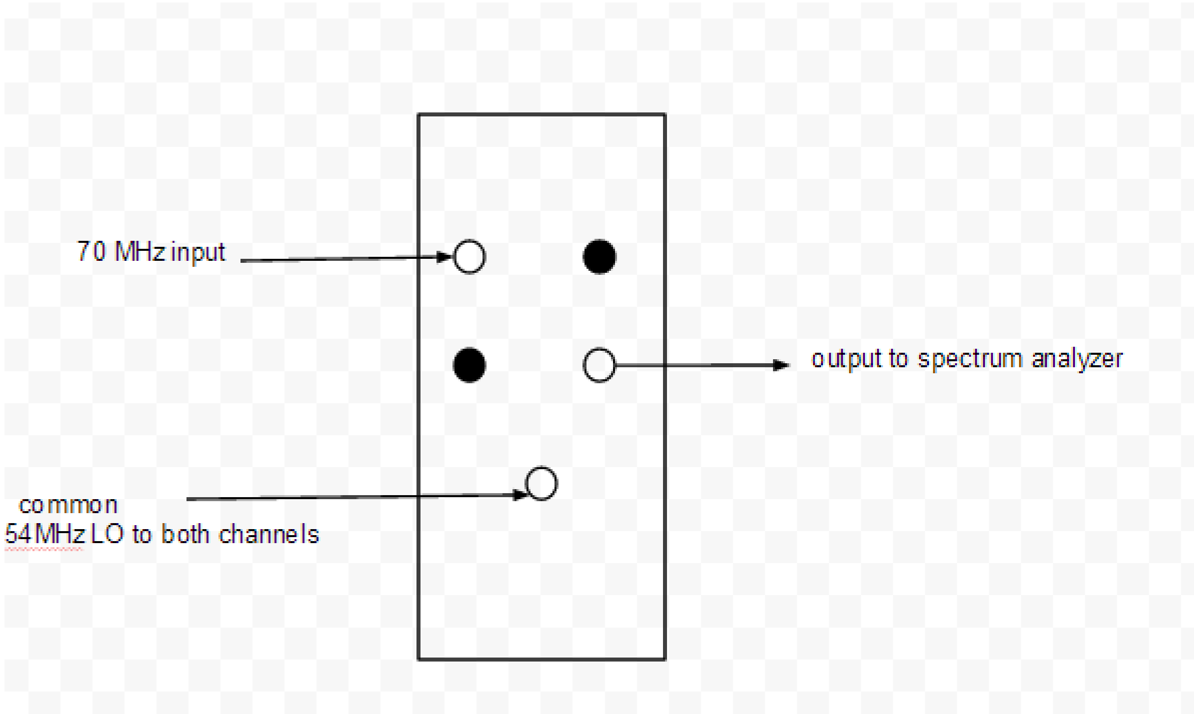
1) The J41-J41 leakage was below -135dBm for the 175 MHz channel. No peak was observed even when the noise floor was bought down to -135dBm
2) The J41-J41 leakage for the 130 MHz channel was found to be an average value of about -126dBm



CONCLUSION :

1) The practical value of J41-J41 leakage is found to be lower for the 175 MHz channel.
2) For the 175 MHz channel, a theoretical reading -130.88 dB was obtained. No peak could be seen on the spectrum analyzer during practical measurement with the noise floor at -135 dBm
3) For the 130 MHz channel, a theoretical reading of -124.11 dB and a practical reading of -126dBm was obtained

5.2.3 INTER CHANNEL LEAKAGE



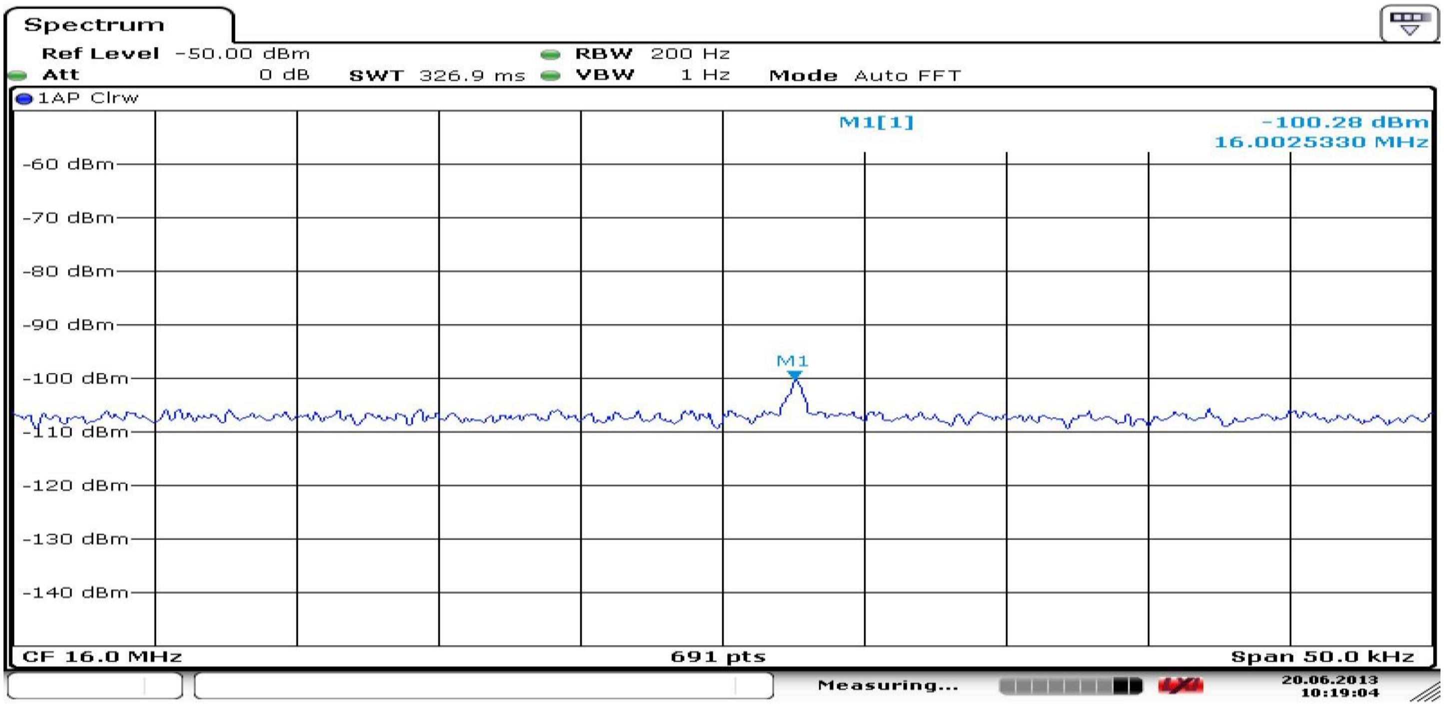
THEORETICAL CALCULATION :

Power input to RF port : -50 dBm
Calculated power level at input stage to mixer in the new IF conversion unit : -28

LO-RF isolation : 56.45 dB
RF leakage to LO port : (-28 - 56.45) = -84.45 dBm
ADP 2-1 power divider isolation =31.23dB
Inter channel leakage = (-84.45 – 31.23) = -115.68

PRACTICAL MEASUREMENT :

A 70 MHz,-50 dBm input is given to channel 1 and the respective output terminal of channel 1 is terminated using a 50 ohm resistance.. The input of channel 2 is also terminated in a similar manner. A LO input of 16 MHz is given which is common to both channels. The output terminal of channel 2 is connected to a spectrum analyzer and the leakage is determined..



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CONCLUSION :

The theoretical value of inter channel leakage is -115.68
The practical value as observed on spectrum analyzer is -100dBm

6. CONCLUSION

Over the course of this project we have learnt about the super heterodyne principle and how it was applied to the units we were working with. We also learnt about the many figure of merits of a receiver system. We have also determined many of these figure of merits by calculating them theoretically and have also determined the same figures of merits practically with the help of an antenna unit , signal generator and a spectrum analyzer.

A program in C++ was also written to help simplify the theoretical calculations involved. The program can be extended to any number of units and will accurately determine the various parameters. The program can be used later as a stand alone program for the analysis of other system and can be modified and worked upon to be make it usable for a wide variety of uses.

We have also dealt with the leakage of signals from either one channel to another or from one unit to another. These leakages were then theoretically calculated and verified practically by conducting tests to measure these leakages in the various units. All the results and observations were noted and discussed.